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MICROPHONES

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MICROPHONES

CONTENTS

1. INTRODUCTION

Microphones have played a very important role in the art of sound broadcasting from its very beginning, and sound production techniques have been mainly conditioned and developed from the properties of microphones available at any given time.

A microphone is a device which changes energy from one form to another, that is, from acoustical energy to electrical energy - it can therefore be termed a transducer.

The main inherent limitations of a microphone stem from its comparison with other 'radiation detecting' devices where its efficiency of conversion is low, and its ability to focus on one preferred signal source is poor. For this reason a microphone must be carefully positioned relatively close to the sound source.

Microphone operational techniques have been evolved over a period of time in order to produce the best possible results for any given type of production. A large number of programs are now stereophonic as opposed to the universal monophonic production. The techniques used for 'stereo' must, however, be capable of giving good compatible monophonic 'reproduction' for single channel listeners.

Some representative high-quality microphones and microphone techniques will be described in this publication .

In the professional broadcasting field, microphones have primarily to be capable of giving the highest fidelity of reproduction over wide-bandwidth systems and therefore are classed in the category of expensive precision instruments. It should be borne in mind that the best microphone in a given application isn't necessarily the most expensive or the latest release. The best ones are those that give the most satisfying 'final' sound after the signal processing is done. Many microphones engineered 40 years ago are still valued today.

Various microphone types are required with non-directional or directional sound 'pick-up' properties and are designed for use in various particular conditions. There are microphones designed for "outside-broadcast" use which have to withstand excessive wind and weather conditions, and others suitable for close talking which are required to give good results in conditions of high background noise. Microphones suitable for boom operation in television studios require a degree of immunity to air movement and mechanical vibration. There are also microphones of which the main features are their small size and unobtrusive appearance, eg. *lavaliers*. These microphones are frequently used within the field of view of television cameras.

When a particular microphone is being considered for use, it should be considered with respect to;

(a) its acoustical mode of operation (Sect.2)

(b) its polar response (Sect.3)

(c) its frequency response (Sect.4)

(d) its electrical operation (Sect.5)

(e) the particular function for which it is required (Sect.9).

Further considerations for choosing the appropriate microphone for a given application are discussed in Section 8 & 10.

Section 12 is intended as a guide to microphone placement. In some applications it is more appropriate to use a multi-microphone technique using cardioid or omnidirectional polar patterns, with perhaps distributed spotter microphones. In other locations an evenly-spaced microphone placement across the stage may be more suitable. It is difficult to lay down hard and fast rules as to microphone placement. There are a multitude of difficulties that may be encountered and in the final analysis it is left to the individual to determine what microphones will be used and the placement which gives the best results.

2. ACOUSTICAL MODE OF OPERATION

A microphone diaphragm is set into motion by the air pressure variations caused by a sound wave. The microphone may have its diaphragm exposed to sound pressure on one or both sides, resulting in two acoustical modes of operation. These modes are, respectively;

(i) *pressure* operated,

(ii) *pressure gradient* operated.

Either or both of these modes may be used to operate the mechanism of a microphone. These modes influence the responsiveness of the microphone to incident sound from any angle, i.e. the *directivity*.

2.1 Pressure Operated Microphone

Fig. 1 shows the basic principle of a pressure operated microphone. A flexibly supported diaphragm is placed across a pressure chamber which is so constructed that the air pressure at the back of the diaphragm is virtually constant. Any change of air pressure at the front of the diaphragm is the result of an incident sound wave which is an alternation of high and low pressures (compressions and rarefactions respectively). Incident sound waves will cause the diaphragm to move from its mean position, inwardly for compressions and outwardly for rarefactions. The excursion of the diaphragm is proportional to the difference in air pressure between the front and rear of the diaphragm. The force on the diaphragm is the product of sound pressure and the area of the diaphragm, and is essentially independent of frequency.

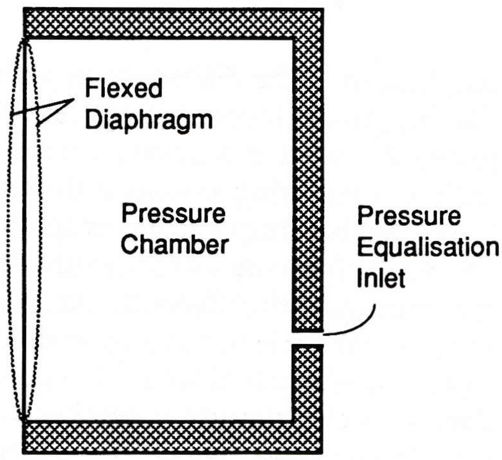


Fig. 1. Pressure operated microphone

A small inlet is provided between the inside of the pressure chamber and the outside air in order that long-term variations in atmospheric pressure do not disturb the mean position of the diaphragm. However, this 'equalisation' inlet as it is called, is not big enough to allow external sound waves to enter the chamber. As all incident sound waves are collected at the front of the diaphragm, this implies that the microphone is equally sensitive to sound waves coming from all directions, ie. non-directional. This is true at low frequencies, however in practice this description begins to fail at high frequencies where the oblique incidence of a high-frequency wave is no longer in phase with the front axial arrival of the same high frequency, so that partial partial cancellation occurs in front of the diaphragm. Also, as the incident sound wavelength approaches the dimensions of the microphone casing, reflection and diffraction effects take place whereby the microphone becomes an obstacle to the arrival of high-frequency sound waves at oblique angles. Both these effects result in reduced electrical output which becomes progressively more pronounced towards the rear of the microphone.

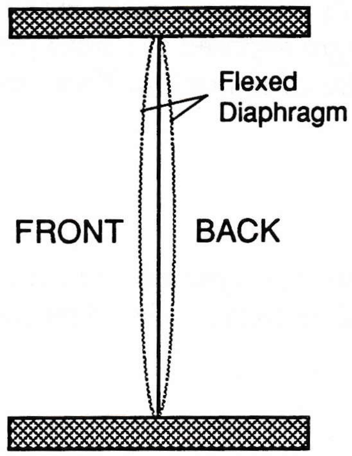


Figure 2. Pressure gradient microphone

2.2 Pressure Gradient Microphone

In a pressure gradient microphone, both sides of the transducer diaphragm are exposed to sound pressure. Fig. 2 illustrates this type of operation. A flexibly supported diaphragm, which may be conical, flat, or a ribbon, is suspended from a housing which is so constructed that, in the absence of sound, the air pressure at the front is always the same as that at the back.

A sound wavefront arriving at the rear of the diaphragm will be out of phase with the same wavefront striking the front of the diaphragm. The reason for this is the difference in path length, and therefore traveling time, between each side of the diaphragm. The difference in path length depends, not only by the thickness of the diaphragm or ribbon, but by the pole pieces and/or the casing. The differences in the phase of the wavefront on each side of the diaphragm causes it to move in accordance with the resulting pressure difference, or the *pressure gradient* (PG). This has an important bearing on the microphone's directional response and will be treated further on. As the frequency of the sound increases, its wavelength becomes shorter relative to the constant path difference and therefore the phase difference increases, causing the PG output of the mic to increase with frequency. When the path difference corresponds to a half wavelength of sound the pressure difference reaches a maximum and thereafter decreases steeplyuntil it reaches a null where the difference in path length equals a wavelength, and thence two wavelengths, etc., as the frequency rises further. The magnitude of the path difference determines the mic's sensitivity and upper usable frequency limit. Acoustic resistance pads may be incorporated in the microphone's design to further reduce the rear pressure and increase the PG output significantly.The PG microphone may be constructed so that pressure operation takes over at frequencies higher than that for maximum PG output to extend its frequency rsponse. To obtain a reasonably flat frequency response, the microphone is usually designed for mass-controlled operation where the system's main mechanical resonance occurs at low frequency and the falling response above resonance counters the rising response with frequency owing to PG operation.

2.3 Interference Principle

This principle utilises an acoustic tube which, attached to a microphone capsule, confers greater directionality. Phase cancellation is used to discriminate against unwanted sound approaching from the side.

The acoustic interference tube, Fig. 3, has a number of transverse slots along its length. Sound coming in at an angle to the longitudinal axis enters the slots and travels the various distances along the tube towards the front of the microphone diaphragm. A number of wavefronts, the number depending on the number of slots, arrive at the front of the microphone diaphragm at different times (hence at different phases) depending on the distance travelled from the respective slot. Sound via the longer path lengths tend to cancel sound via the shorter path lengths when combined in front of the diaphragm. Phase cancellation is proportional to the angle of sound wave incidence to the axis of the tube. For the example shown in Fig.3, the degree of phase cancellation at the diaphragm is greater for sound wave #1 than it is for sound wave #2. Cancellation is severe at middle and particularly at high frequencies. The directivity degenerates to that of the cardioid (refer Sect.3.3) microphone capsule at wavelengths longer than its own physical size. It therefore becomes more sensitive to extraneous low frequency rumble, although some degree of bass cut or slow bass rolloff can be used to reduce this.

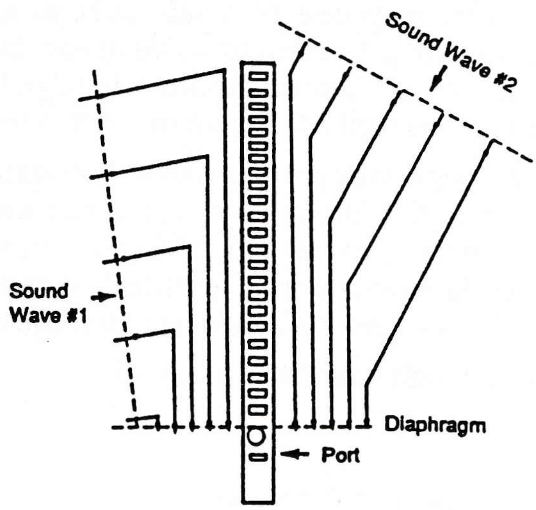


Figure 3. Acoustic Interference Tube

The acoustic interference tube is not directional for reverberation as reverberant sound arrives at the microphone by many different paths and therefore controlled phase cancellation cannot take place.

The longer the acoustic interference tube is, the greater the number of slots it can accommodate, and the higher the microphone's directional efficiency becomes.

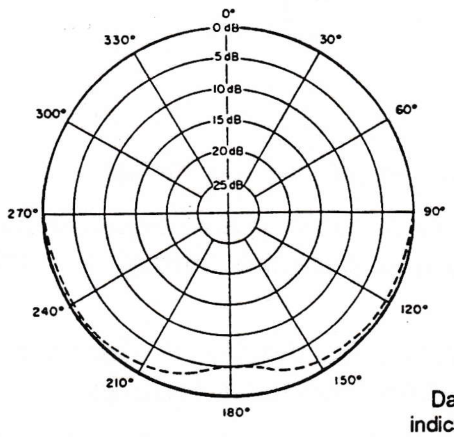
Owing to its length and unidirectional polar response, this mic is known as a gun, shotgun or rifle microphone. Used with a windshield, it is ideal for sporting events, press conferences and boom mounting, where it provides a concentrated sound pickup over a narrow acceptance angle.

3. POLAR RESPONSE

The polar response of a microphone refers to the variation in electrical output corresponding to a sound wave of constant intensity, as the direction of the incident sound is varied. The three dimensional polar response is usually shown diagramatically in two dimensions as shown in Fig's 4 to 7.

The microphone's polar response is measured within an anechoic chamber as a reliable means of eliminating surface reflections which would otherwise influence the measurement, particularly the off-axis response. The measurement is performed by noting the microphone's electrical output while rotating it through 360°, keeping the sound source constant in level and location.

3.1 Omnidirectional Response



Dashed line indicates slightly reduced sensitivity at the rear

Figure 4. Omnidirectional Polar Diagram

Fig.4, shows the *omnidirectional* or 'equal response from all directions' pattern. This pattern is obtained from any *pressure operated* microphone. An ideal response pattern is circular, but in practice it depends upon the size and shape of the microphone itself. There may be some variations from this ideal at high frequencies where the polar response becomes more directional.

An omnidirectional response is advantageous when the ambience is an essential part of the recording or where reverberation or acoustical feedback is no problem.

3.2 Bidirectional Response

For a pressure gradient microphone equally open to sound on both sides of its diaphragm, it can be deduced that the sensitivity to sounds arriving on axis (0° and 180°) will be a maximum owing to a maximum effective path difference between front and rear of its diaphragm, and will have zero sensitivity to sounds arriving at the sides (90° and 270°) where the sound wave is simultaneously incident on each side of the diaphragm (zero effective path difference). Plotting the output for sound sources at various angles to the microphone gives a *bidirectional* or 'figure-eight' polar pattern oriented about the main axis as shown in Fig. 5.

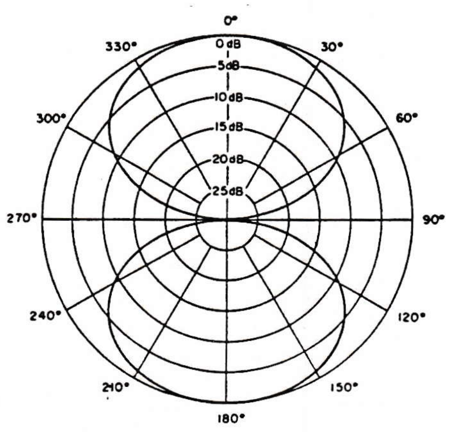


Figure 5. Bidirectional Response

The rear lobe is in antiphase to the front lobe, i.e. a higher pressure arriving at the back of the diaphragm is equivalent to a rarefaction arriving at the front. This type of pattern is used where rejection of 90^0 off-axis sound is important.

