BBC Engineering Training Manual

Microphones

A.E.Robertson

B.Sc.(Eng)., A.M.I.E.E.

Since the original publication of MICROPHONES developments have been so extensive that the present edition has had to be enlarged to three times the size of its predecessor.

Until the last decade microphones operated on one of two principles—pressure or pressure gradient; a third principle, phase-shift, was known, but had no commercial application until the introduction of miniaturised pre-amplifiers. This opened up a whole new range of microphones. Over the same period the directional properties of microphones have been greatly improved, and a large part of the book is devoted to these.

There is now an almost bewildering array of microphones of differing characteristics, and new designs are constantly being produced. The author makes no attempt to catalogue these but concentrates mainly on the principles of operation, and only describes actual microphones if they illustrate an important feature or have some historic significance. The book is intended for the user rather than the designer and so the mathematics have been omitted from the main body of the text, but have been collected together in a series of appendices for any who wish to study the subject more deeply.

A. E. Robertson is the Deputy Head of the BBC Engineering Training Department, and so is in a strong position to write on the subject, being able to draw on the extensive resources of the Corporation.

This book, primarily written as a training manual for BBC technicians, will also prove invaluable to all users of high quality microphones whether for broadcasting, public address systems, or recording of all types, it being remembered that the microphone is the first link in the chain of high-quality reproduction.

359 pages, including 201 text illustrations and 9 pages of art plates

A. E. Robertson was educated at Glasgow University, and after obtaining his practical experience in the heavy current engineering industry he joined the BBC in 1936 and spent seven years in the Studio Centre in Glasgow. This period gave him a wide experience of sound broadcasting not only in studios but also in outside broadcasting and mobile recording techniques. In 1943 he joined the Technical Instructional staff of the Engineering Division as a Station Instructor and when in 1946 the Corporation decided to centralise its teaching effort in a new Training School at Evesham, he was appointed Chief Instructor of the School. When in 1948 the Training School achieved departmental status within the BBC, Mr. Robertson became the Assistant Head of the Department, and was appointed its Head in 1963.

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Head of BBC Engineering Training Department

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Preface

Much of the literature which already exists on the subject of microphones has been produced primarily for physicists and engineers, yet the majority of those who use high quality microphones in the course of their day-to-day work are technical assistants or technical operators. Their mathematical standards differ from those of the engineer and the physicist and because of this they are unable to benefit from acoustical text books or from the many articles and papers which accompany a new microphone design or a new development. This book is intended then for those who use microphones rather than for those who design them.

With the exception of the Appendices, the information is presented at such a mathematical level that it can be easily understood by those having an educational standard equivalent to that of the General Certificate of Education. Since the book is primarily intended for staff in the Engineering Division of the BBC and because of its specialized nature, it assumes that the reader already has a basic knowledge of electrical engineering in general and of alternating current in particular.

The first edition was published in 1951 and was one of the first of the BBC Training Manuals to be made available to the general public. It was translated into French in 1955 by R. Clouard and was used as a technical manual by Radiodiffusion Francais. M. Clouard added an additional chapter (Chapter 7) in which he described the new microphones developed between 1950 and 1955 and he also modified and extended Chapter 6 to cover the microphones used in the studios of Radiodiffusion Francais.

The development of small pre-amplifiers in the last decade has so widened the designer's field that new and interesting microphones appear at embarrassingly short intervals. Any book which devotes much space to the description of microphones in current use in a broadcasting organization is soon out of date. It is for this reason that the second edition of this book concentrates mainly on principles of operation and only describes actual microphones if they illustrate an important development or if they have some historic significance.

It is recognized that the manual is likely to be used as a reference book rather than a text book and as a consequence some repetition is unavoidable. Chapters 1, 2, 4 and 5 of the original book are retained with only minor modifications but eight new chapters and several appendices have been added. Chapter 3 has been revised and now includes the phase-shift method of operation. Omni-directional and bi-directional microphones are dealt with in Chapters 7 and 8 and since these microphones are now well established, the subject is treated broadly. This is in contrast to Chapter 11 which discusses the development of the directional microphone in much greater detail. I felt this to be necessary because most text books when dealing with directional microphones merely state the force or directional equations, or if they are derived mathematically the treatment is at a level which is unsuitable for most technicians. In this chapter I would have liked to point out the similarity between the microphone array and the diffraction grating, or the similarity between the Beverage aerial and certain types of line microphones but it is realized that if the reader is already familiar with these subjects, the similarity will be obvious and if he is not, there is little point in pressing the analogy.

In conclusion, I wish to express my indebtedness to colleagues and friends whose comments and suggestions have on many occasions been invaluable. I am particularly grateful to J. W. Godfrey who took over all the editorial work in connection with the publication of the book and to George Mackenzie for his interest in the book and for his helpful suggestions and criticism.

Finally I wish to thank Michael Talbot-Smith for his assistance in checking the galley and page proofs.

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1963

Acknowledgements

A book of this type is by its very nature 'a posy of other men's flowers' and the author has taken full advantage of the information contained in original papers widely scattered throughout the literature. Acknowledgements have been made in references and footnotes to the many authors and if there are any omissions it is hoped that these are few in number. There are however other writers and authorities whose books and articles have influenced every chapter of this book and whose contributions to it cannot be recognized by a mere reference or footnote. The author therefore wishes to express his thanks and his especial indebtedness to L. L. Beranek, M. L. Gayford, L. E. Kinsler, A. R. Frey, P. M. Morse, H. F. Olson and D. E. L. Shorter.

Symbols

A	Area
B	Magnetic flux density
C	Capacitance (electrical)
C_a	Capacitance (acoustical)
C_m	Compliance
c	Velocity of sound propagation
D	Diameter
d	Effective path-length (front to rear of microphone)
F	Force
f	Frequency
f_r	Frequency of resonance
I	Current
I_o	Sound intensity
L	Inductance
l	Length
λ	Wavelength
M	Inertance
m	Mass
m_a	Acoustical mass
P	Power (electrical)
P_a	Power (acoustical)
p	Pressure
p_o	Free-wave pressure
ρ	Density of medium
Φ	Magnetic flux
Q	Quantity
R	Resistance (electrical)
R_a	Resistance (acoustical)

R_m	Resistance (mechanical)
r	Radius
S	Stiffness
t	Time
θ	Angle
φ	Angle of phase difference
U	Volume current
V	Voltage
u	Velocity
W	Energy (electrical)
W_a	Energy (acoustical)
W_m	Energy (mechanical)
x	Displacement amplitude
X	Reactance (electrical)
X_a	Reactance (acoustical)
X_m	Reactance (mechanical)
ω	$2\pi f$

1

Microphones in a Broadcasting Service

1.1. Introduction

THE MAIN PURPOSE of this book is to explain the elementary principles of microphones in general, to examine some of the features which distinguish one type of microphone from another and to give engineers a better understanding of the problems associated with the design and operation of microphones used in a broadcasting service.

