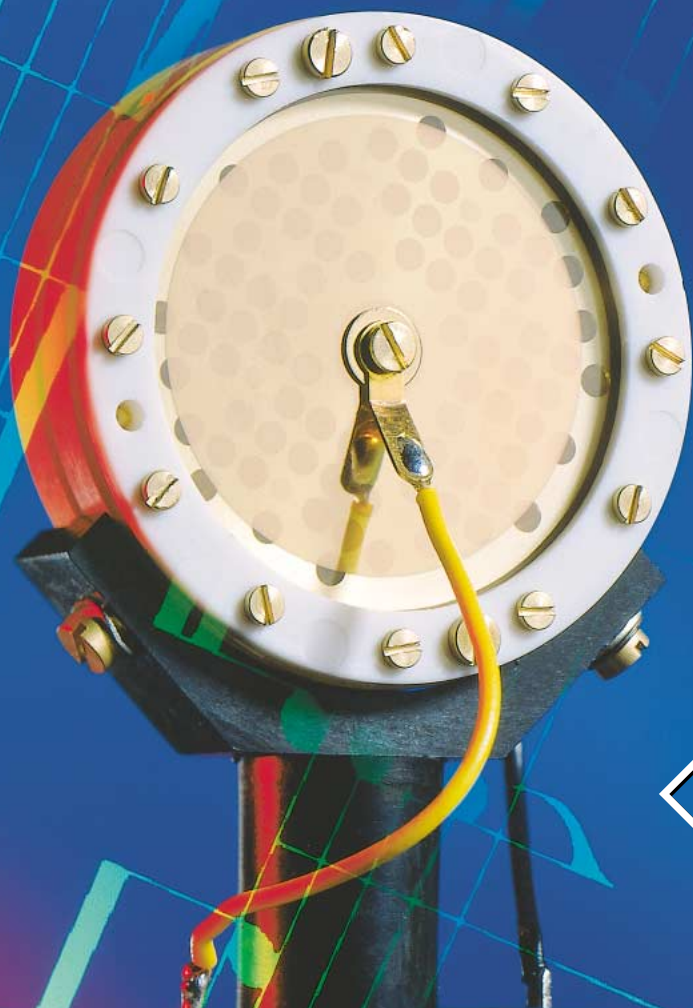


Gerhart Boré / Stephan Peus

Microphones

Methods of Operation and Type Examples



Microphones for Studio and Home-Recording Applications

Operation Principles and Type Examples

by Dr.-Ing. Gerhart Boré and Dipl.-Ing. Stephan Peus

An Engineering Service of

Georg Neumann GmbH | Berlin



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Preface to the fourth edition

The microphone is the first link in every chain of electroacoustic transmission. Problems caused by poor inherent characteristics or unfavorable microphone placement can seldom be corrected by subsequent processing equipment. Thus it is worth learning about the properties and quality attributes of the various types of microphones available.

This work was originally written as a contribution to the German book “Taschenbuch der Unterhaltungs-Elektronik” (“Handbook of Entertainment Electronics”). Its purpose is to convey some basic knowledge about microphones and the way in which they work, to all who are professionally or privately involved or interested in the recording and transmission of sound, and to help them use microphones effectively. No specialized knowledge is required. Readers will learn about the various types of microphones in existence and the characteristic features of those most commonly in use. Following a resume of some of the aspects common to all types, condenser and dynamic microphones are described in somewhat more detail.

Above all, this fourth edition has been revised to reflect newer developments in microphones and updated standards. On the whole, however, the character and style of the “Boré booklet” has been preserved. This small compendium has served as companion and tutorial guideline for generations of sound engineering students and other persons interested in audio engineering. We hope that it will continue to provide guidance and inspiration to microphone users everywhere.

The author thanks Mr. M. Schneider for valuable comments and additions.

S. Peus

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1. Terminology of microphone characteristics

Terms defining the characteristics of microphones are laid down in IEC 60268-4. A glossary of these terms, which are essential for understanding microphone specifications, is provided in the Appendix, together with brief explanations of their meanings.

2. Classification

Microphones can be classified according to various criteria:

Passive transducers*) convert acoustical energy directly into electrical energy (and vice-versa) without the need for any external power feed. This group includes dynamic, magnetic and piezoelectric microphones, as well as condenser microphones using dc polarization.

Active transducers*) convert electrical energy from an external source synchronously with the sound vibrations they receive. Carbon microphones and RF-condenser microphones make use of this principle.

Microphones which operate as **displacement-controlled transducers** generate a voltage output which is proportional to the displacement of the diaphragm. This applies to all capacitive and piezoelectric microphones. The ohmic resistance of carbon microphones likewise varies in proportion to the diaphragm displacement.

Velocity transducers is a term applied to all magnetic microphones operating on the basis of the law of induction. Their output voltage is proportional not to the extent, but to the velocity of the diaphragm displacement.

Almost all microphone types are manufactured according to their application: 1. as 'pressure transducers' (predominantly omnidirectional) or 2. as 'pressure gradient transducers' with special directional properties.

*) These definitions are taken from IEC Publication 50-08, Section 15. They appear to be somewhat less logical than the formerly recognized classifications as "pure transducers" and "control voltage transducers", although based apparently on terms used in studio technology, in which, for example, an "active" equalizing network must have an electrical power source, while a "passive" equalizing network manages without one. – With measured-variable transducers, the distinction between "active" and "passive" is mostly in the exact opposite sense.

All microphones will show a special response when their dimensions approach or become greater than the wavelength of the impinging sound. This effect is at first disregarded in the following, since it is discussed later in a special section.

Table 1 offers an overview of the most important microphone classes and types used in practice. For professionals and home-recording hobbyists, condenser and dynamic types have become the microphones of choice.

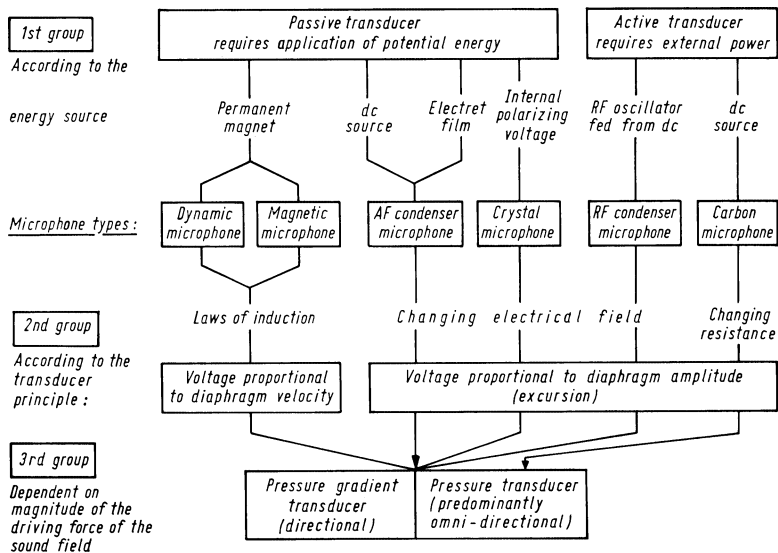


Table 1 Classification of microphone types

3. Microphone fundamentals

3.1 Pressure transducers

Only the front face of a pressure transducer is exposed to the sound field. The diaphragm responds equally to all sound pressure fluctuations occurring at its surface, regardless of the direction from which the sound waves emanate. Pressure transducers thus have no directional characteristic, and are by nature “omnidirectional”.

3.2 Pressure gradient transducers

These microphones have a figure-8 directional pattern along the longitudinal axis. They respond to the momentary sound pressure difference occurring between two points A and B, which are a slight distance apart in the sound field.

In the arrangement shown in Fig. 1, sound waves from 0° and 180° produce the largest difference in sound pressure and are received most strongly. By contrast, the sound arriving from 90° is received simultaneously and at the same strength at A and B, and thus produces no pressure difference. The field transmission factor T_F or sensitivity of such microphones conforms to the rule:

$$T_F = T_{F_0} \cdot \cos \vartheta$$

T_{F_0} = field transmission factor (sensitivity) with sound arriving perpendicularly to the diaphragm.

ϑ = angle between the perpendicular to the diaphragm and the direction of sound incidence.

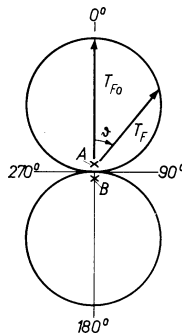


Fig. 1 Figure-8 characteristic

The sound pressures occurring at points A and B are compared either electrically or mechanically by the microphone.

In the first case the output voltages of two identical, closely adjacent microphone units are connected in antiphase. In the second case both the front and rear of the diaphragm are exposed to the sound field, so that only the momentary difference between the forces acting at the front and the back results in a movement of the diaphragm. The difference between points A and B is determined by the path that sound impinging at 0° or 180° must travel from one side of the diaphragm and its mounting around to the other diaphragm face.

Since every pressure difference occurring in the sound field immediately gives rise to a sound particle velocity in the direction of this pressure difference or gradient, the voltage output of pressure gradient microphones is always proportional to the sound particle velocity.

Sometimes, these microphones are also referred to as “velocity microphones”. However, it would be preferable to confine this term to microphones with compliant diaphragms, i.e. possessing little acoustic impedance, and thus largely following the movement of the air particles.

3.2.1 Behavior in a plane sound field

In an approximately plane sound field, a pressure difference between points A and B occurs only because sound impinges on both points at equal strength but with a phase difference. Since the distance between points A and B is often no more than a few centimeters, the phase angle φ and the resultant pressure difference Δp are very small for long wavelengths ($\varphi = 1^\circ \dots 3^\circ$ at 40 Hz). At constant sound pressure, they increase with rising frequency, that is to say the pressure gradient, in contrast to the sound pressure, represents a driving force which increases as the frequency rises.

This behavior is illustrated in Fig. 2:

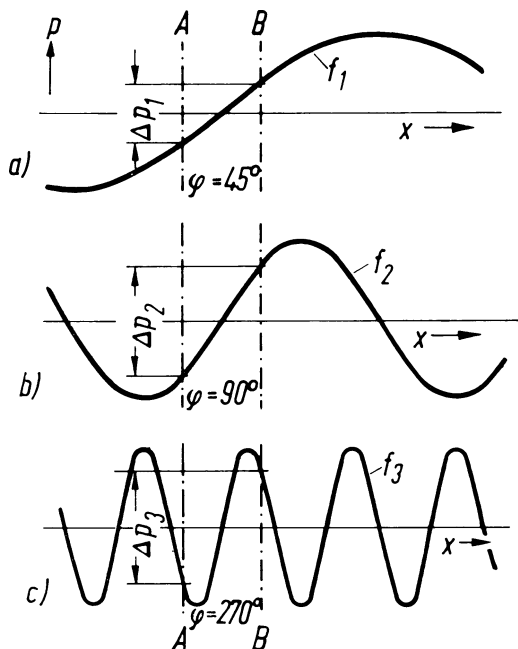


Fig. 2 Occurrence of sound pressure differences $\Delta p_1 \dots \Delta p_3$ between two points separated by the distance A-B at three different frequencies (in plane sound waves moving from left to right).

The amplitude distribution of three sound pressure waves travelling in direction x is represented at a particular instant in time, all three waves having the same amplitude but different frequencies $f_1 \dots f_3$. At the points $x = A$ and $x = B$, the three waves give rise to three different momentary sound pressure values: in Fig. 2 b), the double frequency f_2 results also in approximately twice the Δp value (and thus twice the displacement force applied to the diaphragm) as compared with f_1 . In practice, the distance A-B cannot be made so small that it is smaller than the half-wavelength $\lambda/2$ for all frequencies of the response range. A certain frequency f_t is characteristic for each microphone type. At frequency f_t , the half-wavelength $\lambda/2$ equals the path A-B, so that $\varphi = 180^\circ$. Above f_t the sound pressure difference Δp , which is decisive for the diaphragm movement, becomes smaller again (Fig. 2c and Fig. 3).