3.3 Cardioid Response

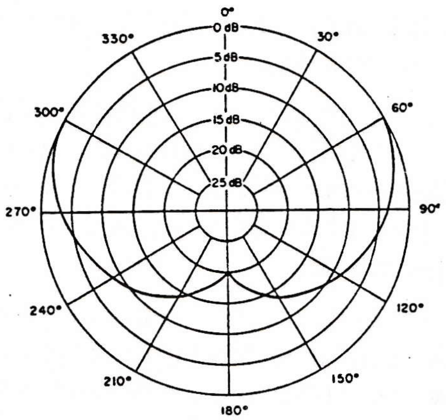
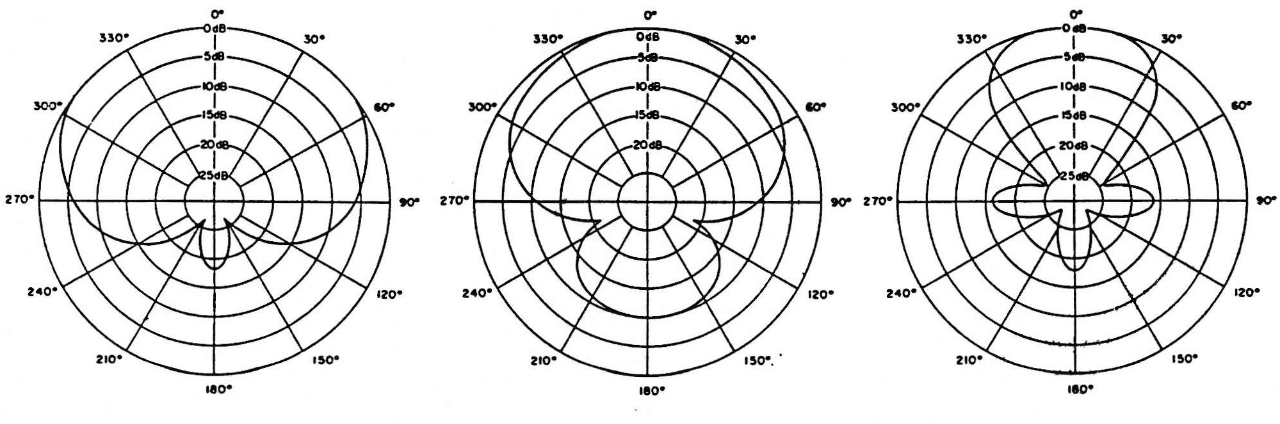


Figure 6. Cardioid Response Pattern

Fig. 6 shows an example of a *unidirectional* or *cardioid* (heart shaped) response, which results from the combined simultaneous action of pressure operation and pressure gradient operation. Addition of the outputs occurs in front of the diaphragm, and cancellation from the omni contribution at the rear.

The cardioid microphone characteristic is most commonly used in both recording and live performances, where ambient noise, room reverberation and feedback from loudspeakers should be suppressed. The interference tube (Sect. 2.3) enables narrower, more unidirectional, cardioid polar patterns to be obtained from a microphone.

These super directional polar patterns, called *supercardioid* and the more narrower *hypercardioid,* are used to pick up more of the wanted sound and less of every other sound. These patterns (Fig. 7(a) & 7(b)) appear between the figure-8 and cardioid patterns in which the rear lobe of the figure-8 becomes progressively smaller and the front lobe more cardioid. The hypercardioid polar pattern has less side sensitivity than the supercardioid and has a larger rear lobe. The shotgun polar pattern has a flattened-out front lobe, with a series of very small rear lobes (Fig. 7(c)).



(a) Supercardioid

(b) Hypercardioid

(c) Unidirectional 'Shotgun'

Figure 7. Superdirectional Polar Patterns

If a cardioid, particularly a supercardioid or hypercardioid, microphone operates close to a small sound source, the wavefront is more spherical and the pressure gradient is increased, especially at low frequencies. This is responsible for the considerable rise in bass response that is obtained from a directional microphone used close to a sound source (eg. singer or speaker), known as the *proximity* effect.

4. FREQUENCY RESPONSE

A microphone is said to have perfect fidelity when the electrical changes it produces follow exactly the sound impulses responsible for these changes.

If it is desired to reproduce the sound generated by a symphony orchestra, which contains practically all the frequencies likely to be encountered in music, the microphone would have to respond equally to all audio variations from 30Hz to 15kHz or even higher. It is important that only the frequencies present in the original sound be reproduced and that no additional frequencies are generated.

The ideal (but not necessarily the most practical) microphone will have a flat frequency response from 20Hz to 20 KHz, possibly with a deviation of no more than $\pm$1dB. In practice it is extremely difficult to design a microphone to have a flat response over its frequency range. It is not always desirable to have a flat response over the full audio spectrum. For example a microphone with extended bass response would emphasise the presence of low frequency rumble from air conditioners and other mechanical equipment, mechanical vibrations in boom operations, traffic noise etc. Also directional microphones exhibit proximity effect which emphasises bass frequencies when working 'close up'. In such situations it may be desirable to introduce bass attenuation. In practice many microphones are designed with boosts and rolloffs in their frequency response to compensate for varying acoustical conditions, environmental problems or creative needs. Switchable filters for bass rolloff are featured on many microphones.

The frequency response of the super cardioid, hypercardioid and shotgun microphone varies depending upon the direction of the sound. The frequency response of the sound picked up from the sides and rear is often not flat. This makes many of them unsuitable in some situations, eg. music recording.

5. MICROPHONE TYPES AND CHARACTERISTICS

Microphones for broadcasting may be divided into two major classes, depending on their principle of operation. The moving coil and ribbon types are representative of one class; condenser and electret the another.

5.1 Moving-Coil Microphone

A moving-coil (dynamic) microphone consists of an insulated coil of fine copper or aluminium wire attached to a light but stiff duralium or plastic diaphragm positioned in the circular gap of a powerful, specially shaped permanent magnet as shown in Fig.8. As the diaphragm is moved by an incident sound wave, an electromagnetic principle applies whereby the motion of a conductor (the coil attached to the diaphragm) in a magnetic field (the static magnetic field within the gap of the magnet) causes an emf to be generated in the conductor and current to flow in the external circuit of the conductor.

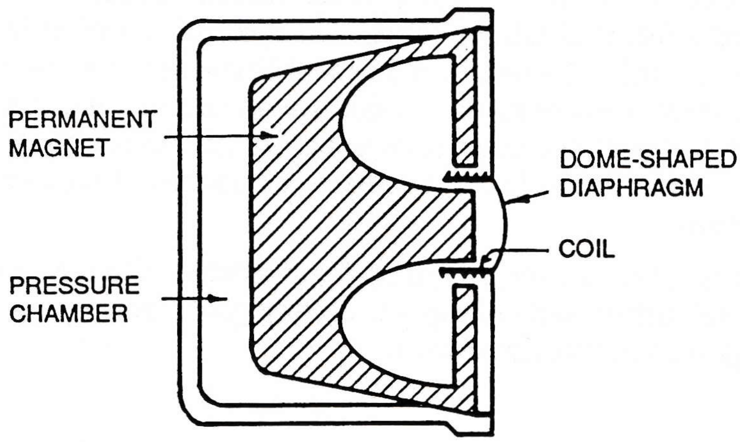


Figure 8. Moving-coil microphone

The induced current is an electrical representation of the incident sound wave.

The magnitude of the induced emf is given by

$$E = Blu$$

where B is the flux density
l is the effective length of the coil in the gap
u is the diaphragm velocity

For maximum sensitivity (voltage output for a given acoustic pressure) the mic design requires B and l to be as large as possible. Even so, the sensitivity is relatively low compared to condenser microphones, but this is not a serious limitation for the applications in which they are commonly used, namely hand-held for vocals, interviews, lavalier and close balance between percussion instruments.

Irregularities in the frequency response are mainly caused by resonant effects. The main resonant peak results from the diaphragm and coil system and occurs at mid frequency, eg. 800Hz. There are other subsidiary coupled resonant circuits involving the combination of compliance and mass of air in confined spaces, the diaphragm surround and other mechanical structures. Built-in acoustic and mechanical damping in the form of acoustic-resistance elements, ie. slots, holes, air passages, trapped air spaces, or a ring of felt or porous metal, is used to smooth frequency response anomolies at the resonant points. The acoustic resistance element used to control the main resonance at mid frequency provides a 'resistive' control over the greater part of the frequency range. This resistance control also makes moving-coil microphones less susceptable to wind and mechanical interference. As acoustic damping reduces sensitivity, a compromise between damping and sensitivity must be met in the design stage.

Moving-coil microphones may be either pressure operated(omnidirectional) or pressure gradient operated (directional). Units capable of both pressure and pressure gradient operation are common. To achieve this, the inside of the pressure chamber is not completely isolated from sound pressure variations in the air but is connected to the outside by a series of holes in the casing. These holes are of such a size and so positioned that the desired effect is obtained. These holes constitute what is called an 'acoustic phase-shifting network'. The degree of phase shift can be varied on microphones which feature an adjustable shutter to allow a controlled amount of pressure operation, thus allowing the directional characteristics to be varied. Microphones whose directional characteristics are the result of acoustic phase shifts within it are sometimes called *phase shift* microphones.

Moving-coil microphones are highly rugged and reliable. They are reasonably insensitive to environmental factors and are capable of very good sonic characteristics. Therefore they find many uses in broadcast work.

5.2 Condenser Microphone

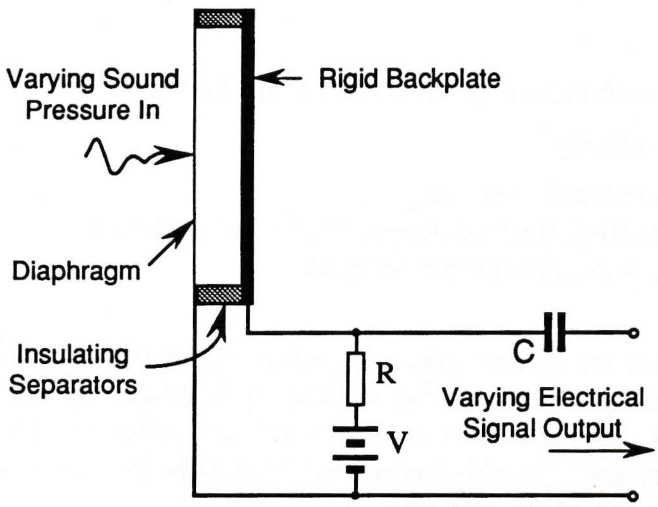


Figure 9. Condenser Microphone

This microphone has enjoyed great popularity in professional fields, partly because of the good signal-to-noise ratios obtained, its wide frequency response and relatively small size. It operates according to the electrostatic principle rather than the electromagnetic principle by which dynamic microphones function.

The condenser microphone, shown in Fig. 9, consists of a thin metallic or metallised diaphragm, stretched and supported close to, but insulated from, a rigid metallic back plate.

The diaphragm and back plate form the plates of a capacitor (condenser). If a polarising voltage is applied through resistor "R", the condenser will charge up until the potential across it is equal to the applied voltage. When it is fully charged the relation is:

$V = Q/C$ where V is the applied polarising voltage

Q is the charge on the condenser

C is the capacitance of the condenser.

With changes in the relative position of the diaphragm with respect to the fixed plate, which occurs when a sound wave impinges upon the diaphragm, the value of 'C' varies. If the time constant of the circuit C x R is long with respect to the time taken for C to change in value (that is in practice, if the product CR is much longer than the period for the lowest frequency sound wave which is to be reproduced), the value of Q will remain unchanged during this period and hence the value of V will change.

This change is V, written as ΔV, will be equal to $Q/\Delta C$ and is the signal output. This small voltage is often amplified by an inbuilt amplifier with high input impedance (necessary because of the small value of the transducer's source capacitance) and a low output impedance (to prevent high-frequency attenuation in the microphone cable).

The diaphragm of a capacitor microphone is significantly lighter and more sensitive to sound pressure variations than the diaphragm/moving coil system of the moving coil microphone.

Condenser microphones may be pressure operated, or partly pressure, partly pressure gradient operated, when an 'acoustic phase-shifting' network, of the type previously described, is used. This type of microphone is of particularly high quality and finds extensive use, especially for orchestral work. The ease with which it is possible to control the polar-response pattern of certain condenser microphones, especially dual-element types (Sect.6.2), extends their range of usefulness.

5.3 Electret Condenser Microphone

These are essentially very similar to conventional capacitor microphones but require no polarising voltage, as they have a permanent charge on the capacitor element. The electret capacitor element is charged by the manufacturer during the microphone construction process. The charge can be 100V or more but loses much of this in use. Consequently the capacitor plates are closer than in the standard condenser microphone, causing the electret dynamic range to be the lesser of the two types. Sensitivity suffers as the permanent charge reduces over time.

The amplifier (impedance converter) inside the microphone case can be energised by a small battery hence the electret needs no external power supply. Electret microphones have a uniform frequency response and good transient response, therefore they are suitable for high quality recordings. They can also be made smaller and more inconspicuous than conventional capacitor microphones.

5.4 The Ribbon Microphone

The ribbon microphone is a variation on the moving coil microphone. It is a dynamic microphone having a metallic membrane which functions directly as a diaphragm and moving conductor. As shown in Fig.10, the ribbon (the equivalent of the 'coil') consists essentially of a flat or corrugated piece of aluminium alloy suspended in a strong magnetic field between two pole pieces. The movement of the ribbon in the magnetic field produces a voltage corresponding to amplitude changes of the incident sound wave causing the movement.

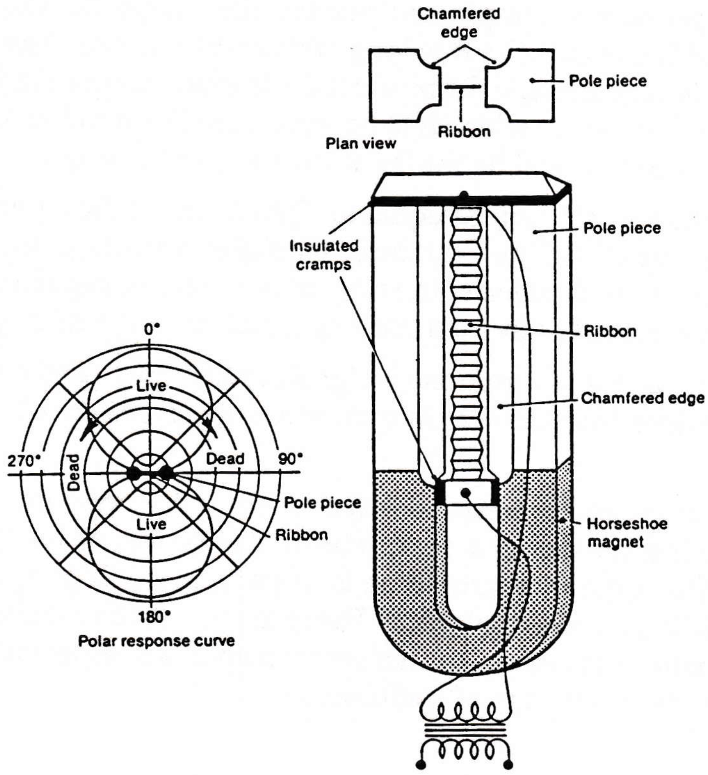


Figure 10. Ribbon Microphone

A matching transformer is built into the case of the microphone in order to step up the small output voltage of the ribbon and to raise the very low ribbon impedance to that of the moving coil to more closely match an amplifier's input impedance.

The ribbon microphone in its standard form is an example of a pressure gradient operated type, i.e. it is open to sound on both sides of the ribbon diaphragm. It is also possible to feed one side of the ribbon through an 'acoustic phase-shifting network' (Sect.5.1). By means of an adjustable shutter, a controlled amount of pressure operation is allowed, hence the polar response may be continuously varied from the standard figure-8 to a cardioid pattern.

The main advantage of the ribbon over the moving coil is that the extremely low mass of a ribbon diaphragm enables it to respond to transients faster than the heavier moving coil diaphragm . Early ribbon microphonse tended to be bulkier than many other types owing to the size of the magnet system and did not stand up well against the demands shortcomings and therefore they are becoming more accepted.