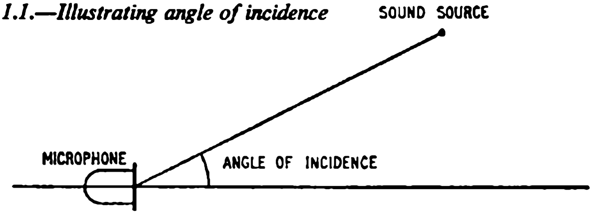
The design of an efficient microphone presents many problems which do not arise with other equipment used in the broadcasting chain. This is because the function of the microphone is the translation of sound into equivalent electrical signals; the design must therefore take into account acoustical as well as electrical principles. It is important that we should understand, from the outset, how general acoustical conditions affect the electrical performance of the microphone.

The first point to consider is the obstruction to the passage of sound waves caused by the microphone when it is placed in the sound field. This at once affects the pressure exerted on the microphone diaphragm, and consequently modifies the electrical output; this effect varies considerably as the frequency of sound varies and as the angle of incidence changes. By angle of incidence we mean the angle formed by an imaginary extended line passing through the microphone from front to back and another line joining the centre of the diaphragm to the sound source (Fig. 1.1). The degree to which obstruction and angle of incidence affect the frequency response can be controlled to a large extent by careful choice of size and shape of the microphone.

Secondly, the pressure on the diaphragm is affected if resonances occur in the air adjacent to the diaphragm. Such resonances may be caused by a cavity immediately in front of the diaphragm, formed by the clamping ring; the cavity may be very shallow, but even so, the resonance produced may considerably affect the frequency response of the microphone. It can be minimised by careful design, and in certain circumstances may be utilised to compensate for losses introduced by other parts of the system at the same frequency.

Thirdly, the mechanical and electrical impedances of the microphone may vary with frequency, and this in turn may alter the

Fig. 1.1.—Illustrating angle of incidence



relationship between the pressure applied to the diaphragm and the output voltage of the microphone.

These are some of the main problems to be encountered and the methods by which they are solved vary with different types of microphone. It is well known, however, that there are large differences in the performance of commercial microphones of different types, and a knowledge of the capabilities and limitations of all the types suitable for broadcasting is essential to engineers at studio centres.

1.2. Requirements of Broadcasting Microphones

1.2.1. GENERAL REQUIREMENTS

Before considering the principles involved in microphone design, we must give some thought to what is required of a microphone by a broadcasting service. The first requirement is fidelity of performance, which implies that a constant level of acoustical input produces a constant level of electrical output over the required audio-frequency range, i.e., the response of the microphone should be independent of frequency.

Secondly, it is desirable that the frequency response should be reasonably independent of the angle of incidence; this calls for discrimination in the size and shape of the microphone, which should be such that the body of the microphone presents minimum obstruction to the sound wave. This is particularly important at high audio frequencies when the wavelengths may approach the physical dimensions of the microphone.

In addition to these fundamental requirements, the performance of the microphone should conform to the high standards expected from other apparatus in the broadcasting chain. In particular, the microphone should be free from harmonic generation; it should respond to transients; its normal output level should be high in relation to self-generated and thermal noise; it should be unaffected by adjacent electrostatic or electromagnetic fields; its mechanical construction should be sufficiently robust to withstand handling in service use.

1.2.2. DIRECTIONAL REQUIREMENTS

The directional requirements of a microphone vary considerably with broadcasting conditions. Sometimes it is desirable that the response should be omni-directional; at other times, local conditions may call for discrimination between sounds reaching the microphone from different directions. This is particularly important when sound waves from a given source reach the microphone by different paths.

In an enclosed area such as a broadcasting studio, sound waves reach the microphone by a direct path from the source and also by numerous indirect paths because of reflections from floors, walls and ceilings; the reflected sound may form an appreciable part of the total sound reaching the microphone.

For a given ratio of direct to indirect sound the effect may be to add atmosphere or colour to musical sounds, but this same ratio may be quite unsatisfactory for speech, robbing it of its natural quality.

In practice, the indirect sound consists of repeated reflections from a number of different surfaces, the cumulative effect of which is called *reverberation*. At each reflection, a certain amount of sound energy is absorbed by the reflecting surfaces; the amount of absorption is dependent upon the physical properties of the surfaces, and varies with frequency. Because the absorption of reflected sound varies when the frequency varies, the strengths of individual tones forming the indirect sound may be different from those of the direct sound. The direct/indirect sound ratio is thus of great importance and its proper control is one of the fundamental problems confronting designers of microphones and of broadcasting studios.

In the early days of broadcasting, the sensitivity of the microphone was comparatively low and a "close" technique was employed, i.e., the microphone was placed close to the performers. An incidental result of this technique was a high ratio of direct/indirect sound; because of this, studio acoustics were relatively unimportant. In later years improvements in microphones and amplifiers made it possible to increase the distance between microphone and performers; this use of a " distant " technique not only produced a good psychological effect on the performers, but also increased the relative level of the indirect sound, giving colour to the broadcast. With the use of distant technique, however, it was found essential to control the acoustics of studios by suitable choice of reflecting and absorbing surfaces. If satisfactory acoustical conditions can be obtained in this way, the distant technique may be applied with advantage and a low ratio of direct/indirect sound maintained; given these conditions the directional discrimination of the microphone should not vary with frequency.

Where acoustic conditions are poor, it is necessary to revert to a close technique or, if this is impracticable, a microphone having directional properties may be used with a distant technique. Working under these conditions, the directional discrimination of the microphone results in a high ratio of direct/indirect energy in its output although the ratio may be relatively low at the acoustical input.

1.2.3. SPECIAL REQUIREMENTS

All microphones give a reduced output when the distance from the sound source is increased, but for some broadcasting purposes it is almost essential to use microphones with special discrimination against sound from remote sources.

In broadcasts from industrial plants, race or sports meetings and boxing matches, for instance, unwanted sounds from remote sources may be many and powerful, so that the noise reaching the microphone may drown the voice of the commentator.

To meet the requirements of such operating conditions, microphones have been developed for which the response/frequency characteristic is dependent on the distance from the sound source. An equaliser is required with a microphone of this type, because the low-frequency response of the microphone alone is excessive for sound sources which are very close (Section 10.2).

The equaliser introduces large attenuation at low frequencies, hence the response/frequency characteristic becomes reasonably flat for the commentator's voice, but there is very little response to low-frequency sounds from the remote sources.

1.3. Constant-velocity and Constant-amplitude Microphones

We have seen that a broadcasting service may use omni-directional microphones for one purpose and directional microphones for another; selection is largely a matter of operational requirements dictated by acoustical conditions. We must now consider the working principles of different types of microphone, paying particular attention to the nature of the voltage-generating device.

With almost all types of microphone, the generation of voltages is dependent upon sound waves setting up mechanical vibrations in a moving element. The voltage generated by the vibration of the moving element may be proportional either to the velocity or to the amplitude of oscillatory displacement. Microphones can therefore be classified broadly, as follows:—(a) constant-velocity; (b) constant-amplitude. (The thermal or hot-wire microphone is in neither classification, having no moving element.)

1.3.1. CONSTANT-VELOCITY MICROPHONES

A constant-velocity microphone is one in which the output is proportional to the velocity of vibration of the moving element. If the sound intensity is constant and independent of frequency, the velocity must be constant over the whole of the frequency range in order to give a constant value of electrical output.