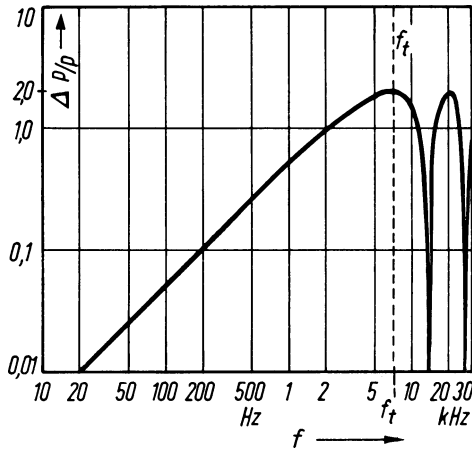


Fig. 3 Frequency response of the pressure gradient, i.e. the pressure difference Δp between two points in a plane wave sound field. Separation of points 25 mm (f_t = transition frequency).

Microphones designed to work satisfactorily above the frequency f_t must therefore make use of another principle when this frequency (between 4 and 10 kHz, depending on the size of the microphone) is approached. At the highest frequencies, they must operate as pressure or interference transducers.

3.2.2 Behavior in a spherical sound field

When a point source of sound is approached, the sound pressure rises at a ratio of $1/r$ (r = distance). Therefore in a spherical sound field, there is at the two sampling points A and B of a pressure gradient microphone a distance-related as well as a phase-related pressure difference (Fig. 4). This distance-related pressure difference results from the difference in the distances r_1 and r_2 of the two points A and B from the sound source Q. This distance-related sound pressure difference is of equal magnitude for all frequencies, and unlike the phase-related difference of section 3.2.1, is not frequency dependent.

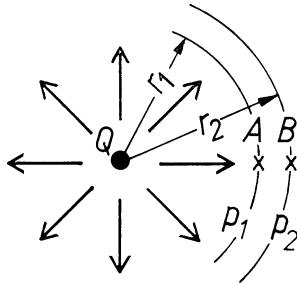


Fig. 4 Spherical sound field ($Q = \text{sound source}$)

For this reason, its effect is most noticeable in the low frequency range, where the forces acting on the diaphragm as the result of the phase shift are weakest (Fig. 3). The practical result is that pressure gradient microphones tend to boost low-frequency components when held close to the mouth, i.e. when the distance r from the sound source is approximately equal to the length of the sound wave (Fig. 5).

For small microphones, this is expressed by the function

$$\frac{e_8}{e_0} = \frac{1}{\cos \alpha}, \text{ where } \tan \alpha = \frac{\lambda}{2 \pi r} = \frac{54.14}{f \cdot r}$$

where:

- $e_8 =$ output voltage of a pressure gradient microphone with figure-8 characteristic,
- $e_0 =$ output voltage of an omnidirectional microphone with the same sensitivity at 0° ,
- $r =$ microphone distance from a point source of sound in meters,
- $\lambda =$ wavelength in meters,
- $f =$ frequency in Hz.

Example:

At a microphone distance $r = 10 \text{ cm}$ and frequency $f = 40 \text{ Hz}$, the boost amounts to

$$\tan \alpha = \frac{54.14}{40 \cdot 0.1} = 13.5; \quad \alpha = 85.77^\circ; \quad \cos \alpha = 0.074$$

$$\frac{e_8}{e_0} = \frac{1}{0.074} = 13.57 \quad 20 \log 13.57 = 22.65 \text{ dB}$$

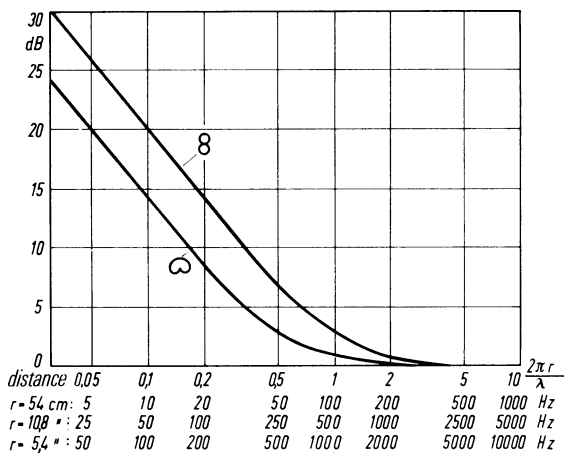


Fig. 5 Increase of the field transmission factor of pressure gradient microphones when close to speaker

3.2.3 Cardioid characteristic

Of special interest is the result when an omnidirectional and a figure-8 characteristic are superimposed – namely the so-called cardioid characteristic as shown in Fig. 6, provided that both components are of equal magnitude at the zero incidence angle.

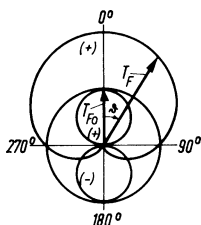


Fig. 6 Cardioid characteristic resulting from the superposition of an omnidirectional and a figure-8 pattern

Represented in polar coordinates, the rule here is:

$$T_F = T_{F_0} \cdot (1 + \cos \vartheta)$$

ϑ = angle of incidence of sound
 T_F = field transmission factor
 T_{F_0} = field transmission factor for $\vartheta = 0^\circ$, i.e. sound impinges from the front.

The distinguishing feature of a microphone of this type is its unilateral sound pick-up over a wide angle of aperture. Three different versions are possible:

- a) The output voltages from two closely adjacent capsules, one with omnidirectional, the other with figure-8 polar pattern, are coupled together.
- b) One part of the diaphragm has only its front face exposed to the sound field, while another part has both faces exposed to the sound field.
- c) A pressure gradient transducer is constructed so that the sound arrives at the rear face of the diaphragm via an acoustic delay element. Most of today's cardioid microphones operate on this principle. Sound openings to the back face are designed as an acoustic low-pass filter, the collective transit time of which corresponds to the desired sound delay time t_1 through the back face and whose limit frequency f_t ensures that the back face is barred to high frequencies. This turns the microphone into a unidirectional pressure transducer (comp. Section 3.2.1 and Fig. 3).

The operating principle of c) is depicted in Fig. 7, in which D represents the diaphragm, M a suitable mounting and L an arrangement through which sound waves require a specific transit time t_1 .

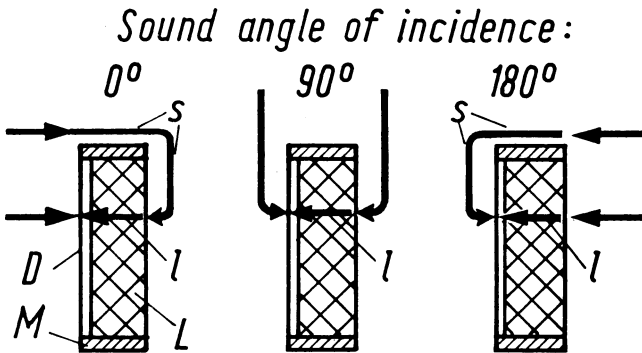


Fig. 7 Operating principle of a cardioid microphone with acoustic delay element

If the time t_1 is made to be equal to the time t_s required for the sound to traverse the path s , then since $t_s = t_1$ for 180° sound incidence, the forces acting on the front and rear face of the diaphragm will be in phase, and the diaphragm will remain motionless. Sound arriving at 0° is delayed on its way to the rear face of the diaphragm by $t_s + t_1$, at 90° by t_1 i.e. only half

as long. The total effect for a microphone with delay element is the same directional characteristic as that shown in Fig. 6 (omnidirectional and figure-8 characteristic superimposed).

The accentuation of low frequencies when the microphone is very close to the speaker commences one octave lower than with pure pressure gradient transducers with a figure-8 response (Fig. 5). For sound arriving at the front it can be calculated as

$$\frac{e_{\omega}}{e_0} = \sqrt{1 + \frac{\lambda^2}{16 \pi^2 r^2}} = \sqrt{1 + \frac{732}{f^2 \cdot r^2}}$$

e_{ω} = output voltage of a cardioid microphone,

e_0 = output voltage of an omnidirectional microphone with the same sensitivity at 0° ,

r = microphone distance from a point source of sound in meters,

λ = wavelength in meters,

f = frequency in Hz.

Example:

At a microphone distance $r = 10$ cm and frequency $f = 40$ Hz, the boost amounts to

$$\frac{e_{\omega}}{e_0} = \sqrt{1 + \frac{732}{40^2 \cdot 0.1^2}} = 6.84 \quad 20 \log 6.84 = 16.7 \text{ dB}$$

3.2.4 Hyper-, super- and subcardioid characteristic

Microphones as depicted in Fig. 6 or 7 are occasionally dimensioned so as to prevent sound impinging at the rear from resulting in absolute zero output. These have asymmetrical figure-8 response patterns as shown in Fig. 8. They have the advantage of suppressing sounds at 90° , i.e. sounds arriving at the sides, more effectively than microphones with a cardioid characteristic.

The degree of attenuation is as follows

angle of incidence	sub-cardioid	cardioid	super-cardioid	hyper-cardioid	figure-8
90°	2.5 ... 3.5 dB	6 dB	8,7 dB	12 dB	∞
180°	6 ... 10 dB	∞	11,5 dB	6 dB	0 dB

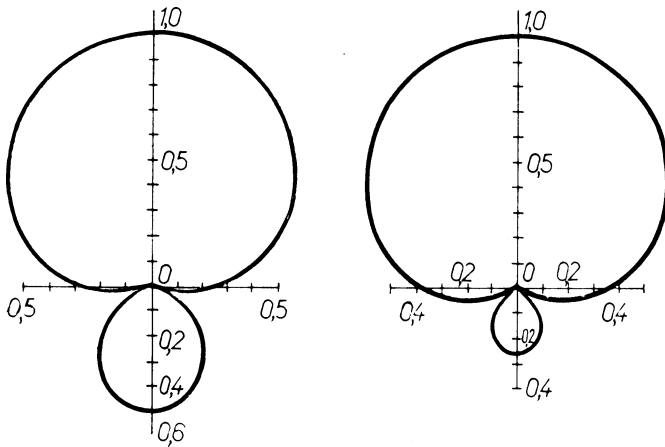


Fig. 8 Hypercardioid (left) and supercardioid (right) characteristics

The cardioid characteristic is most advantageous when it is necessary to attenuate sounds emanating from behind the microphone. The hypercardioid accentuates sounds arriving exactly from the front at 0° in preference to sounds of similar intensity emanating from all around, and is therefore able, for instance, to give greater prominence to a speaker or solo instrumentalist in comparison with reverberation sound. Cardioids and figure-8's pick up in diffuse sound fields only a third, hypercardioids only a quarter of the sound that an omnidirectional microphone having the same sensitivity for sounds at 0° incidence would register.

The parameters for supercardioids lie somewhere inbetween. If the recording room is imagined as a front and rear half-room divided by a plane including the microphone diaphragm, then the supercardioid represents the one special case in which the difference between the sounds picked up from the two half-rooms is at its greatest.

The subcardioid (also: hypocardioid, wide cardioid, wide-angle cardioid) lies halfway between the omnidirectional and the cardioid characteristic. It can be useful for recording sound sources extended over a wider angle.

For the user, it is often more important that a directional microphone can be placed further away from the speaker without any significant loss of direct sound, and thus of transparency and presence. The distance for a sub-

cardioid may be 1.3 ... 1.4 times, for a cardioid or figure-8 1.7 times, for a supercardioid 1.9 times and for a hypercardioid 2.0 times that of an omnidirectional microphone.

3.3 Influence of microphone dimensions

Whereas the frequency bandwidth of visible light - expressed in the terminology of acoustic - is less than a single octave, audible sound covers the best part of ten octaves.

Examples of the wavelengths of sound in air are

at 16 000 Hz	2.1 cm
at 3 200 Hz	10.5 cm
at 320 Hz	105 cm = 1.05 m
at 32 Hz	1 050 cm = 10.5 m

Microphones with dimensions similar to or greater than the wavelengths being picked up present an obstacle for the sound waves: these are either partially or completely reflected upon reaching the microphone. At the same time, sound arriving perpendicularly to the diaphragm exerts, depending on the shape of the microphone, up to 10 dB more force on the diaphragm as the result of pressure build-up.

With sound from the side or rear bending and shading effects appear. Sound waves impinging diagonally do not strike all parts of the diaphragm simultaneously, giving rise to interference cancellations that are dependent on both direction and frequency. Microphones that depend chiefly on these effects for their directional pattern are therefore termed "interference transducers" (s. Sec. 6.8).