One face of the ribbon is open to sound waves, and the opposite face is connected, by means of a tube, to a folded, acoustically damped, pipe. This pipe is contained within the centre section of the microphone case. The connecting tube is slotted, directly behind the ribbon, and an adjustable vane is provided so that the effective area of the slot may be varied. This permits the directional characteristic of the microphone to be changed to suit particular program requirements. The adjustable vane is moved by means of a slotted shaft and click stops permit the vane to be accurately positioned.

When the shunting inductance is switched into circuit, the low frequencies are attenuated. This attenuation is desirable when the microphone is operating in the pressure gradient mode and is being used close to the sound source. Under these conditions the distance between the sound source and the microphone influences the pressure gradient. With close spacing the pressure gradient is considerably increased, boosting bass frequencies.

6. MICROPHONE VARIATIONS

6.1 Pressure Zone Microphone

It has been found in practice that conventional microphones have difficulty in satisfactorily combining signals received on a direct path between the sound source and the microphone, and those received from a reflected path. Most of the problems are associated with the reflections from a primary boundary such as a wall, floor, table, and so on. The distance between the conventional microphone and the boundary can be quite large. This can lead to anomalies in frequency response, which becomes significant at high frequencies since the relative phase difference between the direct and reflected sound wave increases as the wavelength becomes shorter. Alternating constructive and destructive interference occurs between the direct and reflected sound waves. The cancellation nulls appear at the frequency for which the distance from the surface is 1/4 wavelength and its harmonics. The resulting distortions resemble the teeth of a comb when displayed graphically, as Fig.11 shows.

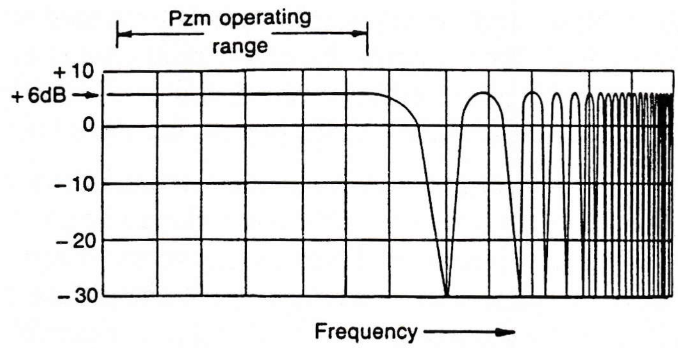


Figure 11. Comb Filter Distortions

The principle behind the pressure zone microphone (PZM), Fig.12, is that direct and reflected sounds approaching a boundary surface arrive in phase within a thin zone at the surface. If a microphone is placed within this zone, the direct and reflected signals reinforce each other and are treated as one signal with a smooth frequency response, even at high frequencies. To ensure adequate low frequency response, the plate is usually placed on the floor or table in the acoustic environment, effectively increasing the area of the plate (at least 2 square meters works well). This is necessary because low frequency wavelengths will exceed the dimensions of the metal plate of the. On the other hand, standing waves can occur when the wavelength of the incoming signal approaches the dimensions of the transducer. This would lead to a high frequency roll-off.

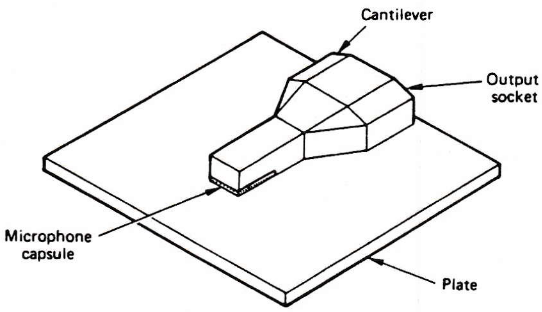


Figure 12. PZM Microphone

Although there are some cardioid PZMs available, the PZM transducer is predominantly a pressure calibrated electret capsule, mounted so that it faces the boundary and lies within the pressure zone. It has a hemispherical polar pattern, with the plate of the microphone defining the equator of the hemisphere. Therefore the quality of the pickup is not altered by moving sound sources.

6.2 Dual-Element Microphone

The problem with single-element moving coil pressure-gradient operated microphones is that they become progressively more directional at high frequencies due to the obstruction of the casing. Dual element (or two-way), single pattern dynamic microphones overcome this inherent defect. The audio frequencies are split between two transducers, one smaller than the other. The smaller transducer is designed to handle the higher frequencies and provide a cardioid pickup with a relatively small acoustic phase correction network. The large transducer handles the lower frequencies with a larger arrangement of phase and frequency correcting elements in the tubular casing behind. The apertures in the tubular casing are set well back in the casing as the sound wavelengths handled are long. This has the effect of improving the output at low frequencies, and incidently of reducing the bass rise when the microphone is used close to a sound source (proximity effect).

The two dynamic transducers are mounted coaxially and each faces in opposite directions, with the high frequency unit placed closer to the front grill and facing forward. Both transducers are coupled to an inductive-capacitive-resistive crossover network (around 500Hz) that is phase corrected.

The polar pattern is maintained over a wide range of frequencies, the on axis frequency response being flat up to 14 kilohertz or more. The transient response approaches that of condenser microphones.

A type of dual diaphragm microphone, frequently encountered, is the twin diaphragm condenser microphone. Each diaphragm is on opposite sides of a common centre plate. The phasing of one of the twin diaphragms can be electronically altered so that the cardioid polar patterns are acoustically combined in phase or out of phase. The in-phase, back-to-back patterns combine to form an omnidirectional polar pattern. The out-of-phase patterns that occur when the the polarity of one diaphragm is reversed, combine to form a bidirectional polar pattern. By decreasing the voltage applied to the rear diaphragm, its sensitivity is decreased, making other intermediate polar patterns possible, including supercardioid. Many modern dual-element condenser microphones have the pattern selector switch for all polar patterns mounted on the body of the microphone.

6.3 Stereo Microphone

These microphones contain two separate condenser capsules (condensers being used for their sound quality and small dimensions) mounted within the one housing, permitting stereo pickup with a single microphone. The small spacing between the capsules (about 3.5cm) guarantees negligible time differences between the two outputs. The upper microphone capsule can be rotated through 180° to provide any offset angle desired.

6.4 Hand-Held Microphone

This microphone must be robust as they can be easily dropped or knocked. When used at very close range, with the axis of the microphone pointing towards the mouth, breath noises, heavy sibilants, and the 'pops' of p's, b's and t's are accentuated, especially if a cardioid pattern is used. Handling noise and the proximity effect of close use are other problems of hand-held mics which need to be addressed.

The susceptibility of a hand-held microphone to mechanical shock and handling noise is minimised by design, eg. the transducer may be isolated from the body by supporting it in an elastic suspension system. They are usually provided with close-talking windshields to reduce pop noise and also built-in bass filters to reduce proximity effect. A filter may also be added to reduce sibilance.

As the microphone is carried around by the user, the problem of 'howlback' (or howlround) will arise if the microphone is inadvertently oriented so that it picks up the floor monitor or other sources of direct loudspeaker sound. Howlback causes an amplifier in the mic/loudspeaker chain to break into oscillation. This situation is also possible with microphones fixed to a stand or boom if they are pointed in the direction of the floor monitor. Howlback is eliminated by using a cardioid pattern microphone with the 'dead side' pointing towards the loudspeaker.

6.5 Lavalier and Clip Microphone

The *lavalier,* or neck microphone (Fig.13), is designed to be suspended from the neck by a string. The more common *clip* microphone is designed to attach to clothing by a clip. The two types of microphones are functionally comparable; both usually have an omnidirectional pattern and both require external equalisation if it is not built in. An omni pattern is useful when the wearer wants unwavering pickup of ambient sound as he moves around, however, more directional patterns are now available for the situation where predominant pickup of the wearer's voice is desired.

Being widely used in television conference and interview shows, they are very useful mics but they have a number of specific problems. They are susceptible to pickup of clothing and cable rustling noises. If they are not equalised they also suffer from 'chest dips' and hollow sound.

The curve of Fig.14, curve A, shows a peak of about 700 Hz caused by chest resonances. The droop at frequencies above 2 KHz is caused by the microphone not picking up the beamed high frequencies from the mouth sufficiently well. Curve B is the inverse of curve A and represents the correction required to compensate for chest resonance and high frequency falloff.

The lavalier and clip microphones may be either condenser or dynamic types, the electret condenser providing the smallest (most inconspicuous) size obtainable.

(M 111 N. Beyer Dynamic)

Figure 13. Lavalier Microphone

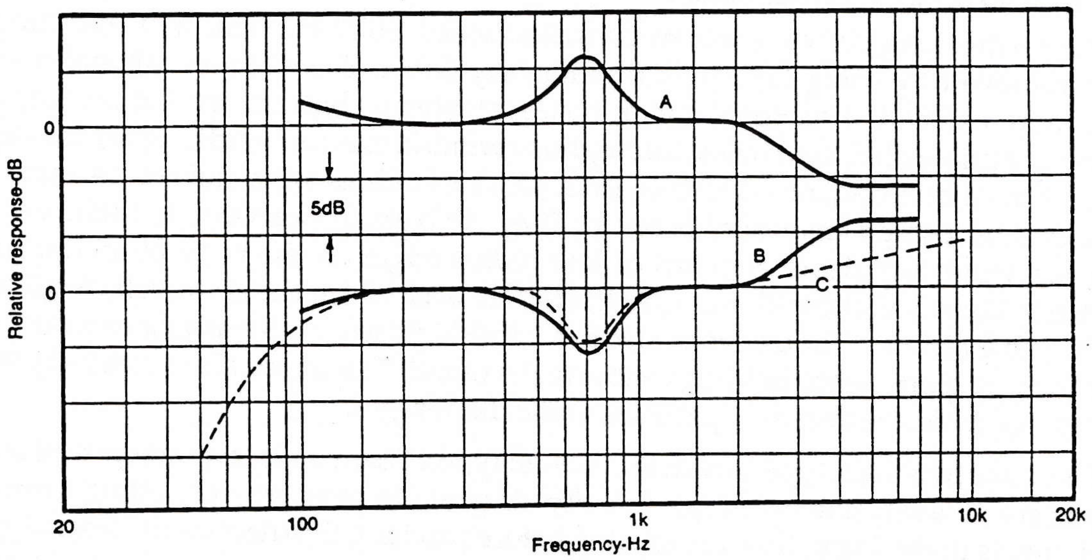


Figure 14. Frequency Response of Lavalier and Clip Microphone

6.6 Radio Microphone

Radio microphones are used where complete freedom of the trailing microphone cable is needed. The pickup unit consists of the microphone (eg. hand held, lavalier, etc) and a small, low powered (battery) operated, frequency modulated transmitter worn by the user, or an integrated mic and transmitter combination. A sensitive receiver picks up the VHF or UHF frequency modulated signal sent out by the transmitter. The receiver may be of the rack-mounted base station variety or small portable types for use with ENG or EFP. The choice of frequency depends on the presence of interfering signals in the vicinity. Interference from car ignition, radar, neon or flourescent lights, and other electrical interference is less in the UHF region. Several radio microphones may be used at once, each tuned to a different frequency.

Signal processing in the form of companding (compression and expansion) is used to reduce noise and increase the dynamic range. Compression and limiting is used to prevent overload of the transmitter.

A problem with radio microphones is that there are nulls in reception as the one wearing the microphone transmitter moves about ; the result of standing wave conditions within the room. This problem can be overcome by using the principle of diversity ; a receiver with two or more separated antennas. The antenna with the highest output is selected. Diversity reception may consist of a pair of identically tuned receiver circuits per channel of a receiver, each fed from separate, spaced antennas. The circuit having the highest output is selected. Switching between the two circuits is accomplished at high speed by an electronic monitoring system (perhaps a microprocessor switching system).

6.7 Microphone Windshields

Most of the energy in wind noise is at low frequency. Subsonic wind noise can effectively be removed by using high-pass filtering, say 60Hz. The effect of air noise is more pronounced with directional microphones, causing turbulence around the diaphragm and the phase-shift vent holes. Microphone windshields (or windscreens) are supplied to fit most microphones. The best examples can reduce wind noise by around 20dB, helped by a small amount of bass-cut. They are basically air velocity filters which cut down wind turbulence at sharp or low radius edges on the body of an unprotected microphone. Windshields increase the radius over which the air travels, resulting in a smoother airflow. The user may also fit a windshield to his microphone to reduce breath effects, eg. pops, when holding it close to the mouth.The screen also protects against the entry of moisture and dust particles to the diaphragm.

There are two main types; foam and basket types. Foam is almost transparent to sound pressure waves, however high velocity air gusts loses energy travelling through the channels in the foam, thus dissipating before reaching the diaphragm. Foam types are appropriate for pressure operated microphones, however pressure gradient microphones are best protected when their sound inlets are all located within a basket-type windshield.

Some microphones have a windshield integrated with the casing. In general the more effective windshields have a greater adverse influence on sound quality. The frequency response of the microphone will be impaired, its bass sensitivity will be reduced, and its polar response, if it is directional, will be widened. For this reason it is advisable not to choose a windshield that is more effective than necessary. For fixed vocalist microphone, for example, a separate pop screen may be placed between the vocalist's mouth and the mic to achieve an equivalent result.

7. CONDENSER MIROPHONE POWER SUPPLIES

Unlike other types of microphones which are output devices only, the condenser microphone also requires a DC input voltage for both the internal amplifier and the polarising voltage for the condenser elements.

Condenser microphone transducers operate at such a high input impedance (approx. 10MΩ) that they cannot be connected to a cable without active impedance-matching circuitry. The matching function is accomplished by the microphone amplifier. In professional practice the amplifier's operating current, or powering, and the polarising voltage is supplied through the microphone cable rather than by a battery incorporated within its housing. Internal batteries, especially if used in a high-output microphone, would need rather frequent checking and replacement.

Microphone powering is usually incorporated into studio equipment such as consoles and mixers. Powering is accomplished by two main methods. Each method utilises the standard signal carrying, two-conductor shielded cables, to supply power to the microphone (instead of separate conductors for signal and power).

7.1 Parallel Powering

With parallel powering (also known as 'AB' or 'T' powering) the operating voltage is placed across the two modulation leads of the cable. This powering method, shown in Fig.15, is not often found in studios today.