Moving-coil and ribbon microphones are examples of the constant-velocity type and the open-circuit voltage V is given by

$$V = Blu \times 10^{-8} \text{ volts}$$

where B = density of magnetic flux (gauss),
l = length of conductor (cm),
u = velocity (cm/sec).

B and l are constants, and the output voltage is therefore proportional to the velocity. The physical characteristics of the moving system are so chosen, that for a constant level of input, the velocity is substantially constant throughout the frequency range.

1.3.2. CONSTANT-AMPLITUDE MICROPHONES

A microphone in this classification is one in which the output is proportional to the displacement amplitude of the moving element, and care is taken in the design to ensure that, for a constant value of acoustical input, the displacement amplitude is the same at all frequencies within the working range. The crystal microphone (Fig. 1.2) is an example of this type. When a sound wave impinges on the crystal surface, a deformation or deflection of the crystal is caused, which is proportional to the sound pressure. This causes an e.m.f. to be generated, owing to the so-called " piezo-electric activity " and the output voltage is proportional to the product kx, where k is a constant depending on the crystal system, and x is the effective

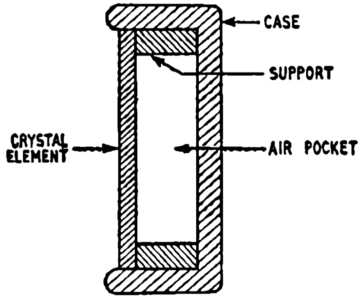


Fig. 1.2.—Cross-sectional representation of a direct-actuated crystal microphone

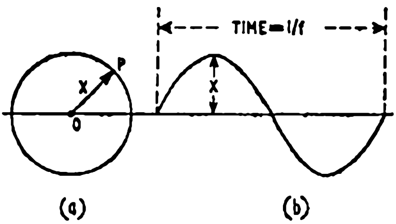


Fig. 1.3.—Graphical representation of a sinusoidal displacement

displacement of the plate from its neutral condition. Since k is constant, the output voltage is proportional to displacement.

The various types of microphone may be classified into appropriate groups, as follows.

Constant-velocity	*Constant-amplitude*
Moving-coil	Carbon
Ribbon	Piezo-electric crystal
Inductor	Electrostatic (condenser)
Moving-iron	Electronic
Magnetostriction	

1.4. Relationship between Velocity and Amplitude

The relationship between velocity, u, and amplitude x, in a vibrating mechanical system if phase effects are neglected is expressed by

$$u = \omega . x$$

where $\omega = 2\pi f$ and x = maximum displacement value (*see* Appendix 1).

If, as with a constant-amplitude microphone, x is constant, the maximum velocity u is clearly proportional to the frequency f. With a constant-velocity microphone, it can be shown that the maximum displacement value, or amplitude x is inversely proportional to frequency.

The graph in Fig. 1.3 (b) represents a cycle of sinusoidal displacement, of maximum value x. Such a graph can be traced out by rotation of the vector OP, of length x, as in Fig. 1.3 (a).

The distance travelled by the point P in one revolution is $2\pi.x$ and the time taken for one revolution is $1/f$ where f is the number of revolutions per unit time. We have then:

$$\text{distance} = 2\pi.x$$

$$\text{time} = 1/f$$

$$\text{velocity} = \frac{\text{distance}}{\text{time}} = \frac{2\pi.x}{1/f} = 2\pi f.x$$

If velocity is constant, $x \propto 1/f$.

2

Sound Waves in Air

IN ORDER TO UNDERSTAND the functioning of microphones, we must have some knowledge of the nature of sound and of its propagation through the atmosphere. It is still more important that we should know how sound energy from a point source, spreading out uniformly through an ideal medium, is reduced in intensity and how the intensity is dependent on the distance from the sound source.

2.1. Sound Pressure

The normal human ear registers a sensation of sound when periodic variations, in the form of sound waves, are superimposed on the steady atmospheric pressure, provided that the periodicity of these variations lies within what is usually termed the audio-frequency range.

The precise limits of this range depend to some extent on the age of the hearer; normally the high-frequency limit decreases as age increases. The range also varies with different individuals of similar age. The extreme limits are about 15 c/s for the lower and 20,000 c/s for the upper end of the range.

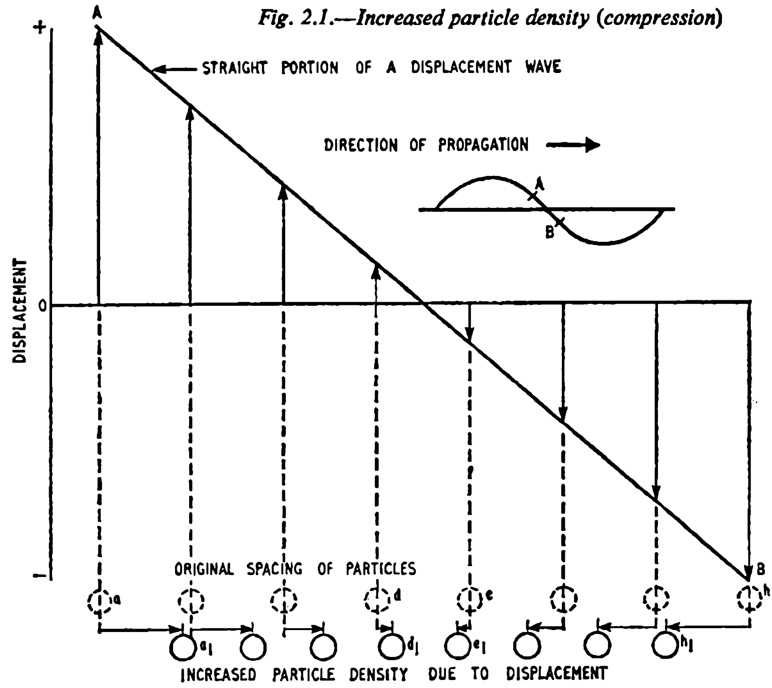
The magnitude of pressure variation which must be reached before sound becomes perceptible is not the same for all audio frequencies, but whatever the frequency, it is an extremely small fraction of the steady barometric pressure. For example, the normal atmospheric pressure is approximately 1,000,000 dyn/cm^2, yet the alternating pressures produced by speech are about 0·25 dyn/cm^2. There is, however, a very wide range of possible sound pressures: the lower limit of the range, at which sound is just audible, is referred to as the *threshold of hearing;* the upper limit at which sound becomes painfully loud is referred to as the *threshold of feeling*. For frequencies in the middle part of the audio range, the r.m.s. value of pressure variation approximates to 0·0002 dyn/cm^2 at the threshold of hearing and 1,000 dyn/cm^2 at the threshold of feeling.

2.2. Nature of Sound Waves

Sound is propagated through the atmosphere by a wave motion which is longitudinal in character, the air particles having an oscillatory displacement along the axis of propagation. When we speak of particles we imply elementary air masses, each very small, but containing millions of molecules. The displacement of the particles may be represented graphically, with the convention that displacements in the direction of wave propagation are plotted as distances above a zero axis, and backward displacements as distances below the axis. Fig. 2.1 is drawn to this convention; the line *AB* may be taken to represent a very small portion of the straight part of a sinusoidal displacement wave (shown inset), in the region of zero displacement and negative slope, i.e., when the direction of displacement is changing from forward to backward. The particle density increases when the slope of the displacement wave is negative.