All effects caused by the dimensions of the microphone, particularly the directional characteristic, are of course frequency-dependent. Nevertheless, most microphones change to interference transducers at the upper end of their response range. Otherwise they would have to be no larger than 6 mm in all three dimensions for a limit frequency of 16 kHz.

3.3.1 Pressure microphones

Pressure microphones are most markedly affected by the fact that at high frequencies their omnidirectional characteristic gradually changes to a unidirectional polar pattern – in the case of larger microphones, even to a constantly narrowing lobe shape (Fig. 9). At the same time, sound arriving at the front builds up pressure and the sensitivity of the microphone to higher frequencies increases. This is the reason why the specifications of almost all

pressure microphones show a frequency response that rises by several decibels at the high end.

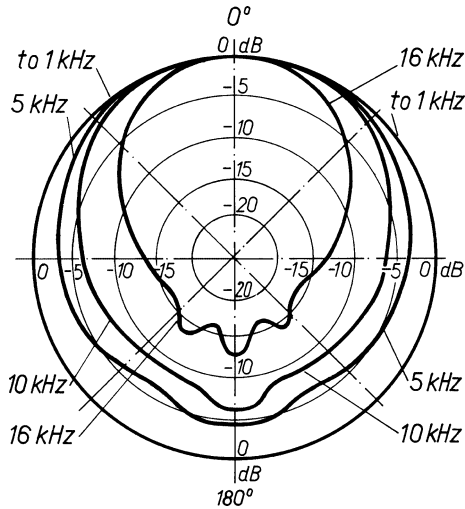


Fig. 9 Polar diagram of a pressure microphone mounted on the face of a cylindrical body 21 mm in diameter

3.3.2 Pressure gradient microphones

Microphones with figure-8 or cardioid characteristics can be designed so that their behavior does not noticeably change to that of an interference transducer at the frequency at which the force acting on the diaphragm reaches its maximum as the result of the pressure gradient in accordance with Fig. 3. This depends on the microphone diameter, the transition frequency and the delay as the sound travels around the microphone to the rear face of the diaphragm being closely correlated. Properly dimensioned, the growing pressure build-up already starts compensating below the transition frequency f_t (Fig. 3) for the gradual decrease in the diaphragm driving force caused by the pressure gradient; above the transition frequency, the microphone gains a single- or double-sided directional characteristic similar to that of a cardioid or figure-8 as the result of interference and shadowing effects. Since the microphones function as interference transducers only within a relatively narrow frequency band, it matters little that the directional characteristic is frequency-dependent in that range.

3.3.3 Free-field and diffuse-field frequency response

If the sound is being recorded in a room, the sonic impression, even for a relatively short distance between microphone and sound source, will hardly be determined only by the 0° free-field frequency response of the microphone, but rather by its diffuse-field frequency response. This applies to the sound reaching the microphone at more or less the same intensity from all spatial directions. The distance from the sound source at which the direct and diffuse components are of equal magnitude is referred to as the “reverberation radius”. The larger the room and the less ‘live’ (reverberant) it is, the greater will be the reverberation radius. It amounts, with volume V and reverberation time T , to

$$r_H = \frac{1.05}{\sqrt{c}} \cdot \sqrt{\frac{V}{T}} = 0.057 \cdot \sqrt{\frac{V}{T}}$$

c = sound velocity in air = 340 m/s.

At shorter distances than r_H , the direct sound predominates. A microphone located here should have a good free-field frequency response and a suitable directional characteristic. At longer distances than r_H , the diffuse sound arriving from all quarters predominates. From here onwards, the sound pressure level remains more or less constant in the whole of the room, and the quality of the recording is determined only by the diffuse-field frequency response of the microphone. A directional microphone will only accentuate to a moderate extent the small percentage of direct sound that arrives in advance of the reverberation and is, of course, important for directional orientation in stereo recording.

In connection with sound reinforcement systems, it should be borne in mind that as soon as the distance from microphone to loudspeaker is made greater than the reverberation radius, it is not possible to raise the critical point of acoustic feedback by employing a directional microphone. This is often not taken into consideration. With large distances, sound arrives at more or less equal intensity from all directions. The “effective reverberation radius” can be increased only by aiming the **loudspeakers** in directions from which reflections can be assumed to be at a minimum – the section of the room in which the audience is seated, for example.

The usual values for the reverberation radius lie between 0.5 and 2.5 m, depending on room volume and reverberation time.

Good-quality microphones today are expected to have free-field and diffuse-field frequency responses that run virtually parallel. By changing the

distance of the microphone from the sound source, it is possible to alter the so-called reverberation balance (ratio of direct sound to room reverberation) without altering the frequency response.

There are only a few pressure gradient microphones that really meet this requirement well. The diffuse-field sensitivity (measurable in a reverberation chamber) is 4.8 dB lower than the free-field sensitivity for microphones with an exact figure-8 or cardioid characteristic and 6 dB lower for hypercardioid microphones. This difference known as the "directivity index" of a microphone. The German HiFi Standard DIN 45 500 prescribes in Part 5 that the directivity index for all directional microphones must be at least 3 dB between 250 and 8000 Hz. In addition, the frequency response curves for all sound incidence angles other than 0° must, between 250 Hz and 8 000 Hz, run parallel within ± 4 dB to the frequency response for the 0° incidence angle, always provided that the field transmission coefficient does not lie 12 dB or more below the corresponding 0° figure. This is to ensure that sound sources positioned to the sides will be recorded with the same frequency response, although more softly.

Fig. 10 shows the polar diagram and frequency responses of a condenser microphone that meets these requirements perfectly.

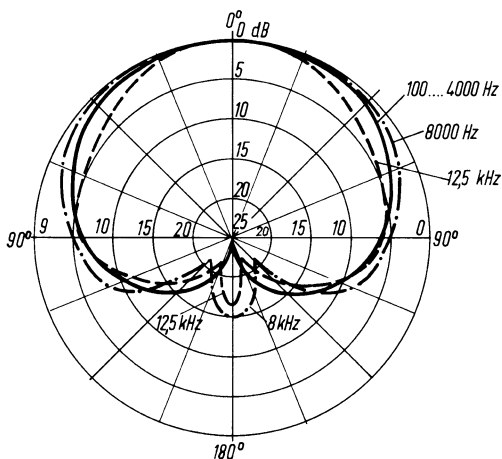
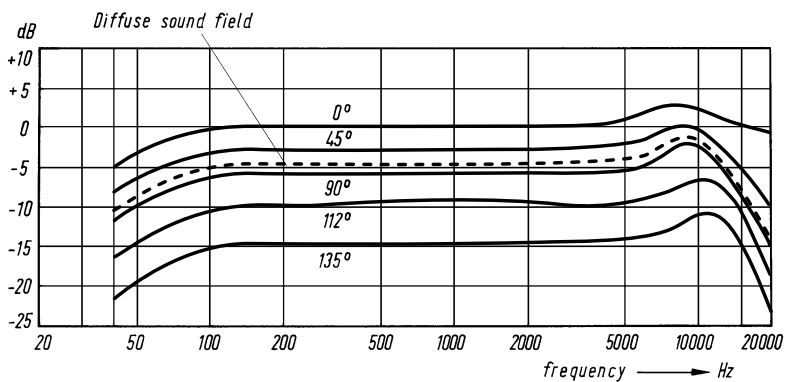


Fig. 10 Frequency responses and polar diagram of a small condenser microphone with cardioid characteristic (KM 84 i, Neumann)

Free-field and diffuse-field frequency response curves never match in standard-size pressure microphones with the exception of the pressure-zone microphones discussed in Section 6.11. These, if realized with a very small diaphragm diameter, are almost equally sensitive to sound arriving perpendicularly and laterally, even at high frequencies.

Most pressure microphones of standard size are built with a relatively flat diffuse-field frequency response, and the free-field accentuation is tolerated (Fig. 11). Due to the polar diagram, the diffuse-frequency response should roll off at high frequencies, as less sound is picked up from the back; however, this is compensated for by the fact that sensitivity to sound from the front increases at high frequencies.

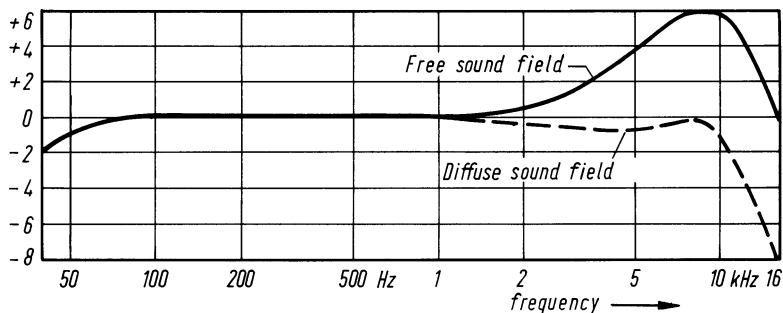


Fig. 11 Frequency response of a small pressure microphone in free and diffuse sound field

DIN 45 500, Part 5, specifies that the response of non-directional microphones to sound impinging laterally at 90° may not deviate by more than 2 dB at frequencies below 1 kHz and not by more than 8 dB at frequencies between 1 and 6.3 kHz from the response to sound arriving exactly from the front.

4. *Dynamic microphones*

Dynamic microphones are velocity transducers and their mode of operation is based on the law of induction: a conducting element is induced to move in a magnetic field by the influence of sound waves. The EMF thus generated is proportional to the velocity of the moving conductor.

A velocity of motion proportional to the sound pressure is produced in systems able to follow the movement of the air particles without any variation in frequency response, i.e. with no constraint except that of friction. The diaphragm resonance of dynamic pressure microphones is therefore arranged to be at the middle of the response range and is heavily damped.

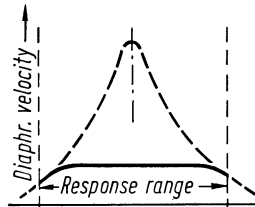
Dynamic directional microphones, which respond to pressure gradients or particle velocity and therefore are subject to increasing forces as frequency rises (as shown in Fig. 3), must be operated at the top of the downslope of their resonance curve in order to yield a flat frequency response, i.e. above their mechanical resonance frequency. This means that they must be “low-tuned” and predominantly mass-controlled in operation.

Fig. 12 illustrates these relationships and enables a comparison to be made with the condenser microphone discussed in Section 5.

For the user, it is important to know that microphones with a “low-tuned” diaphragm system are naturally more susceptible to disturbance by wind and body noise, footfalls, handling noises, etc. than microphones with mid- and high-tuned diaphragms.

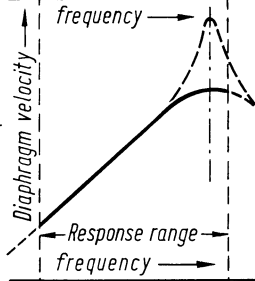
For constant sound pressure

....the particle velocity of air is equal at all frequencies



Dynamic pressure microphones (velocity dependent) are, therefore, „mid-band tuned“ and are resistance controlled

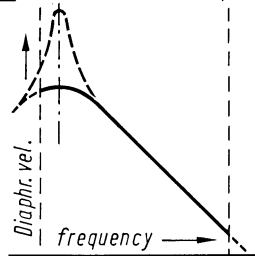
....the particle amplitude of air drops off at higher frequencies



Condenser pressure microphones (diaphragm-amplitude dependent) are, for that reason, „high frequency tuned“

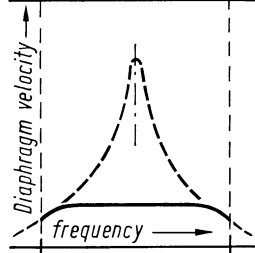
..... the pressure gradient increases with frequency (Fig.3)
Therefore

....the particle velocity of air increases with frequency



Dynamic gradient microphones (velocity dependent) are, therefore, „low-frequency tuned“

....the particle amplitude of air remains static for all frequencies



Condenser gradient microphones (diaphragm amplitude dependent) are „mid-frequency tuned“ and largely resistance controlled

lower boundary frequency upper boundary frequency

Fig. 12 Comparison of diaphragm resonance and response range of dynamic and condenser microphones

4.1 Ribbon microphones with figure-8 characteristic

In ribbon microphones, the sound field acts directly on the conductor, a strip of aluminium foil a few μm thick that is suspended so that it vibrates between the poles of a permanent magnet. The foil is usually 2 to 4 mm wide and a few centimeters long. Its very low impedance is stepped up to a conventional value of 200 ohms by a special transformer built into the microphone.