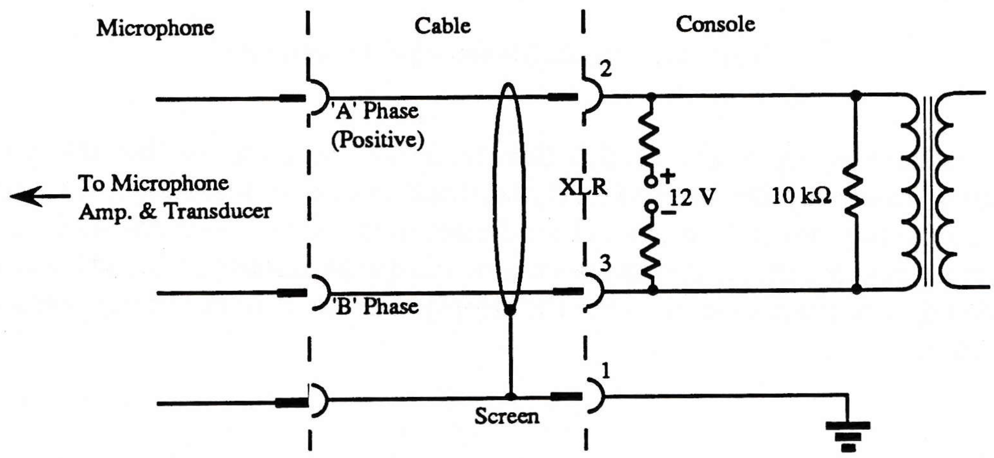


Figure 15. Parallel Powering of Microphones

7.2 Phantom Powering

With this method of microphone powering, the signal output voltage doesn't 'see' the DC input (9-48V), hence the name phantom powering.

With the phantom powering method (Fig.16), the positive pole of the power supply is connected to both modulation leads through a pair of matched resistors (within 0.4%). The current returns through the cable shield. This is the most commonly encountered method of powering condenser microphones in studios.

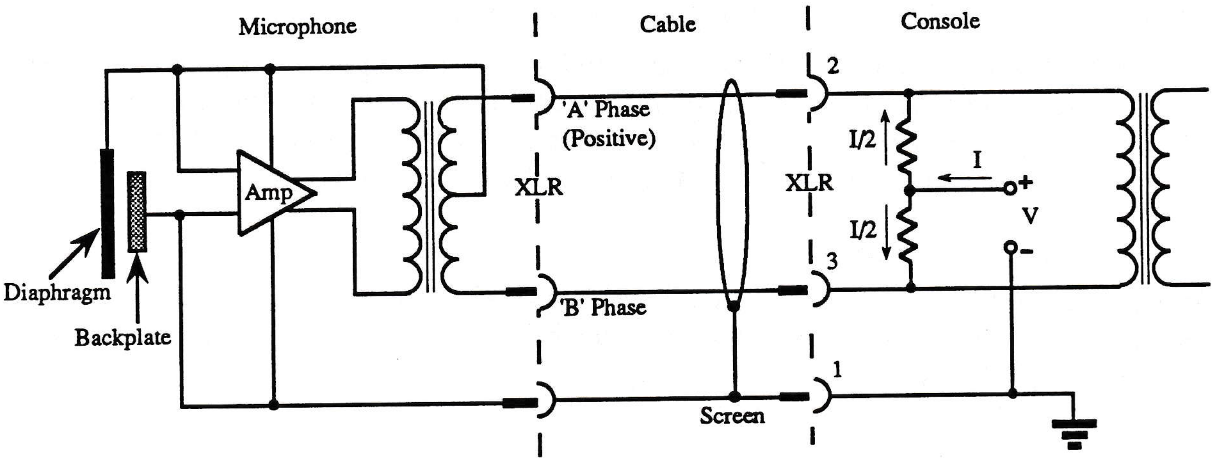


Figure 16. Phantom Powering of Microphones

No DC appears between the modulation leads of the cable so that the particular microphone input may also be used for dynamic microphones with safety. One common power supply may be used for several condenser microphones as there is a high degree of immunity from interference between them. Also, interference induced into the cable shield, being effectively in series with the supply current, is likewise suppressed some 60dB or more.

8. MICROPHONE SELECTION CRITERIA

The selection and placement of microphones for a particular program require an intimate knowledge of both the microphone and the conditions under which it is to be used. Practical experience obtained from working with microphones in different situations assists in selecting the correct microphone.

The important factors which should be considered when selecting and placing a microphone is outlined below.

The factors which govern the choice of microphones are:

(i) The frequency response must be adequate for the type of program to be broadcast.

(ii) The directional response pattern must satisfy the program requirements. For example, background noise must be reduced by the use of sufficiently directional microphones. In some cases, of course, the noise is deliberately included to add realism to the program.

(iii) In TV, the choice of microphones is also dependent on whether the microphone is to be seen in the television picture. Some performances (such as panel discussions) may require the microphone to be seen but in the majority of cases the microphone must be concealed.

(iv) The microphones selected for a particular program should all possess similar characteristics. This is necessary as differences in frequency response, etc., will become noticeable when cross-fading from one microphone to another. This effect can be present even though, individually, the microphones may be quite satisfactory.

(v) Incorrect microphone phasing will also lead to poor frequency response problems and where two out-of-phase microphones are contributing to the program sound simultaneously, the low frequencies will be attenuated. Microphones should be correctly phased when first placed in service and thereafter normal care should ensure that the microphone leads are not reversed.

(vi) Mechanical characteristics; the need to consider whether the microphone is to be hand-held or on a boom, and the problem of wind or air conditioning breezes.

9. MICROPHONES FOR PARTICULAR FUNCTIONS

9.1 Studio Announcers' Microphone

It is an advantage for this type of microphone to be unidirectional as this facility helps in selecting predominately the voice of the announcer from any other noises that may be present in other areas of the studio.

A typical microphone for this purpose is one which incorporates a low frequency roll-off which is about 12dB down at 50Hz.

9.2 Radio Studio Interview Microphone

In the situation where two people are seated on opposite sides of the table, the microphone is required to pick up sounds originating from two directions. Like the Studio announcer's microphone there should be little or no pickup from other directions. The ideal response pattern would therefore be bidirectional (see Fig. 18). A typical microphone used for this purpose is a high quality ribbon type. Care must be used in the placement of a bidirectional microphone to balance out any differences of vocal levels between the two people. This problem is often overcome by using two cardioid microphones back-to-back.

9.3 Radio Studio Drama Production Microphone

At various times, a drama production may require a number of different people to be speaking, and so that they can create a certain quality of sound, there may be a need for considerable space in which they can move when striving for the desired effect.

This type of production generally requires a high quality microphone with omnidirectional properties. Good sensitivity is also a requirement because various speech effects are created some distance from the microphone. In addition, it is desirable for this microphone to have the facility to enable its directional properties to be varied remotely from the control room. This feature may be used to produce many effects.

9.4 Studio Recital Microphone

To obtain better control of balance of sound between a soloist and an accompanist in a studio recital, a separate microphone is used for each person. These have to be high quality microphones to enable faithful reproduction of the high frequencies of the musical sounds and, so that each microphone will pick up mainly one source of sound, a cardioid or a more uni-directional pattern type is required.

9.5 Outside Sports Commentary Microphone

For this type of broadcast a fairly small robust microphone is preferred. It should be light in weight (as the commentator may have to hold it for long periods), and slightly unidirectional (as the commentator is often located in noisy surroundings). It should also be designed for use fairly close to the mouth, and therefore should have bass attenuation to compensate for proximity effect. Hence remote low frequency rumbling noises picked up by the microphone will be attenuated. Commentators often use microphone headsets where the microphone, being attached to headphones via a stem, does not need to be held.

9.6 Orchestral Concert Microphone

Several microphones are usually required when orchestral concerts are broadcast from a large auditorium. The directional properties of the microphones may be varied in order to obtain a balance between the various instruments.

For a general coverage of a small orchestral ensemble, a high quality stereo microphone with a cardioid pattern is frequently used. In locations where orchestral concerts are regularly held, this main microphone is usually suspended. To supplement the pickup of this microphone, several other microphones with a cardioid response (often referred to as spotters) may be used.

9.7 News-ENG-Outside Broadcast Microphone

A highly directional shotgun (hypercardioid or supercardioid) microphone is used. It is mounted in a windshield and often suspended on the end of a boom arm (fishpole).

10. MICROPHONES FOR TELEVISION

The unwanted background noise in a television studio is generally far greater than in a radio studio. This is unavoidable due to the relatively large number of persons in the studio team. The complex and, in some cases, extremely heavy equipment, which is in a state of almost continual activity at the outskirts of the picture area, also contribute to the ambient noise. This noise level can be reduced to an acceptable level in the program sound by use of directional microphones, provided the extraneous studio noises and reverberations are minimal. Microphones with cardioid directional patterns are used extensively. These may be stand mounted (for plays, talks, music, variety programs, etc.) or, in some difficult situations, suspended from battens or from the lighting grid..

Omnidirectional microphones are sometimes employed in the TV studio but are generally limited to the lavalier or clip microphones used for talk programs. In this application the mic is sufficiently close to the sound source to maintain a satisfactory signal to noise ratio.

Where the microphone is to be out of vision it is usually boom mounted. The boom may be either of the semi-adjustable or fully adjustable type. Successful boom operation calls for a high degree of skill on the part of the operator. The boom operator must be familiar with the program in order to anticipate movements which will require microphone repositioning and, when movement becomes necessary, must ensure that lights, studio settings (sets), talent and other members of the studio team are not affected by the manoeuvre.

Boom shadows in pictures must be avoided wherever possible and, in particular, should not fall across the faces of talent. In isolated cases a boom shadow in shot may be unavoidable, but quite acceptable if hidden in the backing and held motionless so as not to invite attention.

In general the boom arm should be positioned so that the microphone is held at approximately 45° above the talent, pointed towards the head and, if possible, positioned directly in front of the speaker's face. The latter is desirable because the higher audio speech frequencies from an average person are confined to a comparatively narrow angle directly in front of their face. An exception to this is where perspective is required from the point of view of the camera, eg. an over-the-shoulder shot. The distance between microphone and talent will vary with the camera angle but in the typical case is from 1 to 2 metres for both television and radio.

In sporting outside broadcasts (OB's), similar considerations apply in both television and radio programs. However, in some television outside broadcasts, the FM radio microphone is a necessity. This applies, for example, in a program situation where the narrator is in vision and required to move through an area or setting which would mitigate against a trailing microphone cable. Such cases can arise in industrial or marine locations, etc.

Television OB's can frequently approach a studio production in complexity, with booms and suspended microphones being employed.

In some types of television programs (e.g. opera and some types of variety programs) it is impossible to obtain satisfactory sound during the performance due to the talent movements and/or camera angles demanded by the program. In these cases prerecorded sound is employed and replayed to the studio floor during rehearsals and performance. The talent mimes the replayed sound and in this way sound and vision are synchronised.

11. CARE OF MICROPHONES

Studio microphones are designed to respond to small amounts of acoustical energy at frequencies up to 15kHz. The moving parts of the system employed are therefore extremely light and sensitive and every care should be taken to ensure that the microphone is not subjected to mechanical shocks. This includes air shocks occasioned by the undesirable practice of blowing on an operative microphone for test purposes.

Cardioid and bidirectional microphones are susceptible to mechanical shock or rumble, as the diaphragm, being light and controlled by the air, tends to stay still while the case moves. In twin-diaphragm microphones there is susceptibility to shock because, unless the diaphragms are connected in parallel so that the induced antiphase rumble voltages cancel out, there is a residual rumble component in any directional mode. A single diaphragm omnidirectional microphone tends to have the stiffness of the air trapped between it and the backplate, the result is that the diaphragm and housing move together and therefore rumble and handling noise are at a much lower level.

Ribbon microphones are especially susceptible to damage by excessive air movement and should be protected by the carrying case when not in use.

12. MICROPHONE PLACEMENT

The ability to place a microphone to obtain the best possible results under a given set of conditions, is acquired by experience. The technique used will depend on the sound source and the acoustics of the studio. These factors will affect the relative intensity of the direct and indirect sound present at the microphone. The microphone is therefore positioned to obtain the most satisfactory sound.

To assist in reducing the number of problems that may occur it is important that the proposed setup be kept simple and use the minimum number of microphones. Some phase distortion will be present when a multimicrophone setup is used, as there are differences in path lengths between the sound source and sound pickup positions. Operational difficulties increase with the use of more sound mixer channels and there is a greater chance of the sound operator making errors.

Additional problems occur in television studios due to the continuous movement of the talent and the acoustic properties of studio settings.

12.1 Placement for the Studio Announcer

One of the simplest of microphone placements is shown in Fig.17 where a single fixed microphone is used as the announcer's microphone. The requirement in this situation is for a clear natural reproduction of the announcer's voice. An important factor, in meeting this requirement is the distance from the microphone to the announcer; it should be such that the acoustic environment does not contribute excessively to the reproduced sound.

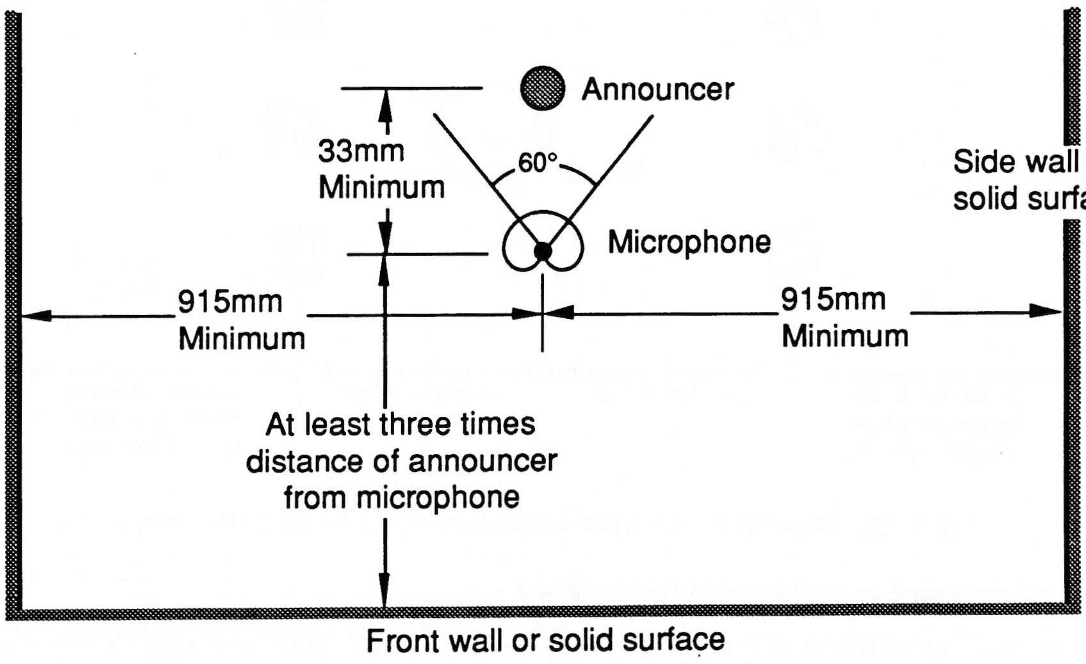


Figure 17. Arrangement of a Cardioid Microphone for the Anouncer

12.2 Placement for Two Person Interview

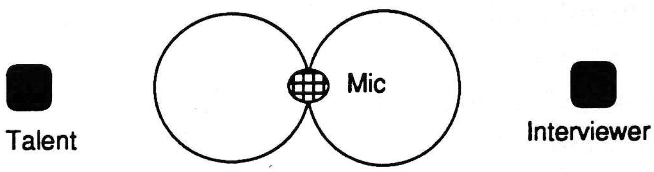


Figure 18. Bidirectional Microphone Placement for a Two-Person Interview

A microphone with a bidirectional pattern may be used for a two person interview or discussion type program. The talent is arranged, as shown in Fig.18, one on each side of the microphone. The required sound balance may be obtained by varying the distance of each person from the microphone.