It can be seen from the diagram how particle density is increased in regions where forward displacement decreases with distance or

Fig. 2.1.—Increased particle density (compression)



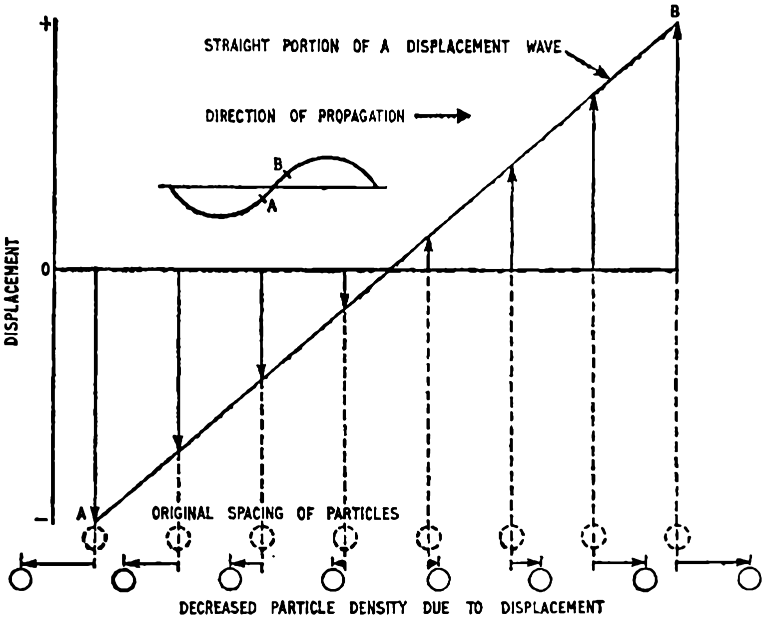


Fig. 2.2—Decreased particle density (rarefaction)

where backward displacement increases with distance: thus, particle *a* is displaced by an amount corresponding to the distance a–a_1, whereas particle *d* is only displaced by an amount corresponding to the distance d–d_1. Similarly there is a corresponding difference between the backward displacements of particles *e* and *h*. This increase in particle density with negative slope of the displacement wave is usually called *compression*.

Fig. 2.2 illustrates another region where the displacement wave passes through zero with a positive slope. It can be seen that in regions where the backward displacement decreases and forward displacement increases, the particle density decreases. This decrease in particle density is known as *rarefaction*.

Having established that particle density increases in proportion to the negative slope of a displacement wave and decreases in proportion to the positive slope, we may proceed to draw a diagram of particle distribution along the axis of propagation of a sinusoidal displacement wave. In practice, the passage of sound causes a disturbance to a whole mass of particles in its path and not merely to a single string of particles; we should, therefore, refer to particle *layers*, as represented in Fig. 2.3.

The density of the particle layer distribution has been made to correspond with the slope of the displacement wave at each point in the propagation path; thus there is compression at C and G, which are points of maximum negative slope, and rarefaction at A and E, which are points of maximum positive slope. At points such as B, D and F, where the displacement is either a positive or negative maximum, the slope is zero and the density of particle layer distribution has a normal value corresponding to the steady atmospheric pressure. The conditions represented in Fig. 2.3 are those of *plane-wave* propagation, i.e., the particle layers are flat.

From our knowledge of the particle layer distribution we are able to draw a diagram showing the sinusoidal pressure wave, because pressure is proportional to the number of particles per unit of volume; the sound-pressure wave therefore has positive

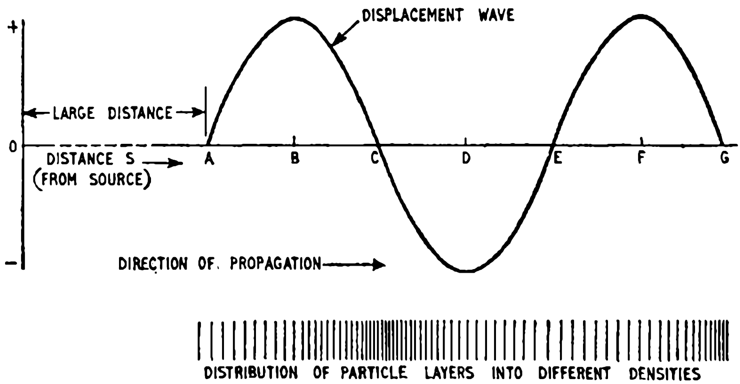


Fig. 2.3.—Displacement wave and particle layer distribution

values in regions of compression and negative values in regions of rarefaction. This is shown in Fig. 2.4, which illustrates the spacing of $\frac{1}{4}$ wavelength (λ) between the pressure and displacement waves and shows the curve of another important quantity called the *pressure gradient* (rate of change of pressure with distance, $\frac{dp}{ds}$).

The pressure-gradient curve may be drawn without reference to the particle layers, since pressure gradient is given by the slope of the