When both sides of the ribbon are exposed to the sound field, a figure-8 pattern results, and the microphone, because of the high compliance of the ribbon, may be designated a “velocity transducer”. The necessary low-tuning usually presents no problem, but it does result in a microphone which is more sensitive to rapid movements, vibration and wind than other types, and this sensitivity increases as the response range is extended to lower and lowest frequencies. On the other hand, ribbon microphones generally have a flat and resonance-free frequency response. Only the components constituting the magnetic ring, which starts from the two pole pieces forming the air gap and closes around the ribbon on the outside, may introduce some irregularity in the high frequency range.

This disadvantage is largely avoided in the microphone depicted in Fig. 13, in which the path of the lines of force is closed by the spiral of soft magnetic

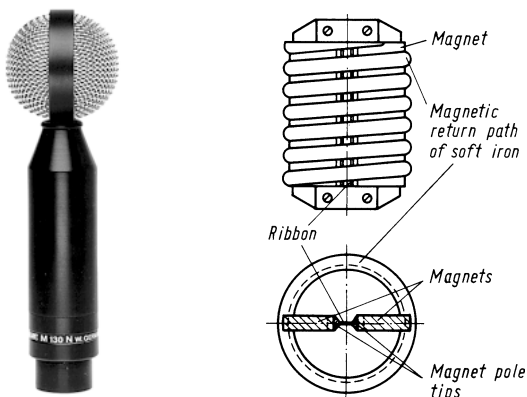


Fig. 13 Illustration and schematic diagram of a ribbon microphone with figure-8 characteristic by E. Beyer (M 130, beyerdynamic)

material enclosing the ribbon and pole pieces and causing little or no interference with the sound field. The transducer system of this microphone is so small that it can be enclosed in a spherical grill housing only 39 mm in diameter.

4.2 Ribbon microphones as pressure transducers

If the ribbon is intended to operate as an omnidirectional pressure transducer, it is terminated at the rear by a tube or cavity lined with acoustically absorptive material. This also acts as a frictional constraint for the ribbon.

4.3 Ribbon microphones with cardioid characteristic

A cardioid characteristic results if only a portion of the ribbon is terminated at the rear, while the remainder is exposed to the sound field on both sides.

In newer microphones, the cardioid characteristic is also produced by an acoustic delay unit arranged behind the ribbon (see also Section 3.2.3).

4.4 Moving coil microphones as pressure transducers

Moving coil microphones have a small self-supporting coil that is fastened to a lightweight plastic diaphragm and moves in the air gap of a powerful permanent magnet in similar manner to the voice coil of a dynamic loudspeaker. However, the mass of the diaphragm and coil together is many times greater than the diaphragm mass of a condenser microphone, and to flatten the frequency response only by damping the vibrating system would only make the microphone too insensitive.

All moving coil pressure microphones are therefore given a resonance “hump” in the approximate middle of their response range. This resonance, however, is scarcely noticeable, if at all, in the output, as the diaphragm is forced into additional damped resonances by air cavities in front and behind with communicating holes or slots. These have the effect of flattening and extending the frequency response upwards and downwards, as indicated in Fig. 14. Modern moving coil microphones can be designed so that their response comes close to that of a condenser microphone, apart from the steep roll-off at the upper and lower cut-off frequencies.

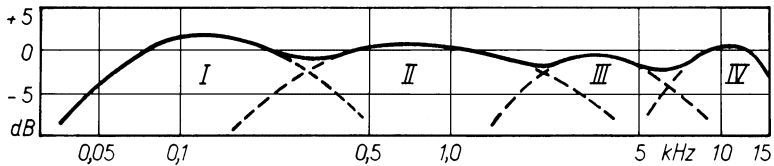


Fig. 14 Schematic frequency response of a moving coil pressure microphone. II = damped resonance of the vibrating system; I, III, IV = resonances of communicating air cavities

Alternating magnetic fields of the kind originating in mains transformers and in tape recorder motors, can induce interference voltages in the moving coil. This can be avoided with a counter-wound compensating coil, the so called “hum-bucking coil”, fitted in the microphone housing tube near to the moving coil and connected in series with it.

Compared to the abundance of unidirectional dynamic microphones of studio or HiFi quality only few omnidirectional types are available, even though pressure microphones are more robust and less sensitive to wind noise and shock noise. Because of these properties, most lavalier microphones, such as the one shown in Fig. 37, are made as moving coil pressure microphones.

4.5 Moving coil microphones with cardioid characteristic

Over the years, the design of moving coil directional microphones has become a special science – to a much greater extent than with other types. In order to lower the self-resonance of the system sufficiently, the moving coil must be provided with a very compliant suspension. This makes the microphone sensitive to shocks and wind noise, and the task of keeping the coil properly placed in the air gap becomes a problem. For this reason the diaphragm is made less compliant than would be actually necessary for the required “low-tuning” of the system. Furthermore, provision is made for a more powerful diaphragm actuating force at low frequencies: for the low-frequency sound components, special sound ports are provided at the back of the microphone to give the sound pressure a large phase shift before it impinges on the back surface of the diaphragm. Because of the diverse lengths of sound pressure delay paths within the microphone, acoustic means are used to ensure that only sound components of the relevant frequency range are allowed to pass through. This method is referred to as the “variable distance principle”.

The so-called “two-way principle” leads to more or less the same result: as with loudspeakers, two microphone systems are connected via an electrical

network. One system with a short acoustic delay transmits the high frequencies. Arranged behind it is a second moving coil system with a large acoustic delay for the low frequencies. The quality of the microphone is dependent to a great extent on the dimensioning of the crossover network, whose function is to ensure that the two frequency responses merge without any discontinuity and that there is no transient distortion along the roll-off flanks.

In all pressure gradient microphones having a long acoustical delay path to the rear face of the diaphragm for low-frequency sound, the proximity effect is considerably less than it is in microphones with a short delay path as shown in Fig. 5.

5. Condenser microphones

The basic construction of a condenser microphone is shown in Fig. 15: a diaphragm with a thickness of 1 ... 10 μm made of metal or metallized plastic is arranged very close to a perforated, electrically conductive oppositely-charged electrode (backplate). Impinging sound waves move the diaphragm and change its distance from the backplate and thus the capacitance of the air-dielectric capacitor formed by the diaphragm and the backplate. Due to the close proximity of the diaphragm to the backplate (5...50 μm), the restoring force and damping of the diaphragm are primarily determined by the air cushion behind it, and can be adjusted to the required value by a suitable choice of diaphragm spacing and by holes drilled in the backplate (but not through it).

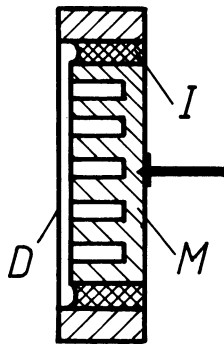


Fig. 15 Sectional view of a condenser microphone (pressure transducer): D = diaphragm, I = insulation, M = backplate

As a displacement-controlled transducer, the condenser microphone must be dimensioned so that the diaphragm undergoes approximately the same displacement at all frequencies in its response range, given constant sound pressure. However, the excursion of the air particles in the sound field decreases at constant pressure in inverse proportion to the frequency.

5.1 Pressure condenser microphones

This type is provided with a “high-tuned” diaphragm. The diaphragm mass (together with the vibrating air mass) is made small, the restoring force large. Blind holes in the backplate, and sometimes an additional air pocket diminish the stiffness of the air cushion sufficiently to bring the diaphragm

resonance into the region of the microphone's upper cut-off frequency. In the rising portion of the resonance curve (below the resonant frequency), the diaphragm velocity then increases with frequency and ensures the required constant diaphragm excursion (see Fig. 12).

5.2 Pressure gradient condenser microphones

The backplates of pressure gradient microphones are drilled all the way through. Since the pressure gradient already constitutes a driving force which increases proportionally to frequency (Fig. 3), the diaphragm may not be "high-tuned", but should have only a (frequency-independent) frictional constraint over the range of response. In practice, the diaphragm resonance produced by the diaphragm mass and the restoring force is placed at the middle of the frequency range in which the microphone is required to function as a gradient transducer. The resonance is then damped to such an extent by air friction in the gap between diaphragm and backplate and also in the backplate itself that it is no longer recognizable as such.

Nowadays only few manufacturers offer condenser figure-8s pure pressure gradient microphones. A figure-8 characteristic is most often produced by means of two closely-spaced cardioid characteristic capsules, whose principle axes are pointed in opposite directions, and which are electrically connected in anti-phase (so called double-diaphragm capsules).

5.3 Cardioid condenser microphones

In order to obtain a cardioid polar pattern, two different principles are applied, which are referred to as "b" and "c" in Section 3.2.3. In some microphones, one portion of the backplate is provided with through holes, another portion with blind holes.

The capsules thus function partially as a pressure gradient transducer and partially as a pressure transducer, the end product being a cardioid characteristic, as shown in Fig. 6.

In other microphones, the backplate is designed as a time delay component and provided with holes, slots and pockets which act partly as friction resistances and partly as energy storers (acoustic inductances and capacitances), lending the backplate the character of an acoustic low-pass network. In the cut-off range of this lowpass network above the transition frequency f_t (Fig. 3), sound impinges on the diaphragm only from the front, and the microphone capsule takes on the character of a pressure or interference transducer.

Although this design enables more exact polar diagrams to be obtained, microphones which function according to the first-named principle enjoy a popularity for certain requirements – possibly because their pressure transducer portions (on which sound impinges only from the front) are able to handle transients with greater precision.

5.4 Condenser microphones with symmetrical capsules

With the introduction of digital technology in most studios, non-linear distortion and inherent noise in such equipment has become negligible, and nowadays these two parameters are more dependent on the specifications of the microphones used. In the case of condenser microphones, both types of interference could be further reduced by the use of symmetrically arranged capsules, whereby both sides of the diaphragm are opposed by a fixed but acoustically transparent counter-electrode.

Because these two counter-electrodes are at the same voltage and their electrical forces of attraction acting on the diaphragm cancel each other out, the diaphragm is not unilaterally biased, and the (admittedly already small) 2nd order harmonic distortion caused chiefly by this initial tension is reduced. Furthermore, distortion due to a non-linear behavior of the air inside the drilling and slots of the backplate at high sound pressure levels is reduced because they are compensating each other by using two identical counter-electrodes at both sides of the membrane. Since the movement of the electrically biased diaphragm gives rise to alternating voltages in both counter-electrodes, the output voltage of the microphone is, with appropriate circuitry, doubled and its signal-to-noise ratio correspondingly improved.

Fig. 16 shows the design principle, Fig. 17 a condenser microphone with cardioid characteristic using this principle.

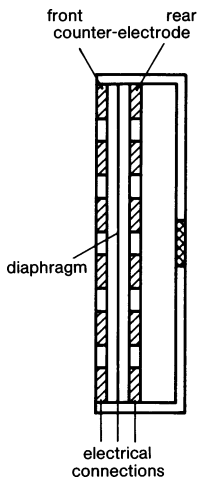


Fig. 16 Design principle of a symmetrical condenser microphone capsule with cardioid characteristic



Fig. 17 Condenser microphone with symmetrically constructed capsule (MKH 40 P48, Sennheiser)

5.5 Dc polarizing method

In the dc polarizing method, the capsule of the condenser microphone is charged via a high-value resistor R to a fixed dc voltage E_0 (40 ... 200 V) as shown in Fig. 18. For the charge Q the expression is:

$$Q = C_0 \cdot E_0 \quad C_0 = \text{capsule capacitance}$$

The resistor R is chosen to be sufficiently high so that the charge Q remains virtually constant in spite of variations in capacity brought about by impinging sound pressures.

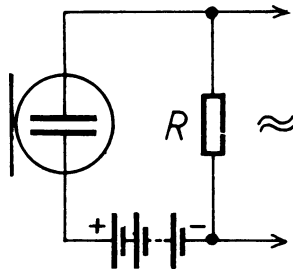


Fig. 18 Circuit of a condenser microphone using dc polarization

For the bottom limit frequency f_u , for which this condition holds good:

$$R = \frac{1}{2 \pi \cdot f_u \cdot C_0} \cdot R \text{ in ohms, } C_0 \text{ in F, } f_u \text{ in Hz.}$$

Since the values of C_0 usually lie between 20 and 100 pF, R must have, for example, for a bottom limit frequency of 20 Hz, a value between 400 and 80 megohms.