12.3 Placement for Discussion or Play

As shown in Fig.19, the sound from a discussion or play involving a small group of up to six people may be easily covered by a single microphone that has a bidirectional pattern. Directional properties are such that the performers can themselves contribute balance and perspective to the sound image, as well as produce simple fading effects by moving their individual position relative to microphone placement.

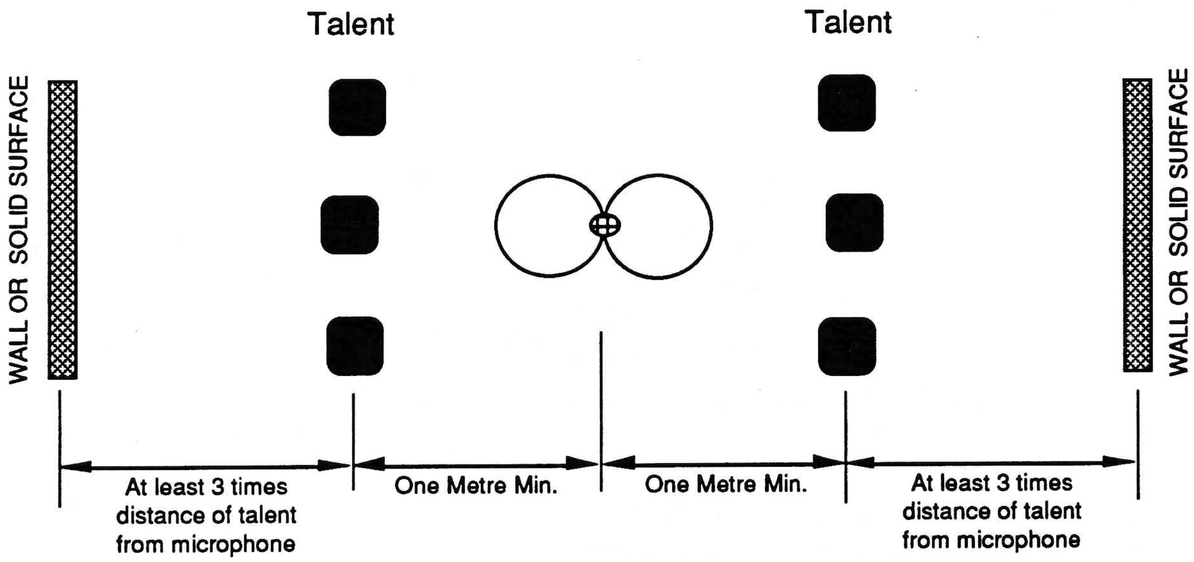


Figure 19. Bidirectional Microphone Placement for a Small Drama Group

12.4 Placement for a Musical Instrument

Good sound balance is extremely important in a musical program. One method that may be used to obtain the desired sound balance is to set up two identical microphones in likely positions and to then make direct listening comparisons. The microphone providing the best sound is left in position while the other microphone is moved to a new position and a further listening comparison test is made. This process is repeated until the desired sound is obtained. If only a single microphone is available the best placement must be determined by performing listening tests with the microphone in different positions, until the optimum sound pickup is obtained. It should be pointed out here that an omnidirectional microphone is generally positioned closer to the instrument than a cardioid, as it produces an in-phase signal in all directions and has no proximity effect.

Usually, only one microphone is required to pick up sound from a solo instrument. The working distance chosen is governed by the required acoustic effect, perspective and sound balance with respect to the other musical instruments.

12.5 Placement for a Small Show Band

A cardioid directional pattern microphone may be used to pick up the sound that originates from a small show band, as shown in Fig.20. The sound balance is dependent on studio acoustics and on an internal balance being maintained within the band.

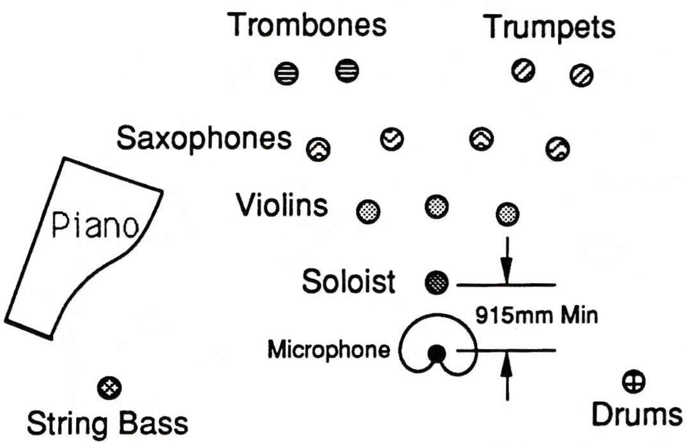


Figure 20. Small Show Band Placement with one Cardioid Pattern Microphone

If greater sound separation is required between sections of the band, then a multi-microphone technique may be used. Separate microphones are positioned so that they are close to the required sound source and each receives sound predominantly from one instrument or section of the band. The electrical signals originating from the different microphones are then mixed to obtain the desired sound balance.

Acoustic screens are sometimes used to isolate the different instrumental sections and to assist in controlling the acoustic properties of a studio. A typical placement of microphones and screens for a show band is shown in Fig.21.

If a stereo effect is required, a pair of microphones may be located behind and above the main pickup unit. The output level of the stereo pair is adjusted until the preferred balance is obtained.

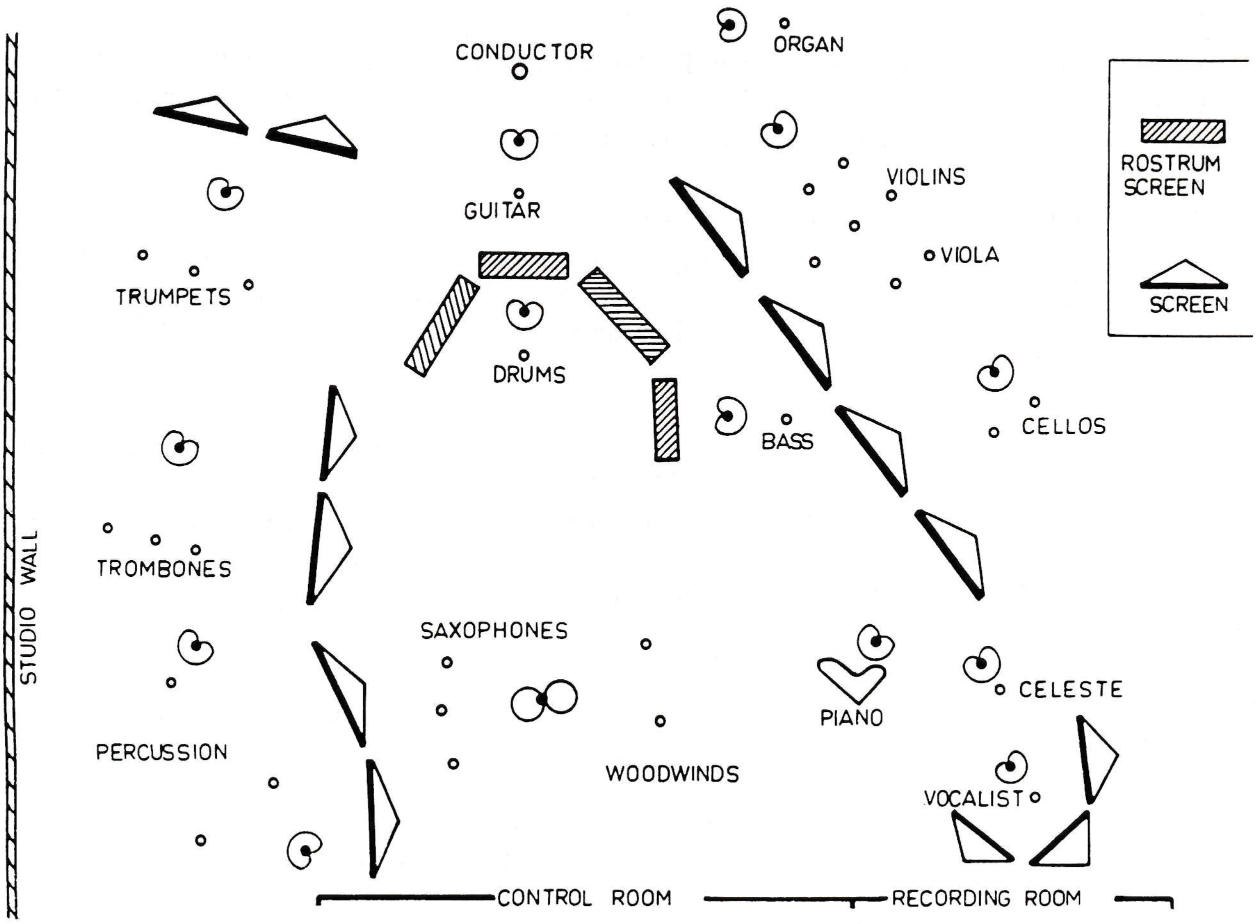


Figure 21. Microphone Placement for a Show Band Performing in a Sound Studio

12.6 Placement for a Symphony Orchestra

There are two common approaches with regard to recording large orchestral works; the single coincident pair microphone technique, and a multi-microphone technique.

A single coincident pair (stereo) microphone is sometimes preferred, as the internal balance that exists within the orchestra can be more easily retained in the reproduced sound when one carefully placed microphone is employed. The positional information of the orchestral sections is also retained when one coincident pair is used. The microphone is suspended in front of (approximately 2 metres) and in line with the centre of the orchestra at an angle of 45° and at a height of approximately 5 metres, varying according to the venue and the actual work being recorded. If too low, the microphone will be over-responsive to the front of the orchestra, but if too high, the orchestra sound will be over-reverberant. The correct distance between the microphone and the orchestra is important to ensure that the instrumental balance is retained.

A vocalist performing with an orchestra will usually take up a position in front of the orchestra and to one side of the conductor. When it is necessary to reinforce the sound originating from the vocalist, a separate unidirectional microphone is used.

Often one or more directional 'spot' microphones are utilised to enhance important orchestral sections. This technique is a hybrid between the coincident-pair and the multimicrophone technique. The outputs of these mono spot microphones are panned in the mix to the same positions in the stereo picture as suggested by the coincident pair. A hybrid stereo microphone placement for a symphony orchestra is shown in Fig. 22(a).

The problem with the above technique is that the natural balance of the musical instruments at the microphone position, due to the many acoustical problems that may be present, is likely to be far from optimum. For this reason large orchestral recordings almost always involve the use of separate microphones for each section of the orchestra, as shown in Fig. 22(b); the main advantage being the creative flexibility to enable the operator to achieve an acceptable balance. For small ensemble works, however, the coincident microphone approach may be entirely satisfactory.

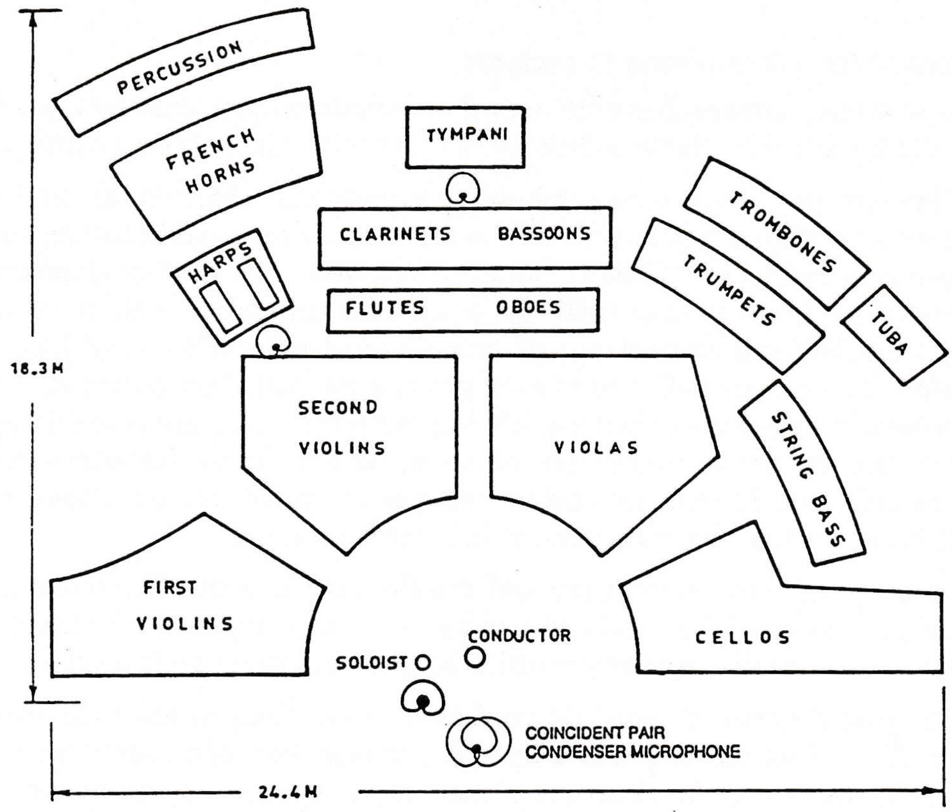


Figure 22 (a). Hybrid Technique of Microphone Placement for a Symphony Orchestra Performing in a Hall

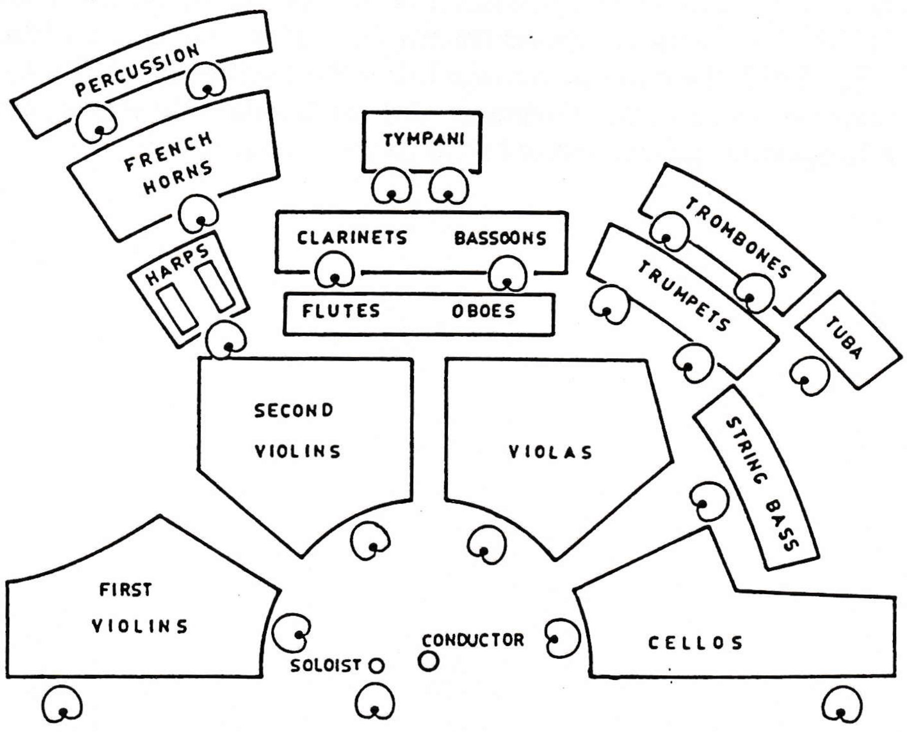


Figure 22 (b). Multimicrophone Placement for a Symphony Orchestra Performing in a Hall

12.7 Placement for Choral Groups

Clarity of diction is the important requirement for the reproduction of sound from a choral group. Two separate, spaced microphones are used. The stereo spread is created by the difference in loudness and arrival time of the voices to each microphone. The closest microphone to one end of the group collects the sound sooner and more loudly than the furthest one. This is known as A-B time intensity stereo. The microphones should be placed far enough away, from the choir, to provide a good overall balance. The distance should not, however, be excessive otherwise the clarity of diction will be affected and in practice a compromise is usually made.