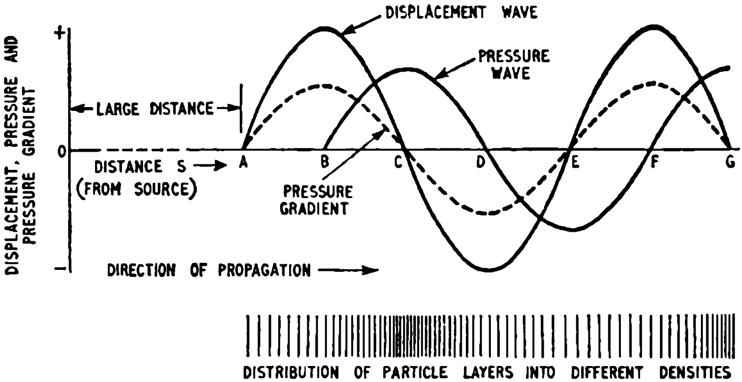


Fig. 2.4.—Displacement, pressure and pressure-gradient waves

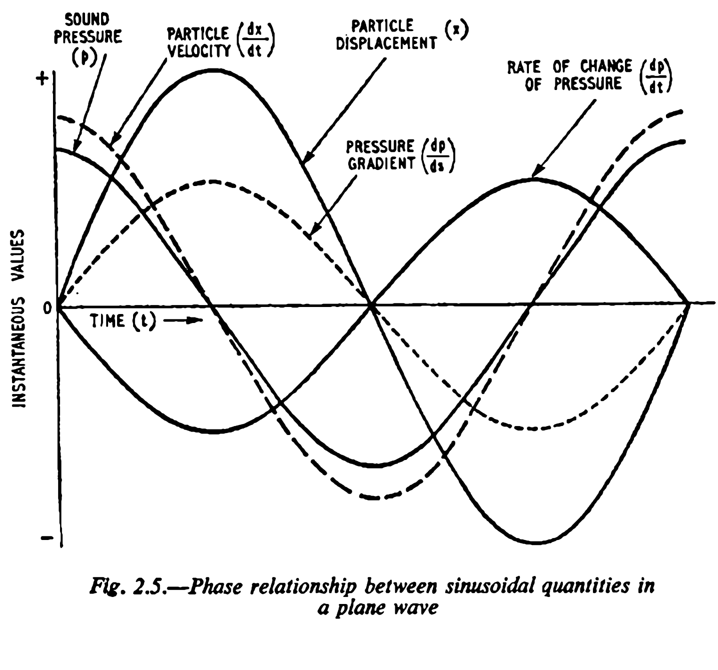


Fig. 2.5.—Phase relationship between sinusoidal quantities in a plane wave

pressure wave. The slope of the pressure curve can be altered by varying either the pressure or the frequency. It is important to realise that although the pressure may be constant, the pressure gradient will vary if the frequency is altered, for pressure gradient is proportional to frequency. With a sinusoidal pressure wave, the pressure-gradient wave is sinusoidal and displaced from the pressure wave by $\frac{1}{4}\lambda$, as shown. From the examination which we have made of conditions in the medium at a particular instant, we are able to deduce the nature of the variations of pressure and other quantities, which occur with time at a point in the propagation path; that is to say, we see the phase relationships of the various quantities in a plane sound wave.

If we consider a point such as *G*, Fig. 2.4, we can see that continued propagation of the wave from left to right will immediately cause the pressure at *G* to decrease, and the displacement and pressure gradient to increase. The various quantities will alternate in a manner which is sinusoidal with respect to time, as shown in Fig. 2.5. The zero point of the time scale in Fig. 2.5 corresponds with the instant for which conditions are depicted in Fig. 2.4.

Now that we have the waves plotted against time, we can see that the pressure wave leads the displacement and pressure-gradient waves by an amount of time equal to $1/4f$ sec, where f is the frequency of the sound (this corresponds to a phase difference of 90°). We are also able to show the phase relationships of two other important quantities by drawing in the waves for the rate of change of displacement x with time t (particle velocity $\frac{dx}{dt}$) and rate of change of pressure with time $\frac{dp}{dt}$. The slopes of the displacement and pressure curves increase with increase in frequency; consequently the rate of change of displacement (i.e., velocity) and rate of change of pressure are both proportional to frequency.

The difference in height of the various curves has no particular significance, but has been arranged in order to make them more easily distinguishable from one another.

2.3. Variation of Sound Energy and Pressure with Distance

If sound from a perfect point source is propagated equally in all directions, the wavefront has the form of a rapidly expanding spherical envelope. The term *spherical wave* is applied to this type of propagation, and since the total energy in the wave is almost

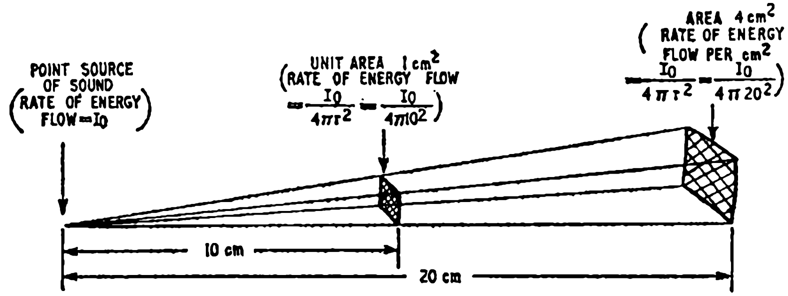


Fig. 2.6.—Illustrating decrease in sound energy as distance increases

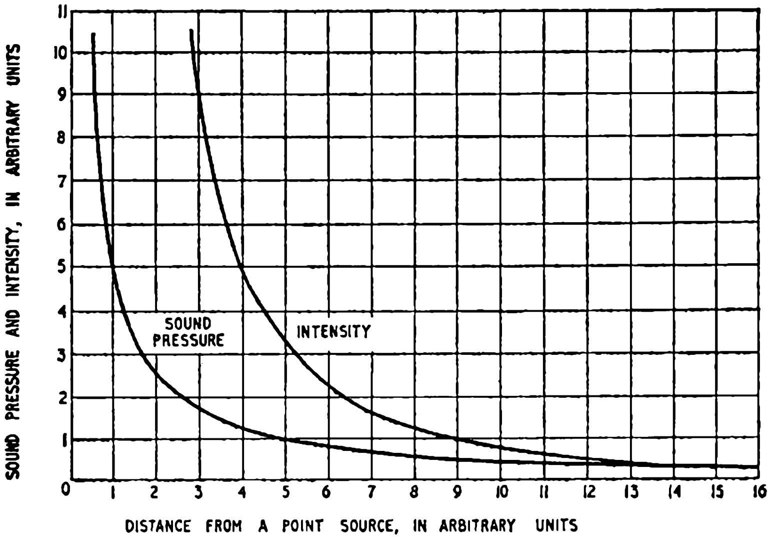


Fig. 2.7.—Illustrating decay of sound pressure and intensity as distance increases

constant, the energy per unit area of the sphere decreases as the wave expands.

Assuming negligible absorption of the sound energy by the atmosphere, the rate of energy flow (or sound intensity I_o) per unit area is given by

$$I_o = \frac{P_a}{4\pi r^2}$$

where P_a is the rate of production of sound energy at the point source (the acoustic power);
r is the distance from the point source (the radius of the spherical envelope).

Thus, even in conditions of zero absorption, the sound energy per unit area decreases as the distance increases, in the manner illustrated in Fig. 2.6.

Practical sound sources do not produce perfect spherical waves, but a section of the wavefront may be similar in form to a portion of a sphere, and the energy is distributed in a uniform manner over an appreciable angle. (At large distances from the sound source, the curvature of the wavefront is so slight that for most purposes the wave is assumed to be plane.)

The rate of flow of energy, per unit area normal to the direction of propagation, is referred to as the *sound intensity*, the unit being 1 erg/sec/cm^2. The intensity represents the power in the wave, and is given by the formula (Appendix 3)

$$I_o = \frac{p^2}{R_a}$$

where I_o = intensity units (erg/sec/cm^2),
p = sound pressure in bars (dyn/cm^2),
R_a = specific acoustical resistance.

The specific acoustical resistance is given by

$$R_a = \rho c$$

where ρ is the density of the medium (g/cm^3),
c is the velocity of propagation of the wave (cm/sec).

With atmospheric pressure corresponding to 760 mm of mercury and a temperature of 20° C, the values of ρ and c are 0·0012 and

34,400 respectively. The product is generally taken as 42, and the formula for intensity then becomes

$$\text{intensity units} = \frac{(\text{bars})^2}{42}$$

As the intensity is inversely proportional to the square of the distance r from the source, the pressure p is inversely proportional to r.

Thus

$$I_o = \frac{p^2}{R_a}$$

$$\therefore \frac{P_a}{4\pi r^2} = \frac{p^2}{R_a}$$

$$\text{and } p = \frac{1}{r}\sqrt{\left(\frac{P_a R_a}{4\pi}\right)}$$

Fig. 2.7 shows graphically the decay of sound pressure and intensity with increase in distance. Sound-pressure change over a given distance is much less when the distance to the sound source is great.