The output voltage $e(t)$ of a condenser microphone using dc polarization is proportional to the applied dc voltage E_0 and – for small diaphragm amplitudes – the relative change in capacity

$$\frac{c(t)}{C_0} \text{ caused by the sound pressure:} \quad e(t) = E_0 \cdot \frac{c(t)}{C_0}$$

$c(t)$ = variable component of capsule capacity
 t = time

5.5.1 The selection of various polar patterns

The dependence of the output voltage $e(t)$ on E_0 is also exploited in some microphone types to control the directional characteristic. To this end, two capsules with cardioid characteristic are placed back-to-back or they can be assembled as a unit with a common backplate, as shown in Fig. 19. The alternating voltages generated on both diaphragms are connected in parallel via a capacitor C . The ratio of the alternating voltage outputs from the two capsule halves and their phase relationship are affected by varying the dc voltage applied to one of them (here the one on the left) either by means of a switch or steplessly by means of a potentiometer. The directional characteristic of the microphone may thus be changed by remote control via long extension leads.

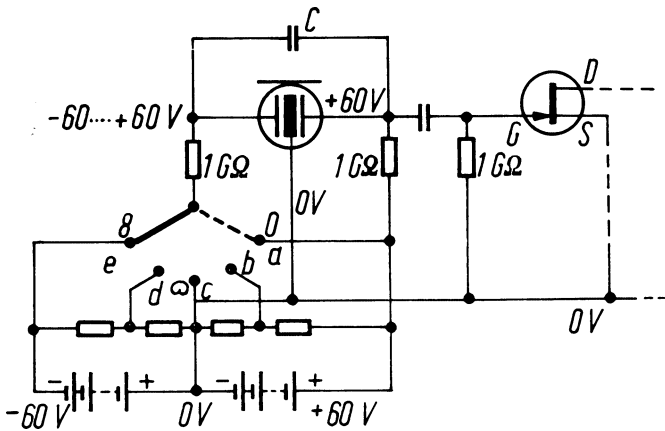


Fig. 19 Circuit of a condenser microphone with electrically switchable directional characteristic

With the switch in Fig. 19 in the center position (contact "c"), the left capsule-half does not contribute any voltage, and the microphone has the cardioid characteristic of the right capsule-half. In switch position "a", the two ac voltages are in parallel, in position "e" they are in counter-phase and the result is an omnidirectional and a figure-8 pattern, respectively.

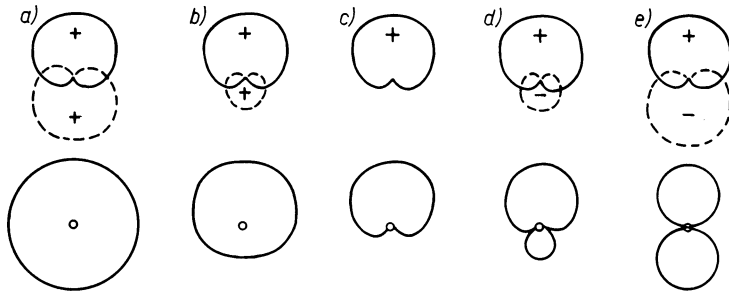


Fig. 20 Polar patterns (below) obtainable with a microphone as shown in Fig. 19 by superimposing two cardioid patterns (above)

The letters “a” to “e” given for the switch positions in Fig. 19, produce the patterns given the same letters in Fig. 20.

The high impedance of the condenser microphone capsule with dc polarization makes it necessary to locate the first amplifier stage close to the capsule. The required high input impedance of the amplifier can be achieved only by the use of field effect transistors (FETs) or vacuum tubes. The term “condenser microphone” denotes a combination of microphone capsule and amplifier stage.

5.6 Electret microphone capsules

The transistorized amplifier of a dc-polarized condenser microphone can be operated by low-voltage dc, for it is only the capsule polarization that requires a higher voltage (no current being drawn). In order to avoid undue elaboration, some manufacturers have taken to providing such microphones with permanently polarized electret foil membranes.

The unordered charge carriers present in foils of poor conductivity are activated by warmth and aligned by the effect of a powerful electric field in such a way that they form dipoles. On cooling down again, these are, so to speak, “frozen” in place, and are capable of retaining a constant charge at the surface. However, for the longest life, materials are used which do not tend to form dipoles, but can accept and maintain space charges. A typical material of this kind is polytetrafluorethylene, also known as “Teflon”.

To incorporate the negative charge carriers, the film is subjected to a so-called “corona discharge” or to electron bombardment in a vacuum.

Although the electret effect has been known for decades, it required a long time until materials had been found which are capable of permanently maintaining the “frozen-in” polarization voltage not only at high temperatures, but also at high humidity levels.

When unilaterally metallized, electret foils can serve directly as microphone diaphragm. As their acoustic properties are sub-optimal, higher quality microphones adapt the “back-electret” technique: The foil is mounted on the surface of the back electrode, and the diaphragm can thus be realized using the standard materials. This of course has the effect of reducing the basic capacitance C_0 (Section 5.5) of the microphone.

The electret microphone operates on the same general principle as the condenser microphones described in the foregoing section with externally supplied polarization voltage, but the latter is now superfluous.

5.7 The microphone amplifier

The dynamic range of many condenser microphones is determined less by the capsule than by the associated microphone amplifier, in which it is limited at the bottom end by inherent noise and at the top end by the increase of non-linear distortion.

Fig. 21 shows the circuitry of a microphone amplifier using a field effect transistor.

Its noise spectrum at low frequencies is dominated by the noise contribution of the high-value resistor at the FET gate, which is increasingly shunted by the capsule capacitance as the frequency rises.

However, in actual practice, this low-frequency noise is not disturbing, as the human ear is largely unreceptive to low-frequency sound waves.

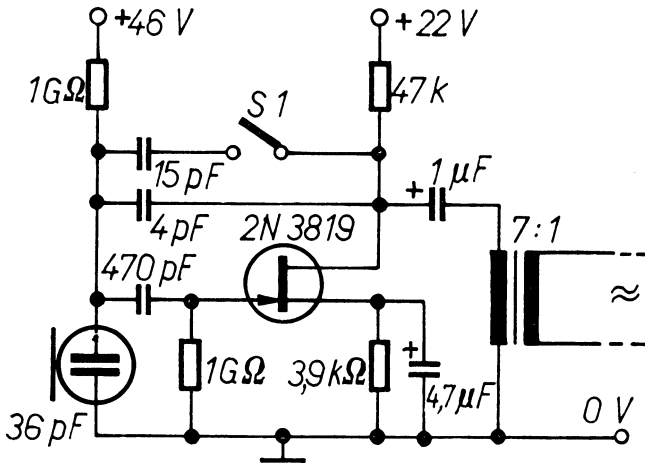


Fig. 21 Simplified circuit diagram of a condenser microphone with 10 dB overload protection switch (Switch S_1)

Above 1 ... 2 kHz, the inherent noise of the field effect transistor itself predominates. This is the reason why only FETs with particularly low noise voltage specification in the mid-frequency range are used for condenser microphones. At the higher end of the response range, the noise components emanating from the acoustic friction resistance of the microphone capsule can no longer be ignored.

According to DIN 45 500, the overload limit of the amplifier must, for semi-professional use, be so high that an overload sound pressure of 10 Pa (114 dB over 20 μPa) can still be handled at less than 1% distortion. The overload sound pressure of older studio microphones is 20 ... 30 Pa (120 ... 124 dB), and, with some types, can be raised by e.g. 10 dB through switchable preattenuation (overload protection switch) between capsule and amplifier.

Newer microphone types can be driven to such an extent that they will handle sound pressures of 160 Pa (138 dB), and, with preattenuation, even 500 Pa (148 dB), without distortion. At these sound pressure levels, non-linear distortion in the microphone capsules gradually begins to intrude. While it is true that such high sound pressures do not occur in musical performances, the high overload stability of the amplifier is nevertheless a very

favorable factor, especially in the case of microphones held close to the mouth of a speaker or singer, or when used to record a loud musical instrument.

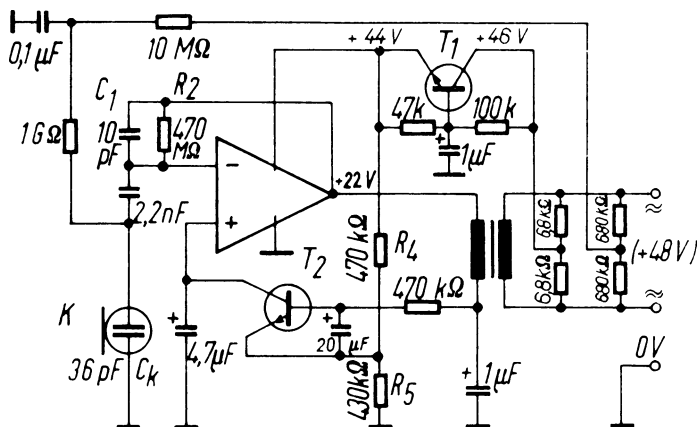


Fig. 22 Condenser microphone with operational amplifier fed by a 48 V phantom supply (s. Section 5.9.2). The capsule works into the FET-equipped inverting input. The value of C_1 determines the voltage amplification C_c/C_1 . The low-frequency cut-off depends on R_2 . T_2 stabilizes the operating points of the circuit: The dc voltage at the amplifier output is compared in T_2 with the mid-point voltage between R_4 and R_5 , phase-inverted and amplified and then fed to the non-inverting input as dc negative feedback. T_1 is used as a filter and electronic resistance (marked R_3 in Fig. 27).

Whereas older dc polarized condenser microphones were occasionally susceptible to interference at high humidities, today's high quality transistorized microphones are extremely stable even under such conditions, despite their very high input impedance.

Apart from design measures, this property is attributable mainly to the use of special insulating materials that do not absorb moisture and also do not permit surface wetting.

5.7.1 Transformerless microphones

Today's space-saving semiconductor technology has made it possible to replace the microphone's output transformer by semiconductor circuits without sacrificing any of the main advantages of transformer decoupling:

Matching the microphone circuit to the line impedance and symmetry of the microphone output in relation to 0-volt potential, and thus protection against interference liable to intrude via the connecting cable.

Over the last years the dynamic range of condenser microphones has been extended more and more, due to modern circuitry designs, and still maintaining low current consumption. Not only the maximum signal levels were increased, but much more the self-noise levels were reduced.

Fig. 23 depicts microphones of this type, with a dynamic range of 131 dB and self-noise level of only 7 dB (A-weighted) in the case of the TLM 103.



Fig. 23 Transformerless dc polarized condenser microphones (TLM 170 R mt and TLM 103, Neumann)

5.8 The RF circuit method

Before low-noise field effect transistors were available, semiconductor technology was applied to condenser microphones in the form of the so-called radio-frequency circuit method, which requires only conventional transistors. With an RF circuit, the microphone capsule operates as an “active transducer” (see Section 2): It controls the frequency or phase of an RF oscillator or represents an impedance in an RF circuit that varies in cadence with the audio frequency.

There are numerous kinds of circuits available for this. In addition to the microphone capsule, all of these microphones also contain an RF oscillator and a demodulation circuit. Only the demodulated audio frequency voltage appears at the microphone output, and the user can therefore seldom tell whether his microphone is using RF or dc polarization.

The most important attribute of the RF circuit is the fact that the capsule capacitance has a relatively low RF impedance. As an example, a capsule capacitance of 50 pF at a frequency of 10 MHz represents only about 300 ohms. The insulation resistances of capsule and circuit are therefore not subjected to such strict requirements as is the case with dc polarization. On the other hand, the electronic components must be of the low-loss type at RF, and the frequency-determining parts, including also the capsule, must be electrically stable, so that the various circuits will not detune each other.

Further features:

No polarizing or bias voltage is required. In principle, low-frequency components down to zero frequency can be transmitted, as long as the capsule can pick them up. This may cause overloading, because low-frequency noise components cannot be electrically filtered out ahead of the demodulator output.