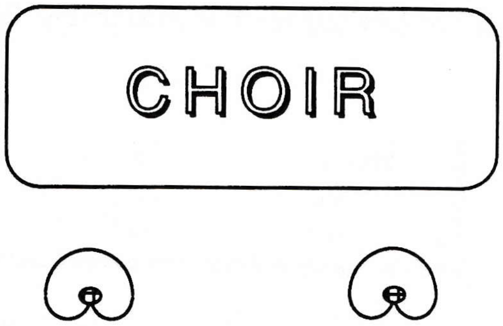


Figure 23. Placement of two cardioid microphones for a choral group

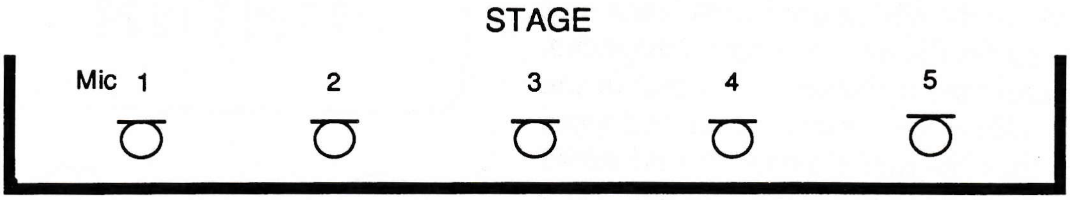
A large choral group presents a larger sound stage and the distance between the two mics need to be widened. The sound reaching the microphones from the middle of the choir lessens and a 'hole in the middle' effect results. This effect can be neutralised by introducing a third microphone located midway between the spaced pair. A portion of the central mics audio is then mixed in with the audio from the left and right side. Cardioid directional pattern microphones may be used as shown in Fig.23.

12.8 Placement for a Televised Opera

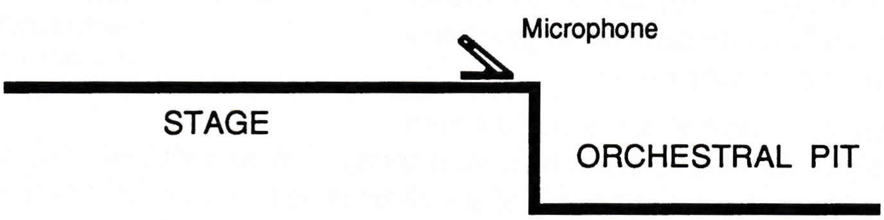
Problems arise here owing to the spatial separation between the on-stage performers and between the singers and orchestra. The problem is to obtain clear diction from mobile singers whilst maintaining sound perspective and a balanced musical effect from the accompaniment. This problem is exacerbated with the larger sets of large operas, usually needing an orchestra of almost symphonic size. The orchestral sound reaching the booms and the singers is likely to be unbalanced, variable and delayed. These problems can be overcome using the following techniques.

12.8.1 Micing the Stage

The microphones should be unobtrusive, i.e. they should not be noticeable to audience or camera. They should be as close to the floor as possible to receive direct sound and avoid reflections from the floor (Fig. 24(b)).



(a) Stage Plan View



(b) Stage Side View

Figure 24. Micing the Stage

The use of cardioid microphones will help with sound separation from the orchestra pit and allow more sound control in the mix.

Depending on stage width, 4 to 6 microphones, positioned along the front edge of the stage and spaced approximately 2 meters apart, should be adequate (Fig.24(a)). The live axis of each microphone should be oriented to their equivalent position in the stereo picture. It is worthwhile to note that for simulcasts, vocal microphones are usually directed closer to centre stage.

For coverage at the back of the stage, Shotgun microphones can be hung from above. The position and number of hanging microphones will depend on the depth and height of the set. For the Sydney Opera House opera theatre, for example, two shotguns, spaced between 8 and 12 meters from the front of the stage and 6 to 13 meters apart, has been found to be effective. Height will again depend on set, but as a guide, 4 to 6 meters should suffice.

12.8.2 Micing the Pit

There is not much option here. A multimicrophone technique is required owing to poor acoustics of most opera pits. Again, cardioid microphones are used to achieve sound separation from the singers so that an effective balance can be obtained between orchestra and voices.

For a very large television opera production it may be a better arrangement to put the orchestra in a separate studio and feed each to the other via loudspeakers. The singers would see the conductor on vision monitors scattered about the set. Hence the orchestra and singers will be effectively much closer together than the usual arrangement would allow, so that sound time lag is not a problem.

Compression can be useful as it allows more microphone gain to be used which increases coverage and helps to maintain vocal presence. The microphones may be subgrouped and a compression ratio of 1:1.5 or 1:2 chosen so that the dynamics are not overly squashed.

12.9 Typical ABC Microphone Placements

Some typical microphone placements that have been used in ABC studios are shown in Fig's 25, 26 and 27.

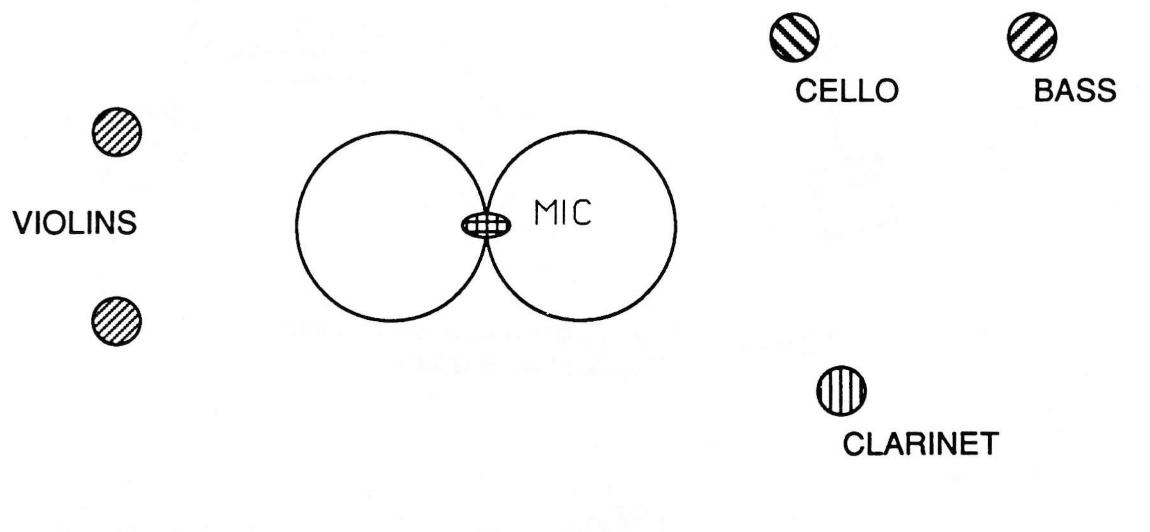


Figure 25. Bidirectional Microphone Placement for a Musical Ensemble

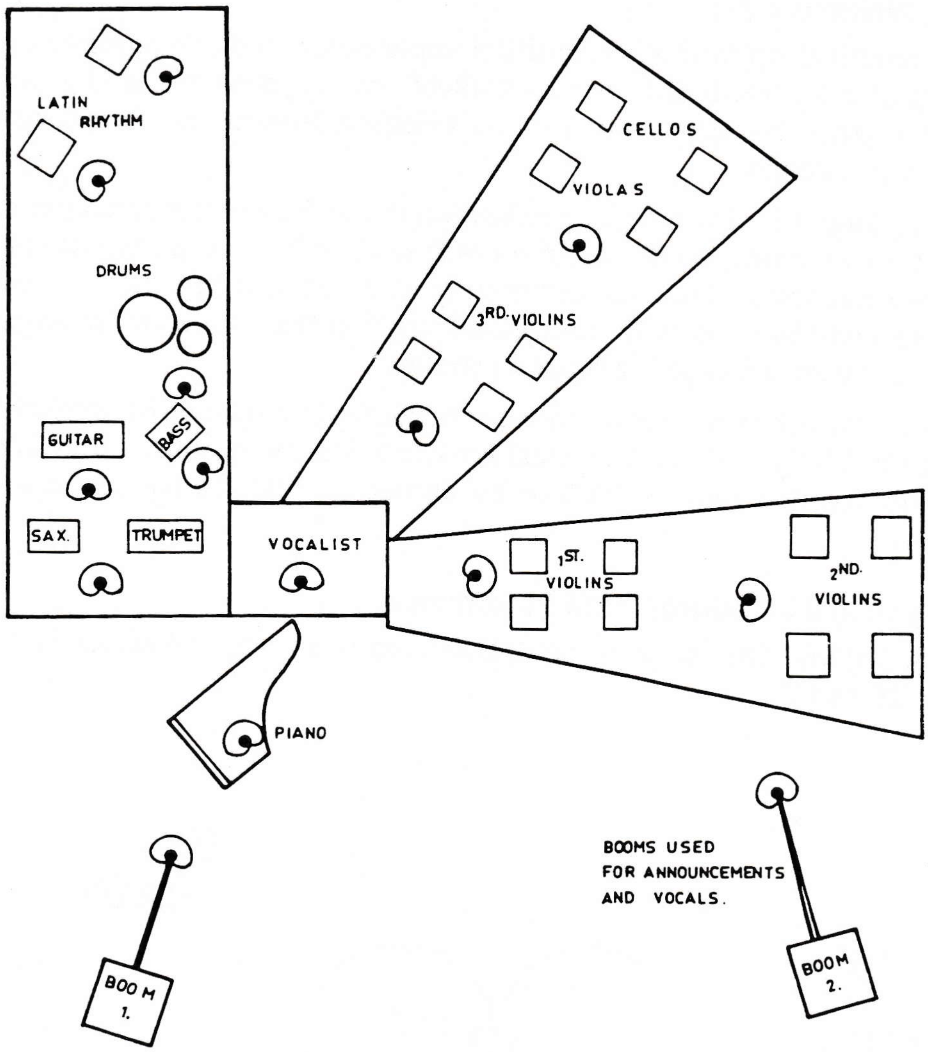
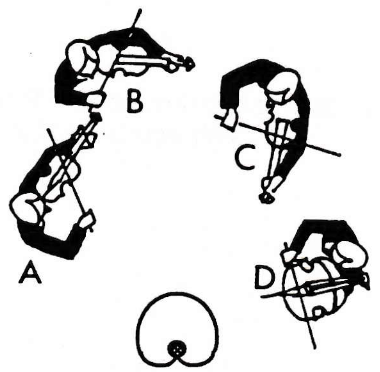


Figure 26. Microphone Placements for a TV Light Entertainment Production



A. First violin

B. second violin

C. Viola

D. Cello

Microphone at 2-3 meters high

Figure 27. Placement of a Cardioid Microphone for a String Quartet

13. GLOSSARY OF TERMS

Acoustic interference tube

A slotted tube attached to a microphone capsule. Sounds entering the slots at an angle to the axis are reduced or eliminated by phase cancellation. It confers highly directional properties to a microphone.

Acoustic phase shifting network

An acoustic barrier to sound which arrives at the rear of a microphone diaphragm. The less restrictive this barrier is to sound, the greater the degree of pressure gradient operation present and the more directional the microphone becomes.

Acoustic screens

Sound absorbent panels used to surround a performer to minimise sound pickup from other musicians.

Balanced and Unbalanced Lines

A balanced line uses two conductors for the signal, surrounded by a braided shield and an outer insulation covering. An unbalanced line carries the signal between one conductor and the shield. The advantage of the balanced line is that electrically induced noise will be picked up equally by both signal-carrying conductors and will cancel out. The screen of the balanced line helps to keep out external electrical fields, as well as provide the return path for the phantom power supplies used with condenser microphoned.

Beaming

The narrow dispersion property of high frequency sound waves.

Boom

A mobile device with telescopic arm to which a microphone is affixed. It enables the microphone position and direction to be varied to follow mobile action.

Close micing

The microphone is positioned close to the lips of a vocalist or to an instrument. Room effects and pickup of background sound sources is minimised.

Coincident microphones

Two identical directional microphones mounted in the same housing, with their live axes angled apart. This arrangement is said to form a coincident pair.

Comb filter distortions

An interference pattern resembling the teeth of a comb when displayed graphically. It is caused by constructive and destructive interaction between direct and reflected sound waves.

Companding

As applied to a radio microphone, companding is a process whereby volume compression of the microphone signal occurs prior to the transmitter and a corresponding expansion of the signal in the receiver. The incidence of signal overload in the transmitter is therefore reduced. The companding process improves the overall system dynamic range.

Compressor

Above a predetermined signal level, a variable gain amplifier (the compressor) ensures that the gain reduces and hence the dynamic range is reduced. Compression ensures that the higher level signals are delivered without causing overload and distortion.

Condenser capsule

A plug-in condenser microphone component containing the diaphragm, but not an amplifier or transformer.

Diaphragm

That part of the microphone which responds to the difference in front-to-rear air pressure. It is the moving element of a microphone.

Directional characteristics (polar response)

Indicates the sensitivity of a microphone to sound pressure from every angle of incidence.

Omnidirectional: Responds equally to sound from all directions.

Bidirectional: Responds equally to sound from its front and rear but not in a plane (Figure-of-8) perpendicular to the live axis.

Cardioid: A heart-shaped response pattern. The pickup is extended in front of the diaphragm but diminished at the rear.

Supercardioid: The polar response intermediate between cardioid and bidirectional. This pattern has a 'cottage loaf' shaped response and is more directional than cardioid however a small rear lobe exists.