3

Operational Forces

HAVING STUDIED the properties of sound waves, in this chapter we shall examine possible methods of obtaining a force from the waves to drive the diaphragm system of a microphone. In general the force can be obtained in three ways: by *pressure operation*, by *pressure-gradient operation*, or by a combination of these called *phase-shift operation*. The method of operation is important, for it determines whether the microphone will accept or discriminate against a sound arriving from a particular direction.

3.1. Pressure Operation

We have seen that when a sound source emits sound, it does so by producing very rapid variations in the air pressure which are superimposed on the steady atmospheric pressure. If the pressure variations can be isolated from the atmospheric pressure, a small cyclical

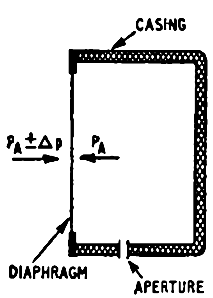


Fig. 3.1.—Cross-sectional representation of a pressure-operated microphone

force is available to drive a light diaphragm system. A microphone which is designed to isolate the atmospheric pressure and utilise the cyclical pressures remaining is called a pressure-operated microphone. It consists simply of a thin diaphragm or membrane (Fig. 3.1), stretched across a case containing air at normal atmospheric pressure.

The front of the diaphragm is exposed to the sound pressures $\pm\Delta p$ which are superimposed on the atmospheric pressure p_A. The back of the diaphragm is in contact with the air enclosed in the case and a hole is provided in the instrument: this hole is small, or so arranged that variations in barometric pressure, which occur comparatively slowly, can cause air to leak into or out of the case, but the rapid variations in pressure which constitute sound cannot be transmitted through the hole to the air in the case. Thus the air behind the diaphragm is maintained at steady atmospheric pressure.

If A is the area of the diaphragm, the total pressure on the front surface is $A\,(p_A \pm \Delta p)$, and the total pressure at the back is Ap_A. The difference of these two pressures is the force F available to drive the diaphragm, i.e.

$$F = A(p_A \pm \Delta p) - Ap_A$$
$$= \pm A\Delta p$$

Thus a force is obtained proportional to the acoustic pressure and independent of frequency. This is true for all sounds, whether they arrive along the axis of the microphone or at an angle to it, provided the dimensions of the diaphragm and case of the microphone are small compared with the wavelength of the sound.

3.2. Pressure-gradient Operation

Pressure-gradient operation differs from pressure operation in that both the front and the rear surfaces of the diaphragm, which is

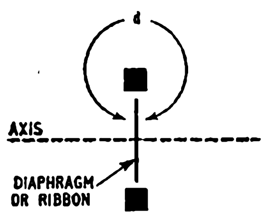


Fig. 3.2.—Simple representation of a pressure-gradient operated diaphragm or ribbon

usually in the form of a ribbon, are exposed to acoustic pressures (Fig. 3.2).

It might be thought that under such conditions no force would be exerted on the ribbon. This would certainly be so if the pressure at front and rear were identical at all times. The sound wave, however, cannot reach both surfaces at the same instant in time, for they are separated from each other not merely by the thickness of the ribbon but by the longer path which the wave must take around the baffle formed by the pole-pieces to reach the rear surface of the ribbon. The acoustic distance separating the front and rear surfaces is called the *path length d.* At a particular instant in time, the pressures at front and back of the ribbon are not identical but differ in phase by an amount depending on the length of the path *d.* In practical microphones *d* is of the order of one inch. The *pressure gradient* or

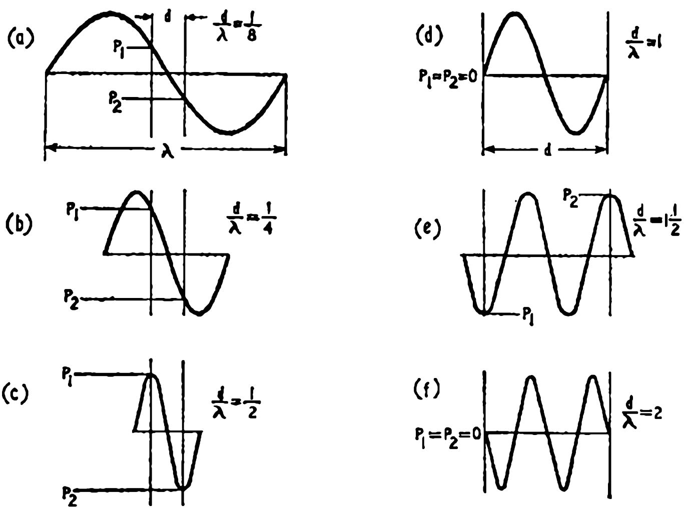


Fig. 3.3.—Force developed by pressure-gradient principle of operation at various frequencies

pressure difference which exists between the two faces causes the ribbon to move from the area of high pressure into the area of low pressure. The pressure gradient depends on the path length and is large if the path length is large. It is not so obvious however that it is also dependent on the frequency of the wave and increases if the frequency increases.

Fig. 3.3 represents sound-pressure waves of various frequencies which are assumed to be moving from left to right. The pressure gradient across the ribbon at any instant can be obtained by marking off on the abscissa a distance *d* equal to the path length (Fig. 3.3 (a)). Since the direction of propagation is from left to right, p_1 represents the pressure at the front of the ribbon and p_2 the pressure at the back.

The resultant pressure p applied to the ribbon by the sound wave is expressed by

$$p = p_1 - p_2$$

If we imagine the wave travelling from left to right, it will be seen that the resultant pressure varies sinusoidally and hence a sinusoidal force is applied to the ribbon. Fig. 3.3 (a) depicts the instant when that force is at a maximum. But the other diagrams in Fig. 3.3 also show the instant when the resultant pressure is at a maximum and it will be seen that the pressure not only varies with time but also with frequency.

Fig. 3.3 (b) shows the maximum value of the resultant pressure when the pitch of the sound has been raised by an octave, i.e. the frequency has been doubled. The diagram shows that the resultant pressure has increased and that the increase is substantially proportional to frequency. Doubling the frequency once more produces a further increase in the force but in this case the increase is small. There is therefore a limit to the range of frequencies over which the force applied to the ribbon is proportional to frequency.

There is also a limit to the maximum pressure which can be exerted on the ribbon. This condition is shown in Fig. 3.3 (c) and is known as the half-wave condition, since it occurs when the path length d is equal to half a wavelength, i.e. $d/\lambda = \frac{1}{2}$. If the frequency is increased beyond the half-wave condition the resultant pressure decreases and is zero (Fig. 3.3 (d)), when the path length is equal to a wavelength, i.e. $d/\lambda = 1$. Beyond the full-wave condition, the force reappears and has a maximum value when $d/\lambda = 3/2$ (Fig. 3.3 (e)), becoming zero at $d/\lambda = 2$ (Fig. 3.3 (f)). It should be noted that in Fig. 3.3 (d), (e) and (f) the scale for d and λ has been enlarged for the sake of clarity, just as the time base of an oscilloscope is altered for the proper examination of waveforms with widely different frequencies.