Formerly, microphones were mostly produced with the phase modulation principle:

The quartz-controlled RF oscillator operates at a fixed frequency (about 8 MHz). The demodulation stage resembles approximately the traditional ratio detector. The modulation circuit, which includes the capsule, is tuned exactly to the oscillator frequency (Fig. 24).

When sound waves impinge on the capsule, the phase of the high-frequency current in the demodulator circuit is shifted in accordance with the sound pressure variations, so that the two diodes receive unequal RF voltages, with resultant modulation at the output.

Noise caused by oscillator amplitude fluctuations is nullified by the limiting effect of the ratio detector.

Noise generated by oscillator frequency fluctuations is held to a minimum by the quartz crystal.

Noise originating in the demodulator circuit and in the effective acoustic resistance of the capsule result for the microphone in about the same signal-to-noise ratio as that typical for a dc polarized microphone with FET transistor.

Fig. 24 shows the basic circuit of a condenser microphone using the radio frequency system.

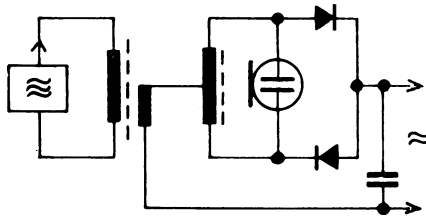


Fig. 24 Basic circuit of a condenser microphone using RF system

Modern RF condenser microphones work with amplitude modulation according to the push-pull bridge circuit principle. An additional back electrode in front of the diaphragm produces a symmetrical transducer. The diaphragm moving between the two back electrodes resembles the center connection of a capacitive potentiometer. The RF voltage is proportional to the diaphragm excursion and produces, after demodulation, an audio signal with extremely low distortion.

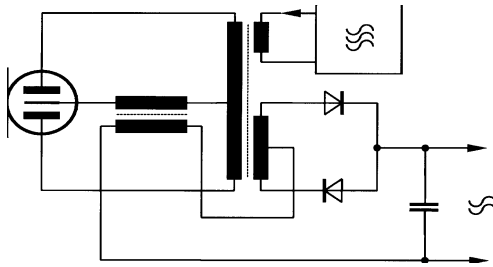


Fig. 25 Basic circuit of an RF microphone with amplitude modulation according to the push-pull bridge circuit principle. (MK 12, Sennheiser)

5.9 Power supplies and connections

Whereas tube-driven condenser microphones always need multi-conductor cables with special conductors for heater and anode voltages, transistorized microphones are usually operated via two-conductor shielded cable.

With microphones for amateur and semi-professional use, one conductor carries the output voltage and the other the dc feed, the common return path being formed by the cable shield. In the professional studio, however, the two AF conductors are required to be at exactly the same potential difference in relation to the reference potential (housing, cable shield, ground); these are balanced lines. Two types of powering are standardized:

5.9.1 A-B powering

According to IEC 1938 (Fig. 26), one AF conductor is connected with the positive pole, the other with the negative pole of the dc voltage source (12 ± 1 V) via two precisely matched 180-ohm resistors.

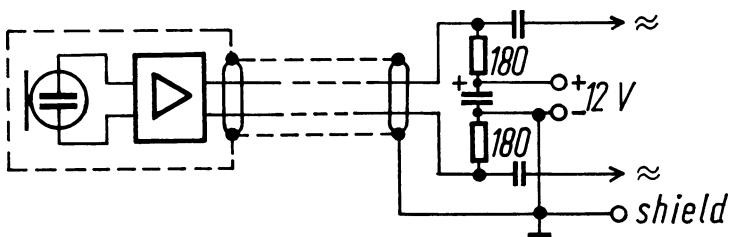


Fig. 26 A-B powering

Isolating capacitors block the dc from the following amplifier stage. The microphone circuit need not be balanced, but may not be electrically connected to the case or the cable shield. Only the midpoint between the two 180-ohm resistors – practically speaking, one of the two poles of the power supply – is connected to the cable shield and the microphone housing, otherwise neither would have any shielding effect, especially in the case of microphones with dc polarization.

If other types of microphones not requiring a dc supply are to be connected, both feed lines must be interrupted.

It is not enough just to cut off the dc voltage, since the other microphones would then be inadmissibly loaded by the two 180-ohm resistors. If the supply current is not switched off, a dynamic microphone or any microphone using an output transformer will deliver distorted output, and ribbon microphones will even sustain damage.

5.9.2 Phantom powering

With phantom powering in accordance with IEC 1938, the dc current is divided, one half being fed to the microphone through each of the two AF conductors and returning to the dc voltage source via the cable shield.

Since both the AF conductors are at the same potential, dynamic and other microphones with balanced and floating output can also be connected to the terminals for phantom-powered microphones without any need to switch off the supply voltage. For the same reason, no blocking capacitors are necessary if the following amplifier likewise has a balanced and floating input, as is usually the case in professional studios.

Fig. 27 shows the circuitry of a condenser microphone designed for phantom powering.

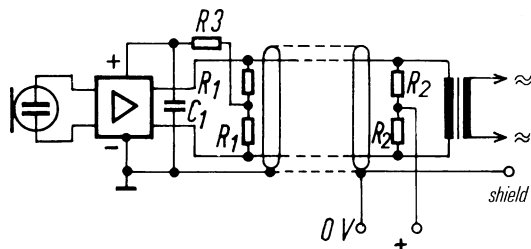


Fig. 27 Phantom powering. The dc could also be fed via the center taps of transformers instead of via the twin resistors R_1 and R_2 . The power supply is connected to the electrical center point formed by the resistors R_2 .

The following values for the phantom powering of microphones are standardized in IEC 1938:

Supply voltage	12 ± 1 V	24 ± 4 V	48 ± 4 V
Supply current	max. 15 mA	max. 10 mA	max. 10 mA
Resistors R_2	680 ohms	1200 ohms	6800 ohms

In order to signify which of the three supply systems is provided, the designations P12, P24 or P48 are recommended.

The cable shield, which carries the dc supply current, may also provide a path for hum or interference caused by ground loops or multiple grounding. In order to prevent this, a high ac resistance R_3 is interpolated in the supply circuit. Together with the capacitance C_1 , this constitutes a filter network for interference voltages superimposed on the supply voltage. In addition, R_3 ensures that only a small fraction of such interference voltages falls off across the twin resistors R_1 and R_2 and – should these not be strictly symmetrical – find ingress into the output circuit of the microphone.

Phantom powering at 48 volts (“P48”) permits the construction of particularly simple and reliable dc polarized condenser microphones, and has therefore achieved the greatest acceptance in professional recording. The relatively high supply voltage can be utilized directly as capsule bias, and the high-value ac resistance referred to above as R_3 can be an ohmic resistor, as part of the supply voltage can be dropped across it.

The capsule bias of dc polarized condenser microphones is also often obtained from an internal dc converter oscillating at a frequency outside the audible range.

The ac resistance R_3 (Fig. 27), which makes the current supply via the cable shield uncritical, is often omitted in microphones designed for 12-volt phantom powering. Some microphone amplifiers are designed to operate at any voltage between 7.5 and 52 volts in phantom powering mode, provided that one or two resistors are appropriately rated.

All condenser microphones may also be operated from batteries. Some are provided for internal batteries. This enables them to be connected to any inputs designed for dynamic microphones without any problems with power supplies. Some microphones with electret capsules even omit the on-off switch for the internal battery, for the current drain can be so low that the battery lasts for over a year anyway. There are a number of studio microphones that may be powered from an external power supply or from internal batteries.

If A-B- or phantom-powered studio microphones are to be connected to equipment not meant for professional use, the input must always be checked to ensure that it is balanced and “floating”. If it is not, an isolating transformer or other dc-blocking devices must be used.

Phantom powered dc polarized condenser microphones are the ones offering the widest choice of models.

A-B powering as per IEC 1938 with its relatively low-value supply resistors affords the audio development engineer less freedom in the design of the circuitry – apart from various other disadvantages – while radio-frequency circuitry offers no possibility, for instance, to modify the directional characteristic electrically by simple means.

6. Microphone types used in recording practice

The point of this section is not to describe specific examples of the wide variety of microphones used in regular practice; manufacturers' catalogs are readily available and provide all necessary details.

Rather, the aim is to follow the foregoing presentation of a number of principles which apply to standard microphones with some details of various special microphones which play an important part in practice.

6.1 Miniature microphones

with diameters ranging from 18 to 22 mm are especially preferred in situations where they should be largely hidden from view – in films and television, for instance.

6.2 Microphones of larger size

enable the capsule to be more effectively protected from structure-borne noise and wind; switches can also be incorporated in the microphone body – for the selection of various polar patterns, e.g., or a multi-stage bass cut-off (impact sound filter) or switchable overload protection to protect the microphone amplifier against overload due to very high sound pressure levels.

6.3 Hand-held and soloist's microphones

are used predominantly for close-talking applications. They allow very high sound pressure levels, and are provided with a pop screen of fine-mesh wire gauze or open-cell foam plastic, which is designed to prevent overloading by explosive consonants, predominantly with condenser microphones. Hand-held and soloist's microphones are available both as condenser and dynamic types.

Although it may appear that pressure transducers would be the first choice for such applications because of their insensitivity to structure-borne noise and lack of proximity effect (Section 3.2.2), special pressure gradient transducers are actually preferred in the majority of cases.

In the case of condenser microphones, their diaphragms are mechanically tauter and in the case of dynamic microphones they are made less compliant, so that, when measured in a plane sound field, their sensitivity has a roll-off of 6 dB/octave to lower frequencies. Low-frequency interference is therefore transmitted only very faintly. At close distances, however, a flat frequency response is obtained, as the proximity effect typical for pressure-gradient microphones (Fig. 5) compensates for that roll-off.

6.4 Noise-suppressing microphones

In microphones intended for speech recording in noisy environments, the diaphragm is made so taut that this sensitivity roll-off starts to become effective at about 1000 Hz for all remoter sound sources, and only close talking at some 2 ... 4 cm distance results in a usable speech frequency response.

These microphones make good speech transmission possible even in very noisy environments, such as motor vehicles and airplanes, and this is mostly further improved by a slight boost in the region of 1 ... 3 kHz (so-called “speech presence”).



Fig. 28 Noise-suppressing dynamic close-talking microphone (MD 425, Sennheiser)

6.5 Flexible or fixed capsule extensions, active capsules

In the case of condenser microphones required to be visually unobtrusive or light-weight – carried on a fishpole boom, for instance – the capsule can be physically separated from the amplifier.

Formerly, a coaxial, low-capacity lead was interposed between the capsule and the amplifier input, which could be made straight or bent. The capaci-

tance of this lead, which, of course, is in parallel with that of the capsule capacitance, is kept low by the fact that the inner conductor has an extremely small diameter. It must also not be flexible, as any change in capacity brought about by a movement of the microphone will be converted to electrical voltages in the same way as diaphragm movements.

To avoid this, for the electrical connection between the amplifier section of the microphone and the capsule, a field effect transistor wired as an impedance converter should be incorporated in the capsule to reduce the high capsule impedance to a value uncritical for long leads. Today's circuit topologies furthermore allow the complete circuitry to be fully miniaturized so as to fit inside the capsule housing, only slightly enlarging it.

6.6 Microphones for room-oriented stereophony

In principle, no special microphones are needed for stereo recordings intended for reproduction through loudspeakers. However, if a two-channel recording is to give a satisfactory sound when played back through monophonic equipment, it should be made by the "intensity stereophony" method: not transit time differences (delay time differences), but intensity differences between the signals carried by the two channels should hold the directional information. Delay differences would result in interference between the signals, and thus to an unsatisfactory mono playback. Such differences are permissible only when one of the two sound components is at least 6 dB weaker than the other.

Delay differences are most easily avoided through the use of so-called "Stereo Microphones". Two directional microphones are arranged closely spaced, and the sound waves impinge on them practically at the same instant. One of the microphones is rotated with respect to the other. The directional characteristics of the two microphones provide the desired intensity differences.

Some dynamic stereo microphones are made up of two identical cardioid microphones in close proximity and at right-angles to each other, mounted on the same stand. In another version, as in most condenser stereo microphones, both capsules are arranged one above the other within a common housing, the upper capsule being rotatable.

Some stereo microphones can also be switched to different polar patterns, and their upper capsule can be turned in either direction relative to the lower (fixed) capsule. This means that the fixed system I can be directed to the left part, and the rotatable system II to the right part of the sound source, regardless of whether the microphone is hanging or standing up-

right (so called **XY-technique**). According to German standards, the left channel should be marked yellow, the right channel red.