Hypercardioid: A cardioid-type pattern but with the off axis response more attenuated. It has a 'cottage loaf' shaped response with a narrow segment of front response and has a larger rear lobe than a supercardioid (but less than a figure-8).

Unidirectional: Cardioid, supercardioid and hypercardioidpolar responses are all essentially unidrectional. The acoustic interference tube of the gun microphone is highly unidirectional in its polar response at mid to high frequency, but cardioid at low frequency.

Directivity factor (sound power concentration)

Specifies how much larger is the power of a point source of sound absorbed by the diaphragm of a directional microphone than is absorbed by an omnidirectional microphone with the same no load sensitivity and at the same distance. An omnidirectional microphone has a directivity of 1, a cardioid 3, a hypercardioid 4, and a shotgun > 4.

Directivity index

Specifies how much weaker the sound pickup is at a specific angle relative to the live axis.

Distortion factor

Distortion occurs when the electrical output does not accurately follow the diaphragm movement, e.g. the coil of a dynamic microphone may lose magnetic equilibrium if the sound pressure level is very high, or the output of a condenser mic overloads the mic preamp of a mixer or recorder (requiring an external pad on the mixer input). The distortion is nonlinear, i.e. harmonics are created in addition to the basic signal. The ratio of the sum of all harmonics to aggregate signal is referred to as the harmonic distortion factor.

Diversity reception

A reception system employing a number of aerials offering the possibility of selecting the strongest signal from instant to instant.

Dynamic range

The usable region between inherent noise at low level and the onset of distortion at high level is the dynamic range, expressed in decibels.

Electromagnetic principle

Dynamic moving coil and ribbon microphones work by this principle. Suppose a diaphragm with a coil attached is located in a magnetic field generated by a permanent magnet. If sound waves strike the diaphragm, it and the attached coil are set in motion. This movement of the coil in the magnetic field induces a voltage in the coil which corresponds to the soundwaves striking the diaphragm.

End-fire microphone

One in which the main axis of polar response is in line with the axis of the body of the microphone. The diaphragm is oriented at right angles to either axis.

Equalisation

Intentional changes made to an electrical signal, mainly to correct for frequency distortions (imbalances) occuring within an audio system. Particular microphones require equalisation owing to the way they are operated, e.g. lavaliers. Equalisation is necessary to restore a reasonably flat frequency response.

Fading effects

The talent appears to recede in sound as the distance from the microphone increases. For television programs, therefore, the microphone must be placed so that as they move about the positional relationship to the camera and microphone must closely correspond. The microphone must therefore be on a line between talent and camera if sound perspective is to be maintained.

Filters

A network of electronic components (generally passive; resistors, inductors and capacitors) which allows some frequencies to pass but attenuates others, eg. a high-pass filter allows only frequencies above a designed cut-off frequency to pass through. Other filters include low-pass and band-pass filters. The steeper the

attenuation is at the cut-off frequency the more complex (having more components) the filter needs to be.
Many directional microphones have selectable bass rolloff filters to take advantage of proximity effect or to reduce the sensitivity to extraneous low frequency sounds. High frequency filters (low pass filters) with various rates of attenuation may also be provided to control inaudible ultrasonic signals that can cause problems.

Fishpole or fishing rod

A lightweight hand-held microphone boom used to pick up mobile sources of sound. Useful in cramped positions or on rough terrain.

Foldback

A feed of selected sources, such as music, effects or a studio microphone, to a studio loudspeaker for the benefit of the performers.

Free sound field

A sound field in which boundary reflections are negligible over the region of interest.

Helmholtz Resonator

A cavity with a small inlet used to dampen unwanted resonant peaks. The cavity resonates and absorbs energy at frequencies determined by its depth, height and width.

Howlback (or howlround)

A closed circuit which includes the microphone, amplifier, loudspeaker and sound path back to the microphone again. Positive feedback peaking will occur at a frequency at which the electroacoustic circuit is most sensitive. The amplitude is dependent on the initial closed loop gain. If this gain exceeds the system losses then the amplitude will build up into a continuous howl at the peaking frequency.

Hum bucking coil

A compensating coil that is connected in phase opposition to the moving coil of a dynamic microphone. It counteracts the interfering effects of induced voltages from nearby sources of 50Hz magnetic fields. Hum sensitivity is reduced by approximately 25dB.

Hum sensitivity

External hum fields are produced by various sources, including power cables, light installations, electric motors, etc. The microphone's sensitivity to such hum fields is expressed as output voltage per induced field strength (μV/μT, μV/mG). The effective output voltage of a microphone that is generated by a magnetic stray field in the most unfavourable alignment, relative to the RMS value of the flux density of the stray field at the microphone site, is referred to as the magnetic field interference factor.

Impedance

The opposition of a component, such as a microphone, to the flow of alternating current, expressed in ohms. It is a combination of resistance, capacitance and inductance values and therefore is frequency dependent (normally specified for 1000Hz). The input impedance of the amplifier (i.e, the microphone load impedance) should be at least three times higher than the impedance of the microphone for best results (i.e. voltage matching instead of impedance matching). Most professional microphones are low impedance types, with an impedance in the range of approximately 50 to 250 ohms. For professional use, low impedance microphones are preferred over high impedance types as high frequency loss over long microphone cables is less significant.

Impedance converter

The output impedance of condenser microphones is very high. Therefore the use of a connecting cable to an external amplifier needs to be avoided in order to minimise interference signals being picked up by the cable. An amplifier with low output impedance is thus built into the microphone and serves mainly as an impedance converter.

Interference

Electrical interference : Includes hum pickup by the microphone and also electrically induced disturbances in the microphone cable, mainly due to man-made sources of electromagnetic radiation nearby, causing hums, buzzes, snaps, crackles and pops. The advantage of using low output impedance microphones is that electrical interference is minimised. The use of balanced lines are preferred between a microphone and a console as electrically induced noise along the cable can be eliminated.

Acoustic interference at microphone diaphragm : The simultaneous rarefactions and compressions experienced by a microphone diaphragm as increasingly shorter wavelengths of the incoming sound wave arrives at an angle to the live axis. This effect causes a drop in output. Large diaphragms suffer more from this effect. The cure is simply to aim the live axis at the sound source.

Comb-filter effect : A common problem with conventional microphones which results from the combination at the microphone diaphragm of the direct and reflected sound waves, causing constructive and destructive interference. This causes the electrical output to rise and fall at intervals universely proportional to frequency, graphically resembling the teeth of a comb.

Interference transducer

A directional waveguide coupler installed in front of the microphone. This acoustic tube has a large number of sound inlet slots that are covered by an acoustic damping material. For lateral sound incidence, phase cancelling interferences cause sound extinctions at the diaphragm, resulting in a lobe pickup pattern and a directivity factor which increases with frequency.

Limiter

Prevents overdriving or overload by setting a predetermined threshold level beyond which the output signal cannot rise, regardless of how much the input level is increased. By adjusting the threshold, the level at which limiting action occurs is controlled.

Live axis

The axis of greatest sensitivity to sound for a directional microphone.

Microphone

A device for changing sound energy to electrical energy.

Microphone capsule

A plug-in acoustic transducer system, not containing the head amplifier (or transformer).

Microphone operation

Pressure Gradient operation: A method by which sound waves have access to both sides of a microphone diaphragm hence the diaphragm responds to the pressure differences. By dimensioning the sound paths appropriately, cardioid, figure-of-8, supercardioid and hypercardioid polar patterns can be produced.

Pressure operation: A method by which sound waves have access to one side of a microphone diaphragm. The diaphragm responds equally to sounds from all directions and therefore has an omnidirectional polar pattern.

Interference principle: Uses phase cancellation to discriminate against unwanted sound approaching from the side, resulting in highly directional characteristics (see interference transducer).

Microphone types

Moving coil (dynamic): The diaphragm is fitted with a copper coil which is located in the magnetic field of a permanent magnet. The impact of sound waves on the diaphragm sets it and the attached coil in motion. By the law of induction, a voltage is induced in the coil which is proportional to the sound wave pressure variations on the diaphragm.

Ribbon: Also a dynamic type, similar in principle to a moving coil except that the coil is substituted by a narrow strip of foil suspended in a permanent magnetic field. In its basic form, the foil is open to sound on both sides and is therefore pressure gradient operated. A current is induced in the foil as it moves in accordance with the air pressure variations of a sound wave.

Condenser: Works as a variable capacitor in which energy from an external power supply is stored between a diaphragm and a fixed base plate. As the diaphragm moves, the distance apart varies and so the capacitance varies accordingly. The varying current that results constitutes the electrical signal. The high impedance of the transducer is converted by an inbuilt amplifier to the required low impedance output.

Electret : A condenser type which does not require the application of a polarising voltage. Instead, a permanent electric charge is 'frozen' into the diaphragm or backplate electrode during its manufacture. This microphone (or microphone capsule) is small and lightweight.

Microphone variants

Dual element:
(1)Two-way; two diaphragms in one microphone body. One is larger than the otherto handle bass frequencies and is connected to the smaller via a crossover network. These microphones have wide frequency response, almost no proximity effect and have a frequency-independent cardioid polar pattern.

(2) Twin diaphragm; two identical diaphragms of a capacitor microphone with a common fixed centre backplate. The sensitivity of one of the diaphragms is altered by varying the DC polarising voltage to it. Hence a number of different polar patterns are possible, from omni through to cardioid as the voltage is increased from 0V.

Differential (or noise cancelling): Its construction takes full advantage of the proximity effect. Two transducer elements are used, one wired out of phase with the other. Consequently it is insensitive to lateral sound and also discriminates against distant sounds in favour of in close sound sources.

Hand-held: These are designed to work close to the mouth. The transducer is usually isolated from the microphone body in order to reduce handling noise. They either have a windshield fitted or integrated in order to reduce breath noises.

Lavalier and clip: Inconspicuous microphones that differs in the way they are worn by the user. The lavalier is suspended around the neck (also called neck or lanyard microphone). The clip type is attached to clothing via a clip.

Pressure zone (PZM): A microphone in which the diaphragm can be mounted very close to a boundary surface. This mounting helps to eliminate the interference effects (comb filter distortion) that occurs when sounds are reflected from a nearby surface onto a microphone.

Radio (wireless): It can be a conventional, hand-held, lavalier or clip microphone, attached to a miniature radio transmitter worn by the user. A receiver, tuned to the transmitter frequency, produces a line-level signal output that can be fed to a mixing console.

Shotgun (gun ,or rifle): A microphone which is fitted with an acoustic interference tube to make it highly directional.

Soundfield: This consists of four microphone capsules in a single casing. They are arranged and oriented in such a way that all the directional information in the 360° three-dimensional soundfield is captured. This soundfield is reproduced through a suitable array of four or more loudspeakers.

Stereo: This enables stereo pickup with a single microphone. A pair of condenser microphone capsules are housed together, closely spaced and with their axes coincident. One of the pair can be rotated with respect to the other to provide the desired offset angle.

Noise voltage (equivalent noise level, or residual noise)

The inherent noise voltage over a defined frequency band, is measured at the microphone output in the absence of sounds and sources of electromagnetic interference. It has various causes, e.g. pressure on the microphone diaphragm due to thermally agitated air molecules. Thermal agitation of the electrons in the moving coil of a dynamic microphone will cause a noise voltage across the coil resistance. The noise voltage is compared to the theoretical output voltage with the absolute value of sound pressure at the threshold of hearing (20 μPa). The measuring unit (dB) is related to the output voltage at 20 μPa (or 0dB SPL). It is measured with a 'noise voltage meter' described in DIN45405 specifications.

Overloading limit

This is not specified for dynamic microphones because of the very high sound pressure level that can be handled. Condenser microphones, however, exhibit nonlinear distortion when this limit is exceeded.

Polar patterns - see directional characteristics.

Pattern control unit

A power supply unit that enables the polarising voltage to a twin diaphragm condenser microphone to be varied manually. By altering the polarising voltage to one of the identical diaphragms the sensitivity of that diaphragm to sound is effected and will influence the polar response of the microphone. The pattern control unit enables a large number of polar pattern variations to be made, from omni through to cardioid.

Phantom power supply

A method of supplying the polarising voltage to a condenser microphone transducer. The voltage is supplied between the earthing shield of the microphone cable and the central connecting point of two matched resistors, the opposite ends of which are connected between the cable *a* and *b* conductors. Hence no DC potential exists across the transducing elements. The same supply also powers the microphone amplifier.

Polarising potential

A DC voltage is applied to the plates of a condenser microphone in order to establish a charge between them. According to DIN45596, three voltages have been standardised 12V, 24V and 48V (most common).

Pressure build-up

When sound strikes a microphone on the live axis, reflections occur within the diaphragm housing, leading to a pressure build-up. This causes a rise in high frequency sensitivity. Lateral (90°) entry of sound has no such effect, therefore the standard microphone with an omni pick-up pattern becomes partially directional at high frequency.

Proximity effect

It is the increase in bass response in most cardioid and bi-directional microphones when used close to a spherical sound wave source. A point source of sound radiates radially from the point and therefore its wavefront is spherical. The closer it is, the greater is the difference in the average distance traversed by the sound wave in ariving at the front and rear of a pressure gradient microphone diaphragm. Therefore the pressure gradient is increased. At wavelengths shorter than twice the difference in path length (ie. at high frequencies) the increase in pressure gradient is insignificant. At low frequencies, however, the pressure gradient is no longer insignificant and the microphone output is increased.

Reverberant sound

Sustaining sound that results from repeated reflections of the original sound source from the surfaces of an enclosed space. The phases of the reflected sounds arriving at an arbitrary point within the enclosed space is mostly random.

Sensitivity (free-field no-load sensitivity)

A term that specifies the ratio of effective microphone output voltage to the effective sound pressure on its diaphragm. This was previously expressed in mV/µbar, today it is mV/Pa, where 1 Pascal = 1 Newton per square meter = 10 µbar. The term free-field no-load sensitivity indicates that the measurement is performed in a free sound field with no load on the microphone. Being frequency dependent the value is usually specified at 1KHz.

Sibilance

The hissing sound of the spoken words containing S or Z. Sibilance is accentuated when such words are spoken directly into a microphone. By use of a windscreen or by talking across the microphone, this condition can be remedied.

Signal-to-noise ratio

The ratio of effective signal voltage corresponding to an effective sound pressure of 1 Newton/square meter = 1 Pa, to the equivalent (inherent) noise voltage, expressed in dB.

Sound Pressure Level (SPL)

The sound field pressure above a reference pressure, that which the human ear can just begin to hear. The lower threshold of hearing corresponds to an effective (RMS) pressure of 20 µPa. This reference pressure is given a value of 0 dB. The level of sound pressure measured above this reference is the SPL, expressed in dB. For example, on the SPL scale, 1 Pa = 94 dB SPL.

Spotters

Supplementary microphones that are used to selectively enhance some part of a performance to achieve an optimum overall balance.