Fig. 3.3 (a), (b) and (c) shows that when the pressure gradient is negative, that is, the rate of change of pressure with time is positive, the force is in the direction of propagation and is proportional to the average slope of the pressure wave. It is not exactly proportional to the pressure gradient unless the path length d is extremely small compared with the wavelength. If $d \ll \lambda$, the average gradient corresponds to the pressure gradient, since it is equivalent to the slope at a " point " on the curve.

An expression for the force on the ribbon can be obtained from the vector diagrams of Fig. 3.4. In Fig. 3.4 (a), the pressure p_1 at the front of the ribbon is represented by the vector AB. The pressure at the back of the ribbon p_2 is represented by AC and is equal in magnitude to p_1 but lags by an angle kd radians where $k = 2\pi/\lambda$ is the wavelength constant (*see* Appendix 2.1). The resultant pressure p on the ribbon is the difference of the two vector quantities p_1 and p_2 and is represented in the diagram by CB.

An expression for this pressure p can be obtained by drawing AL so that it bisects the angle CAB. Since ALB is a right angle,

$$LB = p_1 \sin\frac{kd}{2}$$

and the resultant pressure p represented by CB is

$$p = 2p_1 \sin\frac{kd}{2}$$

Substituting $2\pi/\lambda$ for k, we have $p = 2p_1 \sin\frac{\pi d}{\lambda}$.

At low frequencies λ is large and if d is small compared with λ then d/λ will be so small that it is permissible to write $\pi d/\lambda$ for $\sin \pi d/\lambda$. In such conditions

$$p = 2p_1 \frac{\pi d}{\lambda}$$

$$= 2p_1 \pi \frac{f}{c} d \text{ since } \lambda = \frac{c}{f}$$

Hence, at low frequencies where the path length is small compared with the wavelength, the pressure $p \propto fd$. This is an important concept, for the design of the mechanical system is influenced by the fact that the pressure is proportional to frequency and since the pressure is also proportional to the path length, microphones employing the pressure-gradient principle discriminate against sounds which arrive at an angle to the axis of the microphone (*see* Section 3.2.1).

The vector diagrams in Fig. 3.4 (b), (c) and (d) have been drawn so that they correspond to the conditions shown in Fig. 3.3 (b), (c) and (d). Again it will be seen that the pressure increases with increase in frequency up to the half-wavelength condition when $d/\lambda = 1/2$ and $kd = \pi$. It then decreases and is zero when $d/\lambda = 1$ and $kd = 2\pi$. In the full-wave condition shown in Fig. 3.3 (d), the phase shift is such as to swing the vector AC through 360° so that it now coincides with AB. The acoustic pressures at front and back of the ribbon are identical in magnitude and phase. Consequently there is no pressure difference between the faces of the ribbon and thus no force on it.

The variation of the pressure with frequency plotted against the ratio d/λ is shown graphically in Fig. 3.5. The dotted line has the same slope as the initial part of the force characteristic where $F \propto f$ but the graph indicates that the proportionality between force and frequency is not maintained after about the quarter-wavelength condition when $d/\lambda = 1/4$.

There is a point of interest regarding the second hump in the force characteristic. It may be noted by reference to the equation for the force and to Fig. 3.3 that for values of d/λ between 1 and 2, the force is a negative quantity. Since we are concerned with the magnitude of an alternating quantity, the negative sign merely indicates a reversal of phase with respect to the sound wave. The reversed phase is indicated for the condition $d/\lambda = 1\frac{1}{2}$ in Fig. 3.3 (e) where the instantaneous difference of pressure is such as to produce force in a direction opposite to that of the propagation. With the pressure-gradient principle the alternating force undergoes repeated reversals of phase as the value of d/λ passes through successive whole numbers.

It has been assumed so far that throughout the audio-frequency range the diaphragm or ribbon is very small in relation to the

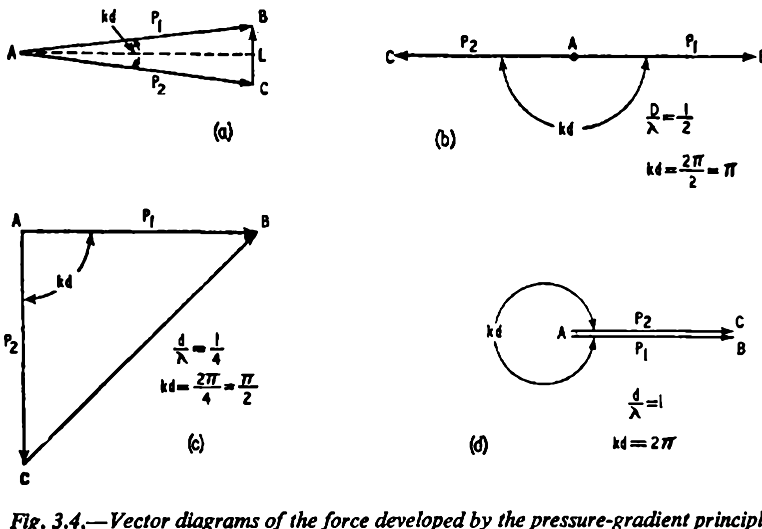


Fig. 3.4.—Vector diagrams of the force developed by the pressure-gradient principle of operation

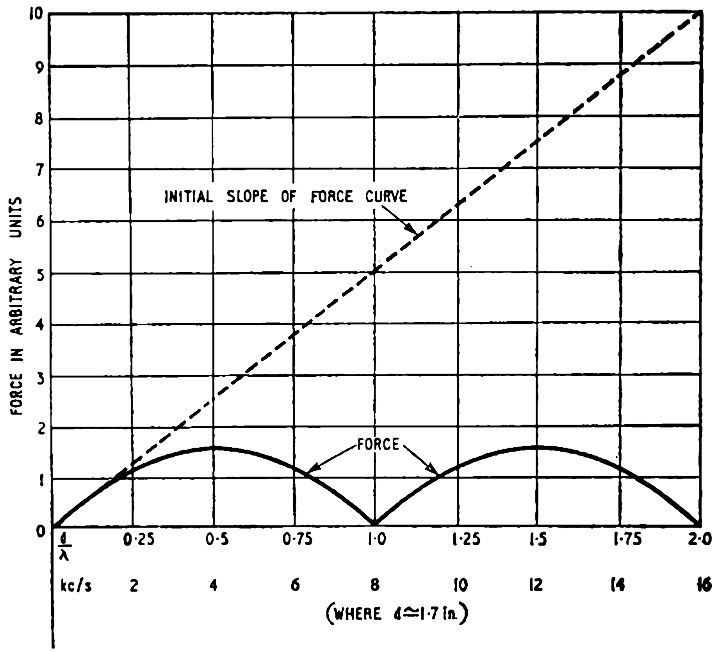


Fig. 3.5.—Pressure-gradient operation with free progressive plane wave

wavelength so that there is no appreciable obstruction of the wave. If the wave is obstructed, either by the ribbon or an adjacent surface, the effective pressures on the ribbon are changed owing to diffraction effects (*see* Section 5.3).

3.2.1. PRESSURE-GRADIENT WITH SOUND AT AN ANGLE TO THE AXIS

The force produced by pressure-gradient operation is not only proportional to the path length, but is also dependent on the angle of incidence of the sound.