Fig. 29 shows a stereo condenser microphone with rotatable upper capsule. The directional patterns can be switched in the microphone itself. There are various other types providing remote control of the polar pattern.



Fig. 29 Stereo microphone with polar pattern switches (USM 69 i, Neumann)

The stereo microphone shown in Fig. 30 is a combination of a shotgun microphone (see Section 6.8) for handling the middle signal and a second microphone system with a figure-8 characteristic at right-angles to the microphone axis, which picks up a side signal (so-called **MS-technique**). The microphone signals are converted to left- and right-hand information in a matrix amplifier. The width of the stereo base can be varied by altering the amplification of the side signal, so that the directional characteristic and the pick-up angle can be remote-controlled despite having fixed microphone systems.



Fig. 30 RSM 191 A-System (Neumann)

For stereo recordings that are not required to also provide high-quality monophonic reproduction and in which delay time (phase) differences between the two channels can be tolerated, two single microphones are set up beside each other at a distance of 17 cm to several meters (so-called **A-B technique**), occasionally together with a middle microphone arranged to pick up both channels; or one of the following set-ups might be selected:

With the **ORTF**-method, two cardioid microphones arranged at a distance of 17 cm between diaphragm centers are turned by 55° to the right and left respectively.

The **OSS**-method according to Jecklin makes use of a 30 cm circular disc lined with absorbent material between two non-directional pressure transducers. As frequency rises, this has the effect of increasing channel separation.

Each of these methods is said to offer certain advantages for specific applications. In general, stereo recordings utilizing both intensity and delay time differences meet with somewhat higher approval.

The user must bear in mind, however, that delay time differences not only impair the quality of mono reproduction, but also make it necessary for the listener to take up **exactly** the same distance from the right and left loudspeaker.

In the case of pure intensity stereophony, this requirement allows more latitude: a delay time difference results here only by reason of the fact that the listener's position is mostly not equidistant from the two loudspeakers. It is constant throughout the entire recording, and can to a certain extent be eliminated by the listener, as it contains no information of any interest.

In conclusion it should be mentioned that additional single microphones are used in most stereo recordings, as spot microphones; their outputs are electrically distributed among the channels, either in equal parts or as required. In order to prevent them from being affected by the localization transmitted by the more remote main microphone or microphones, they may be mixed in only at relatively low levels or via a delay network.

6.7 Microphones for head-oriented stereophony

Recordings made by the head-oriented stereophonic technique use two channels in conjunction with a "dummy head", which is equipped with microphones in place of normal hearing organs. When listening with a high quality stereo headphone, a sound impression is produced which is almost identical to that which the listener would have heard at the location of the dummy head, provided he had kept his head still.

Dummy heads are often used at considerably long distances from the sound source. It has been found that the best results are obtained when they are equalized in such a way that their frequency response is more or less flat in the **diffuse**, and not in the free sound field. Accordingly, the headphone reproduction is most pleasing when the stereo headphone has a flat diffuse-field transmission factor in accordance with IEC 60268-4.

When listening through loudspeakers, the sound impression largely corresponds to that which a conventional stereo microphone at the location of the dummy head would convey, but with a differentiated image of the depth of the room.

In the case of the dummy head depicted in Fig. 31, a special adapter has been inserted between the 4 mm long auditory canals and the condenser microphone assigned to each ear to adapt the small diameter of the auditory canal to the larger one of the microphone. This adapter also contains

acoustic networks for the desired diffuse-field equalization, and renders further equalization measures superfluous.



Fig. 31 Dummy Head (KU 100, Neumann)

With earlier real-head stereo microphones for amateurs, the person making the recording used his own head instead of a dummy head. It has been shown that also the placement of miniature microphones a few millimeters before the entrance to the auditory canal results in acceptable transmission of directional information by headphone playback. The real-head microphone (Fig. 32) consisted of a chin yoke with two extensions which were suspended from the ears and carried two miniature condenser electret microphones, the diaphragms of which faced upwards and, when carried in this way, were held about 10 mm in front of the entrance to the auditory canal. The associated battery power supply unit was carried in the pocket.



Fig. 32 Real-head stereo microphone (MKE 2002, Sennheiser)

6.8 Ultra-directional microphones (shot-gun microphones)

In this type of microphone, the interference effect is utilized over a wide frequency range (to frequencies below 500 Hz, depending on the tube length).

In front of the diaphragm of the microphone is a tube that has an axial slit or numerous openings cut in the wall. Sound impinging on the tube at an angle changes its direction of propagation after entering either the slit or the openings. It is not in phase with the sound components entering at other points and is therefore attenuated (Fig. 33): the path sections a and b are of different length.

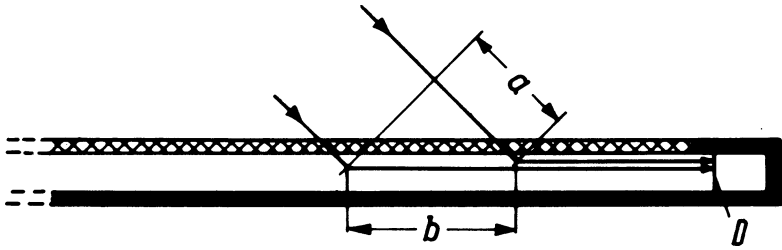


Fig. 33 Operating principle of the single-tube (shot gun) directional microphone ($D = \text{diaphragm}$)

Only for sound impinging parallel to the tube axis the sound components are in phase and are not attenuated by interference. The result is a lobe-shaped directional characteristic.

For the polar diagram to be of similar shape for all frequencies within the response range, the effective tube length must become shorter with rising frequency. To this end the slit is covered with fine gauze, which causes the acoustic flow resistance inside the tube to increase with rising frequency, so that the tube effectively becomes shorter. At the same time the gauze prevents resonances from occurring in the tube.

To compensate for the increasing damping effect at high frequencies, the affected sound components are appropriately boosted by the amplifier. At lower frequencies the tube length is no longer great when compared with the sound wavelength, therefore an approximately lobe-shaped directional characteristic is retained in this range by arranging the microphone system to operate as a pressure gradient transducer with cardioid or hypercardioid characteristic at low frequencies.

Most interference-tube microphones also have perforations at the front of the tube. This has the effect of raising the sensitivity by some 6 dB, especially to sound waves impinging from the front, down to relatively low frequencies, as the result of pressure build-up on the diaphragm (s. Section 3.3).

Interference-tube microphones are almost exclusively designed as condenser microphones.

Fig. 34 shows an interference-tube microphone with a total length of 39.5 cm and a polar diagram as depicted in Fig. 35.



Fig. 34 Single-tube (shotgun) directional microphone (KMR 82 i, Neumann)

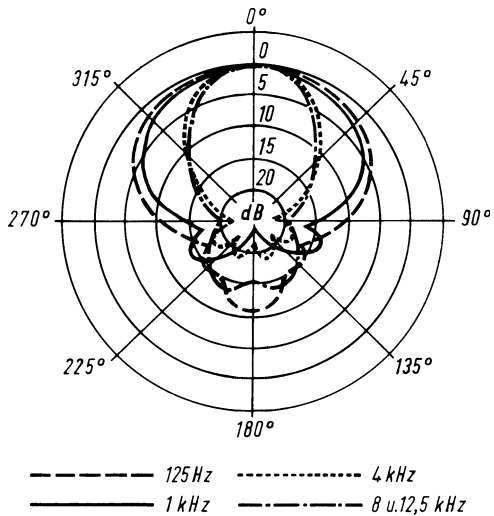


Fig. 35 Polar diagram of the microphone shown in Fig. 34

Since the microphone has to function as a pressure gradient transducer only at **low** frequencies (longer wavelengths), the sound detour to the rear of the diaphragm can be longer than it is with other pressure gradient microphones (cp. Section 3.2.1). This gives rise to wider pressure differences and thus to more powerful driving forces for the diaphragm, and the latter does not have to be so extremely compliant as it must be in microphones which also have to function as pressure gradient transducers at higher frequencies. The microphone is therefore less susceptible to shocks and vibrations, handling noise, etc.

In order to also exploit these advantages for more handy microphones, models were developed with a short interference tube which are especially suitable for recordings in unsettled surroundings, but are also popular as hand, desk or podium microphones. Their lobe pattern is somewhat broader than that of longer interference-tube microphones, but still narrower than a cardioid or hypercardioid, and is occasionally very useful for sound reinforcement.

A microphone of this type is shown in Fig. 36.



Fig. 36 Condenser microphone with short interference tube (KMR 81 i, Neumann)

6.9 Lavalier and clip-on microphones

Singers and speakers who must be free to move around in the course of their performance are often provided with so-called “Lavalier” microphones.

These microphones – mostly pressure transducers – are worn on the chest, suspended from a cord worn round the neck. Their transducer system is protected against solid-borne noise picked up via the case, so that, for instance, noise caused by friction against the wearer’s clothing remains inaudible. The response of the microphone is boosted by some 8 to 10 dB towards the upper end of the frequency scale, as higher-frequency voice components are radiated in the direction of speech, and not towards the chest. In addition, there is a resonance ‘hump’ in the voices of male speakers at about 700 Hz and of female speakers at about 800 Hz, which is the result of sound radiated from the chest cavity. Some Lavalier microphones, equalize this resonance either acoustically or electrically.

Fig. 37 illustrates a dynamic Lavalier microphone.



Fig. 37 Dynamic Lavalier microphone (M 111 N, beyerdynamic)

In television studios, a small “clip-on” microphone is preferred over a Lavalier; it looks like a badge or a piece of jewelry worn on the clothing (on a lapel, for instance), and is hardly noticeable on the TV screen. What is stat-

ed above regarding the frequency response applies likewise to these microphones; however, a resonance 'hump' at 700 or 800 Hz is found only in isolated cases, owing to the different way in which it is worn.

The output of both microphone types is transmitted via a flexible cable and, in the case of condenser microphones also via a small adapter. Very often the link is wireless, being provided by a battery-powered pocket transmitter.

Fig. 38 shows a capacitive lapel microphone, which contains only an integrated field effect transistor and very high impedance resistor, apart from the microphone capsule.



Fig. 38 Capacitive lapel microphone (MKE 2, Sennheiser)

6.10 Wireless microphones

Complete independence of microphone cables is afforded by microphones containing in the case a small transmitter and the batteries needed to power it. Sometimes these are integrated with the microphone itself. A small piece of wire is all that is required for the transmitter antenna. In the case of more recent types operating in the centimeter wave band (500 to 1000 MHz), only a short metal rod serving as antenna is attached to the rear of the microphone or integrated in the microphone housing.

With wireless transmission in enclosed rooms, pronounced standing waves can always be anticipated, especially when using such short waves. Movements of the microphone cause the incoming signal level to fluctuate strongly, and there are many places at which it falls to zero. To compensate for such

fluctuations, frequency modulation is used for transmission, with severe amplitude limitation. In addition, provision is made for diversity reception: an automatic circuit switches the receiver always to the one receiving antenna of several which happens to be delivering the RF signal at the highest voltage at any particular moment. This action is inaudible for the listener.

Newer transmission systems of this kind are provided in addition with a compander circuit to improve the signal-to-noise ratio, with the result that cordless microphones are often used today in place of cable-attached microphones for high-quality applications without any appreciable loss of fidelity.

Fig. 39 depicts a cordless microphone for announcers and vocalists. It can be equipped with electret condenser or dynamic capsules and its transmitter operates at a frequency between 450 and 960 MHz at a power of some 60 mW.



Fig. 39 Cordless electret microphone for announcers and vocalists with integrated antenna (SKM 5000, Sennheiser)

6.11 Boundary-layer microphones

These microphones, also known as PZM (pressure zone microphones), are small condenser pressure transducers which are built flush into a rigid plate so that their diaphragm is (almost) at the level of the surface. The plate of the microphone can be placed on a larger surface, such as the floor, or at-

tached to a wall, etc. This kind of arrangement approximates to some extent the conditions of the “infinite baffle”, which acts as a reflector down to the lowest frequencies.