Stereophonic pickup techniques

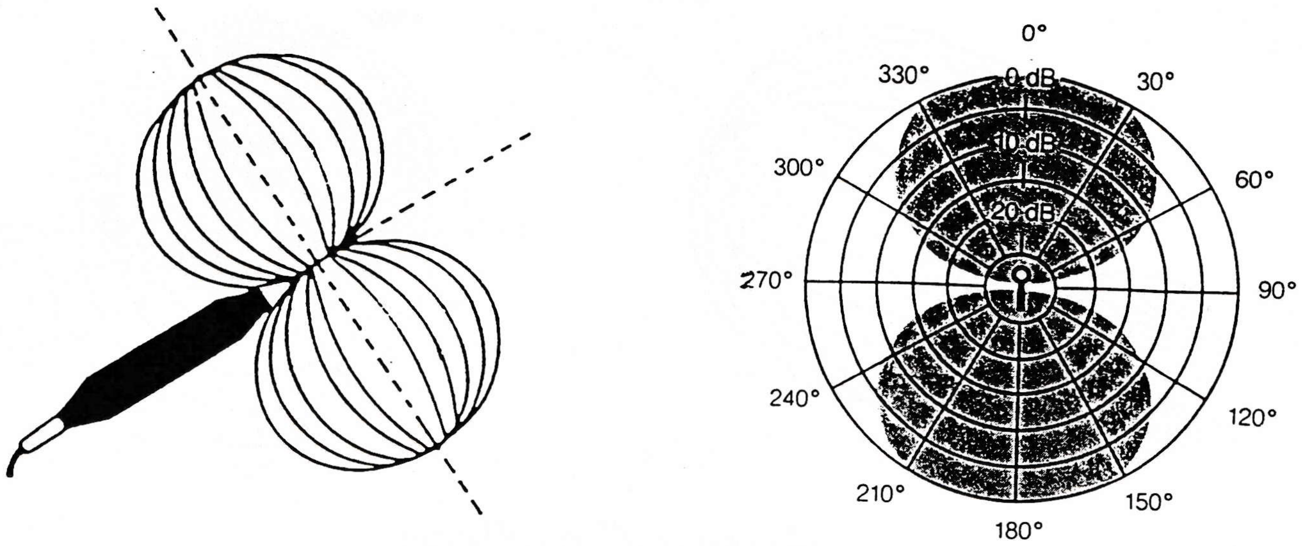
(i) *3:1 **Principle***: This principle uses two, separated microphones for stereophonic pickup. It states that for each unit of distance between the microphones should be at least three times that distance for phase integrity to be maintained.

(ii) *X-Y intensity stereo* : A pair of microphones with identical directional polar pattern, mounted on a bar with heads close together, or may be two coincident transducers in one microphone case. They are arranged so that one picks up the left predominantly and the other the right.

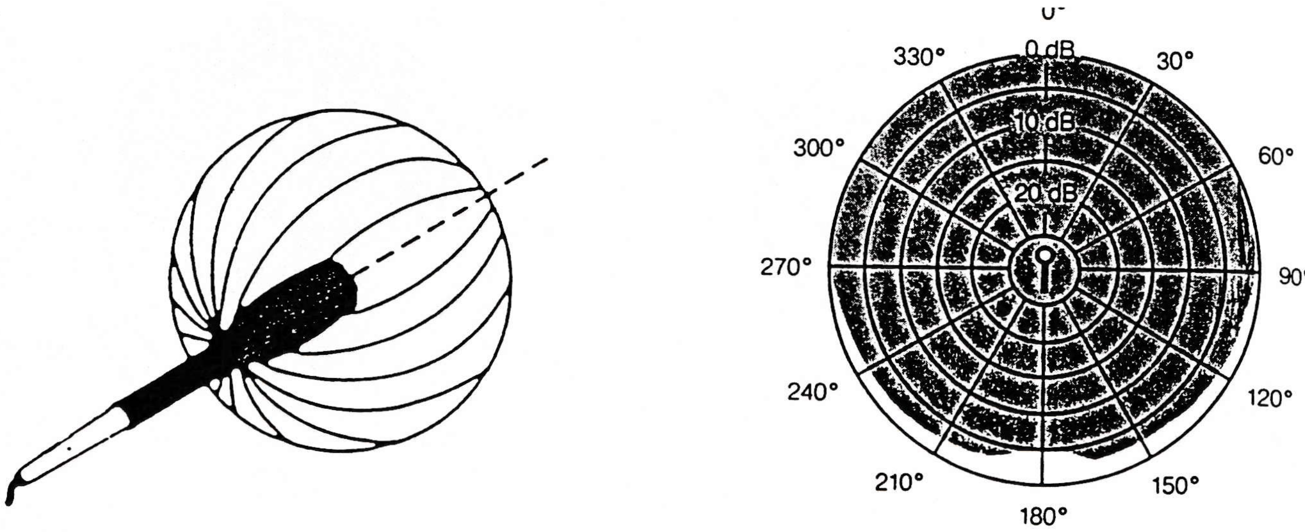
(iii) *M and S method* : This method is used solely to achieve compatibility between monophonic and stereophonic reception in receivers. Either of the above stereo techniques may be employed for stereophonic pickup. The following system of coding is then used to form a signal which is suitable for monophonic reception. The microphone outputs, designated A and B signals, are combined in an external transformer matrix to form sum (A + B) and difference (A-B) signals. The result is equivalent to the electrical output of two coincident microphones, one having a forward facing cardioid pattern and the other a figure 8 facing sideways. The sum and difference signals are called M (mid response) and S (side response) signals respectively. The M signal is the mono signal used for mono reception, while the M and S signals are transformed again in a stereo receiver to reproduce the A and B stereo signals.

Windshield (windscreen)

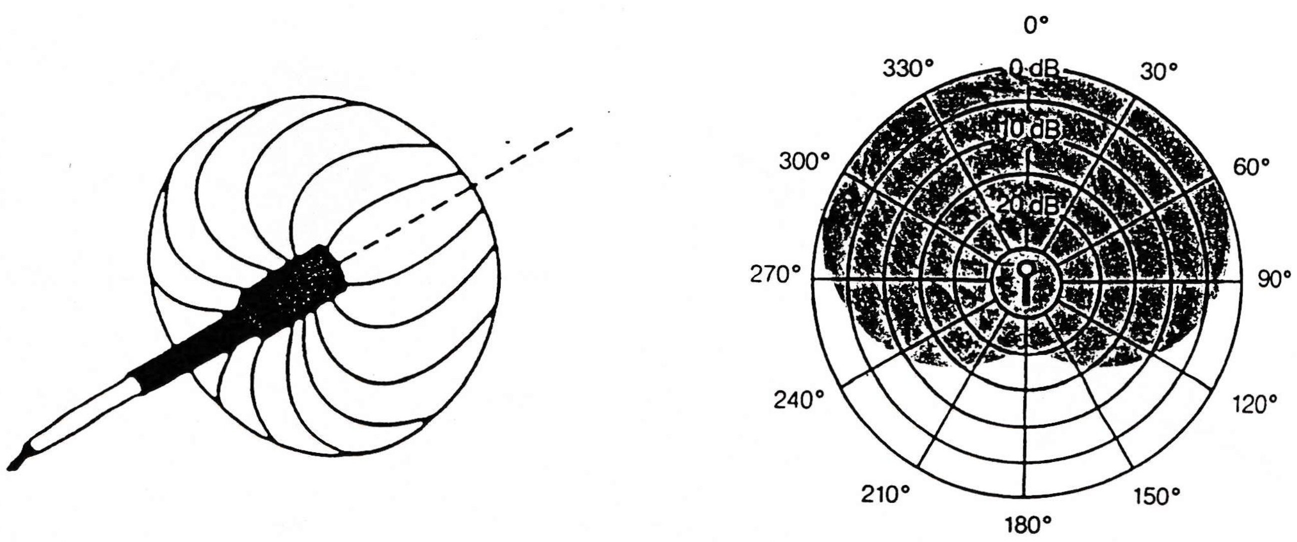
A shield that fits over a microphone. The windshield contours are designed to allow air to flow smoothly around it, thereby reducing wind noise.



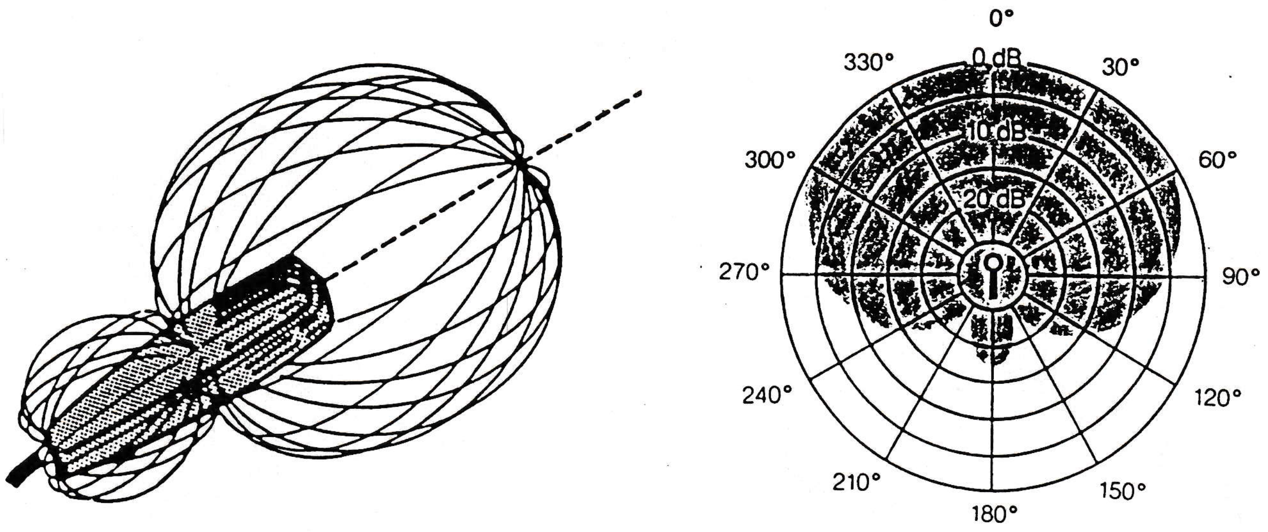
Bi-Directional or Figure-8



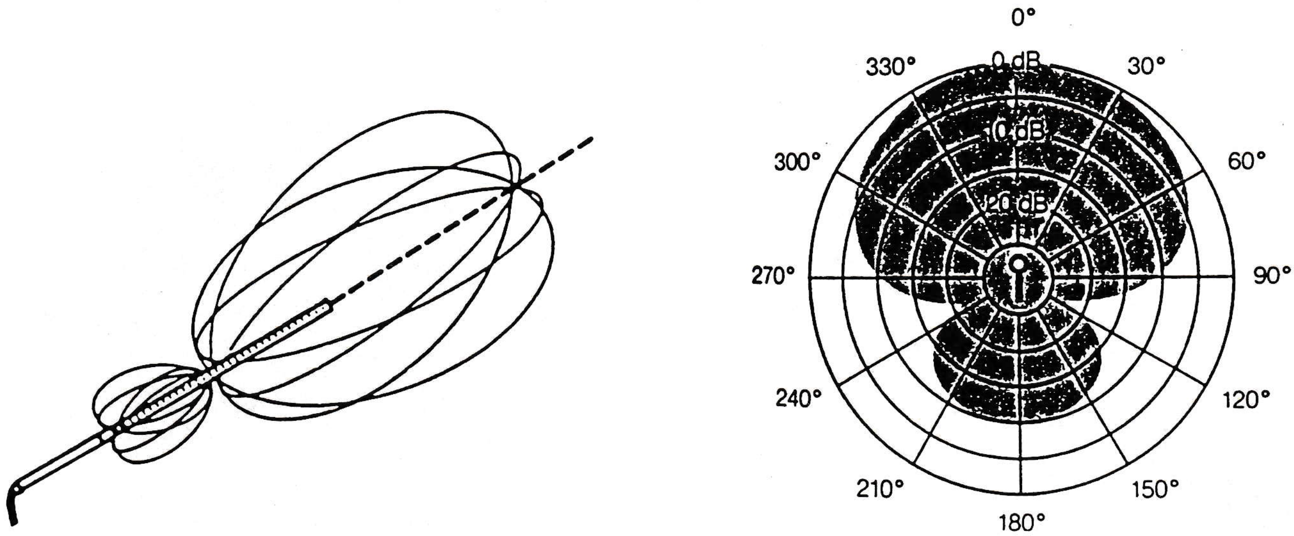
Omni-Directional



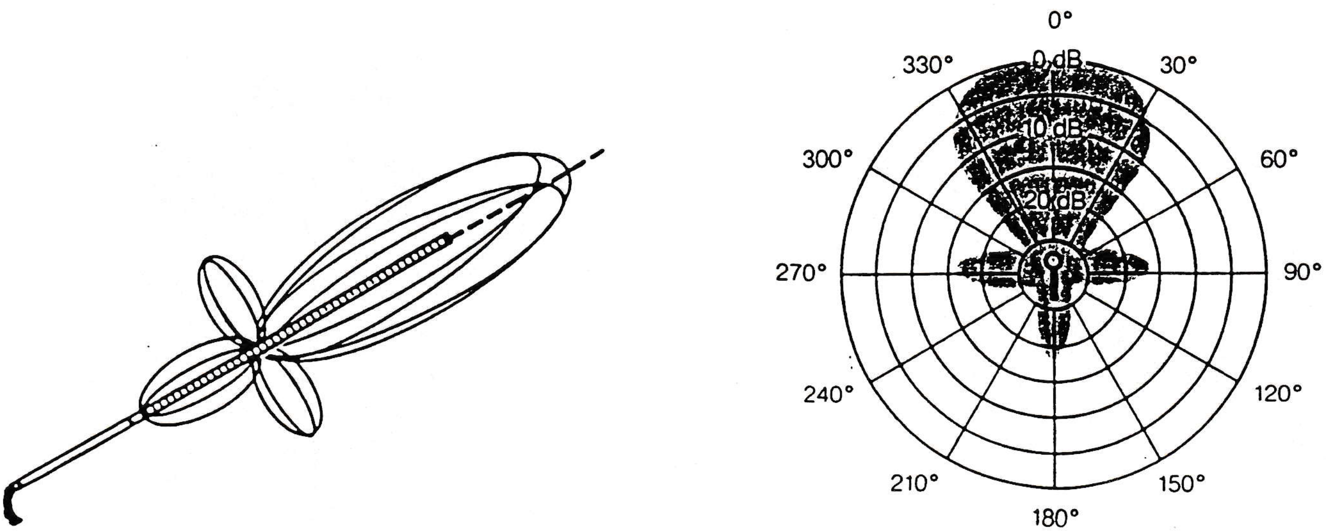
Uni-Directional or Cardioid



Super-Cardioid



Hyper-Cardioid



Shotgun or Ultra-Cardioid