Whereas in the sound field of a room standing waves invariably develop, and with them frequency- and location-dependent sound pressure maxima and minima, a microphone mounted in a boundary surface is in a sound pressure maximum all the time. As a result, fluctuations in the frequency response of the impinging sound pressure play no part, and the useful voltage is doubled. It should be mentioned that when sound arrives obliquely, the high-frequency components undergo some attenuation through interference, as not all parts of the diaphragm receive the sound waves at the same time. A flat frequency response for sound impinging from the side can only be obtained when the diameter of the diaphragm is less than 5 mm.

Some boundary-layer microphones are produced with very small diaphragm diameters, which enable them to handle sound signals arriving from the side without high-frequency loss. They also transmit indirect sounds that reach the microphone via wall, floor and ceiling reflections with a high degree of fidelity, and these are the sounds that convey to the listener important information on the size and nature of the recording room. Transients and impulse sounds retain their original sound character. The free-field and diffuse-field frequency response of the microphone are identical. The listener has the impression – more than with conventional microphones – of being in the recording room, and can identify remote sources quite well, as would a person actually in the room. Also the positioning of the microphone is much less critical. However, boundary-layer microphones can only pick up acoustical waves in half of a room because they are built into plates.

It must be said, however, that the conventional methods of giving a microphone a desired directional characteristic cannot be applied here. Both the pressure gradient and the interference principle make use of transit time delays, which would nullify the desired effects. Still, some microphones are produced with pressure-gradient capsules mounted close to the ground plate, to attenuate the rear “quarter-space”. The proximity to the ground plate excludes any interference arising between direct sound and ground reflections, as can happen with microphones on a table mount.

Fig. 40 shows a boundary-layer microphone with the associated rigid plate. Its aperture is a mere 12 mm in diameter. A removable gauze grille attenuates air noises. With its non-central position of the capsule on this ground

plate without any axis of symmetry, interferences between direct and reflected sound impinging on the diaphragm are avoided. Reflected components arise at all boundary layers, when the sound waves hit the rims of the ground plate.

Since directional microphones, as indicated in the foregoing, cannot be manufactured on this basis, stereo recordings can be made only by so-called A-B method, using the principle of phase differences.



Fig. 40 Boundary-layer microphone (GFM 132, Neumann)

7. Some criteria for assessing sensitivity and operating characteristics

The information that should be contained in a microphone specification is laid down in the IEC 60268 standard, Part 4. For the relevant part of the information, limits are stated or tolerance curves are shown, to which all microphones purporting to be of “home studio” or “HiFi” standard and marked as such must conform.

These refer to (among other requirements):

the **frequency response**, the **polar pattern** and the **directivity index**, the **total harmonic distortion** at 10 Pa sound pressure and the difference in **sensitivity** of the two channels of stereo microphones.

Stricter measurement standards apply to studio microphones and these are to be found in the standard operating procedures of national radio and TV administrations, or are mutually agreed upon between the user and the manufacturer. When assessing the transducing properties of microphones, there must be considered, in addition to the frequency response, the directional characteristic as well as the harmonic distortion, which for studio quality microphones is expressed as “sound pressure level limit for less than 0.5% total harmonic distortion” and also the weighted self-noise level, which is an important factor (for definition, see Appendix).

In dynamic microphones, the self-noise level is usually determined by the free-field voltage response and the ohmic resistance which can be measured at the microphone output, and which is practically the noise source of the microphone.

For 1 mV/Pa (= 0.1 mV/ μ bar) and 200 ohms, the weighted self-noise level is about 32 dB. Microphones for general use therefore should not have a free-field voltage response lower than that, since otherwise the noise contributed by the following amplifier can no longer be considered negligible.

Excellent condenser microphones achieve values of 24 dB and less. Since they deliver 5 to 10 times the useful output voltage, the noise contribution of the following amplifier may be disregarded. Microphones with almost ideal characteristics – a flat frequency response over the audible range and a directional characteristic which is nearly identical at all frequencies – find their application throughout the audio field. However, there are other microphones which display certain unique acoustical behavior patterns, and these are at times preferred for specific applications.

It is difficult to establish clear reasons for choosing condenser microphones over good moving coil ones solely from their technical specifications. Microphones having identical frequency responses, when reproduced through first rate systems do give distinctly different acoustical results. This is understandable, at least to some extent, if the impulse behavior of the microphone is examined.

Fig. 41 shows the output voltage of two cardioid microphones, placed at a distance of 20 cm in front of a spark gap. A capacitor discharging across the spark gap produces an extremely short pressure impulse. The voltages put out by the two microphones show great differences between them.

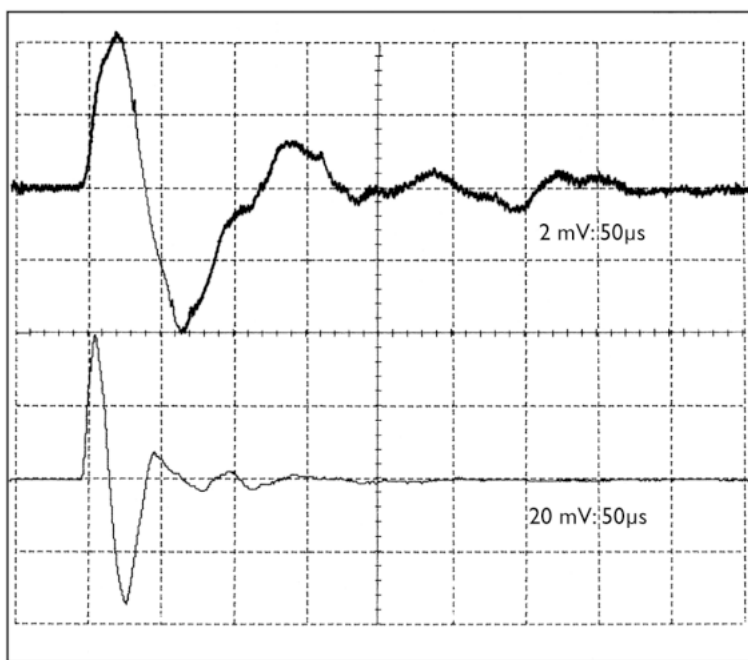


Fig. 41 Output voltages of two cardioid pattern studio microphones when stimulated by an electrical spark discharge (above: moving coil microphone, below: condenser microphone)

Even taking into account the fact that human hearing does not respond to phase shift of individual components in the pulse spectrum, one must note

that the moving coil microphone's output shows damped oscillations occurring within the audible range, which without a doubt, produce sound coloration and may mask the directly following sound signals.

Differences in the acoustical patterns of two seemingly identically specified microphones can be caused by a differing curve for the diffuse-field frequency response. Regrettably, this is omitted from most specifications.

In addition to the parameters determining response quality, the operational characteristics play a great role in judging the performance of a microphone. Enthusiastic HiFi fans are ready to take the greatest pains when using their highly valued microphones, and, when necessary, will operate each microphone through its own dedicated cable. In professional studio environments, the technicians demand microphones that are more rugged and capable of being operated dependably even under changing conditions and over many years. In addition all cables must operate with all microphones in a studio complex and plug into any microphone outlet available. This presupposes use of a uniform powering system.

Although outdoor pickups used to often be done with dynamic microphones recently top quality condenser microphones have taken over. These condenser microphones operate highly reliably when used under field conditions and their operation is not degraded either by high relative humidity or temperature. The high temperatures in motion picture and television studios, when numerous spotlights are on, present no problem to studio condenser microphones.

Appendix

Typical technical Specification

Sensitivity ¹⁾	21 mV/Pa \cong - 33.5 dBV re. 1V/Pa
Max. SPL ²⁾	138 dB SPL
Maximum output level ²⁾	3.5 V \cong 13 dBu
Equivalent self-noise level CCIR 468-3	17.5 dB
Equivalent self-noise level DIN/IEC 651	7 dB-A
S/N ratio CCIR 468-3	76.5 dB
S/N ratio DIN/IEC 651	87 dB
Dynamic range DIN/IEC 651	131 dB

with: 0 dBu \cong	0,775 V
0 dBV =	2.2 dBu
1 Pa \cong	94 dB SPL
20 μ Pa \cong	0 dB SPL

¹⁾ at 1 kHz into 1 kOhm rated load impedance

²⁾ for 0.5 % THD

Some important characteristics of microphones can be illustrated as shown in Fig. 42. Data taken from the technical specification is marked “bold”.

The axes for output level (dBu), output voltage (V), sound pressure level (dB SPL) and sound pressure (Pa) are put into relation with the aid of the free-field sensitivity.

The following steps have to be performed:

- Mark the value for “sensitivity” on the volt axis. This value corresponds to the output at 94 dB sound pressure level.
- Draw a reference line from the above value to the value of “94 dB” on the dB SPL axis.
- Mark the values for “max. SPL” and “equivalent self-noise level” on the dB SPL axis
- Through these values draw lines in parallel to the reference line.

The equivalent values can now be determined in dBu, volts, dB SPL and Pa.

- The S/N ratio is defined as the difference of 94 dB SPL minus equivalent self-noise level.
- The dynamic range is defined as the difference of max. SPL minus equivalent self-noise level DIN/IEC 651.

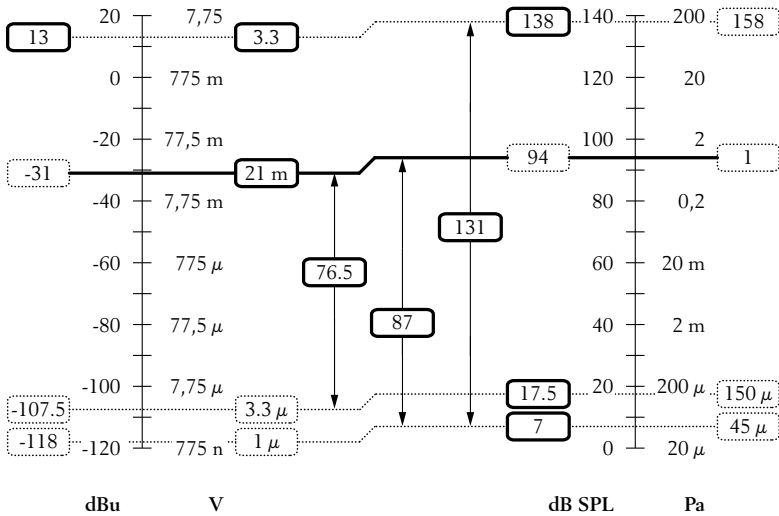


Fig. 42 Illustration of some important technical data

Glossary of terms used for designating microphone characteristics

The **free-field sensitivity** or the **transmission factor** is defined as the magnitude of rms voltage output from a microphone exposed to a sound pressure of 1 Pa. This characteristic is usually given for 1 kHz (1 Pa = 1 Pascal = 10 μ bar).

The **overload sound pressure level** is the max. sound pressure level (SPL) for which the total harmonic distortion reaches a defined value (usually a THD of 0.5%).

The **unweighted self-noise level** is the voltage level output from a microphone in the absence of a sound pressure stimulation. This self-noise, also known as inherent noise, is mainly caused by thermal agitation within the electrical and acoustical resistances.

The **weighted self-noise level** should be measured as a quasi-peak value, using a psophometer via a special auditory weighting network with a frequency curve that takes into account the greater burden imposed by components over 1 kHz. Set into relation with the free-field sensitivity, the **equivalent self-noise level** can be calculated.

The S/N ratio (**signal-to-noise ratio**) is the relationship expressed in dB of the voltage delivered by the microphone at 1 Pa (\cong 94 dB SPL) sound pressure and 1 kHz frequency to its self-noise voltage.

It must be noted that some specifications are not based on the weighted self-noise level as per CCIR 468-3 (quasi-peak value, special filter network), but on the rms value measured via the A-filter of the sound level meter standardized in DIN/IEC 651 (Equivalent SPL and S/N ratio DIN/IEC 651).

This method of measurement yields values for the inherent S/N ratio that may be up to 13 dB higher, and thus appear to be substantially better. This should be borne in mind when comparing specifications.

As is usual in audio engineering, all absolute values can also be expressed in decibels by stating their logarithmic relationship to a reference value, which in this case is defined as "level".

The reference value for sound pressure level is the sound pressure $p_0 = 20 \mu\text{Pa}$ (threshold of hearing at 1 kHz).

The sound pressure level L for a sound pressure p is

$$L = 20 \log \frac{p}{p_0} \text{ dB}$$

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