In Fig. 3.6 the path length between the two surfaces of the ribbon is represented by the distance d between the two points A and B. If the angle of incidence is θ, the acoustic separation or phase shift corresponds not to AB but to the distance AC where $AC = d \cos \theta$.

The phase shift from front to back of the ribbon is therefore equal to $kd \cos \theta$ and for plane waves at frequencies so low that diffraction

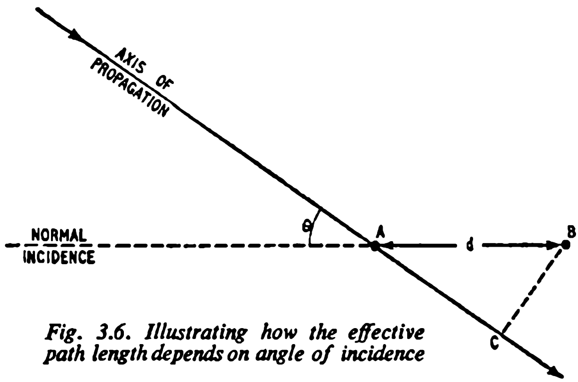


Fig. 3.6. Illustrating how the effective path length depends on angle of incidence

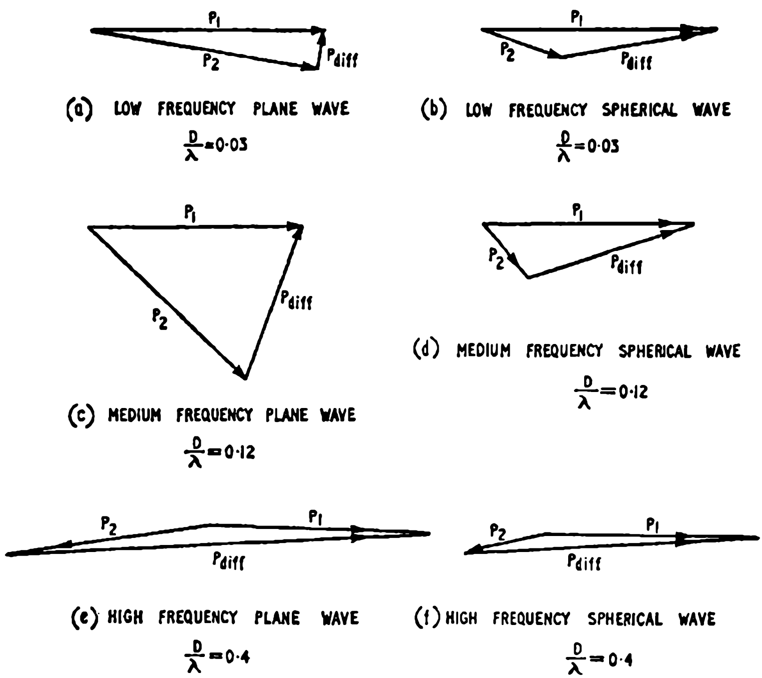


Fig. 3.7.—Vector comparison of the forces in plane and spherical waves

effects are negligible, the complete expression for the force F due to pressure-gradient operation is

$$F = 2p_1 \sin \frac{(kd \cos \theta)}{2}$$

When d is small compared with the wavelength,

$$F \simeq 2p_1 \frac{kd}{2} \cos \theta$$

$$= 2p_1 \pi \frac{f}{c} d \cos \theta$$

If θ is 90°, then $\cos \theta = 0$ and there is no pressure difference on either side of the ribbon and no force to set it in motion. A microphone of the pressure-gradient type is insensitive to sounds at right angles to the axis and is said to be " dead " in this direction.

3.2.2. INCREASE OF FORCE WITH LOW-FREQUENCY SPHERICAL WAVES

The frequency characteristic of the force due to pressure-gradient operation depends on the shape of the wavefront. If a microphone operates close to a source of sound of small dimension, the wavefront will be spherical and the pressure gradient in these conditions is greatly increased at the low frequencies. In a spherical-wave field, the pressure gradient is not only large at low frequencies, but is also dependent on the distance r of the microphone from the sound source. The increase at the low frequencies is easily demonstrable if we compare the vector diagrams for the plane- and the spherical-wave conditions.

3.2.3. COMPARISON OF THE FORCES IN PLANE AND SPHERICAL WAVES

At low frequency the phase angle between the pressures on the front and rear surfaces of the ribbon is small in both plane- and spherical-wave fields. Fig. 3.7 (a) illustrates the low-frequency plane-wave condition, the pressure at the front of the ribbon being represented by p_1 while that at the back is represented by p_2. It will be seen that the pressure difference is small and only a small force is available to operate the ribbon.

In a spherical-wave field the pressure is inversely proportional to the distance from the sound source and if the front of the ribbon is at a distance r from the source, $p_1 \propto 1/r$. If d is the path length, the rear surface of the ribbon is at a distance $d + r$ from the source and the pressure p_2 is smaller than p_1, since $p_2 \propto \frac{1}{d+r}$ (*see* Fig. 3.7 (b)).

A comparison of Fig. 3.7 (a) and (b) shows that although the phase angles are identical for both conditions, the pressure is greater in the spherical-wave field than in the plane-wave field. The increase in pressure difference in the spherical wave is accompanied by a change in the phase of the force on the ribbon. This is important and is discussed more fully in Chapter 9.

At medium frequencies, where the phase angle is appreciable, the increase in the force is only slight (Fig. 3.7 (c) and (d)). As the frequency is raised (Fig. 3.7 (e) and (f)), the force on the ribbon in the spherical-wave field falls in comparison to the plane wave and consequently the output of the microphone is reduced.

The response of a pressure-gradient microphone to spherical waves expressed in terms of the response to plane waves is given approximately for the low frequencies by the ratio of pressure gradients in the spherical- and plane-wave fields.

The r.m.s. value of the pressure gradient in a spherical wave is

$$\frac{P_{max}k}{r\sqrt{2}}\left\{1+\left(\frac{1}{rk}\right)^2\right\}^{\frac{1}{2}} \quad (\textit{see}\ \text{Appendix 10})$$

where r = radius of curvature of the wavefront (the distance between the source and the microphone),

P_{max} = *maximum pressure at sound source,*

$$k = \frac{2\pi}{\lambda} = \frac{2\pi f}{c}$$

The r.m.s. value of the pressure gradient for a plane wave is

$$\frac{P_{max}k}{r\sqrt{2}}$$

The ratio of the pressure gradient in a spherical wave to the pressure gradient in a plane wave is

$$\frac{\dfrac{P_{max}k}{r\sqrt{2}}\left\{1+\left(\dfrac{1}{rk}\right)^2\right\}^{\frac{1}{2}}}{\dfrac{P_{max}k}{r\sqrt{2}}}$$

$$=\left\{1+\left(\frac{1}{rk}\right)^2\right\}^{\frac{1}{2}}$$

For low frequencies, the spherical-wave response in decibels relative to the plane-wave response is given by

$$20 \log_{10}\left\{1+\left(\frac{\lambda}{2\pi r}\right)^2\right\}^{\frac{1}{2}}$$

$$\text{or } 20 \log_{10}\left\{1+\left(\frac{c}{2\pi f r}\right)^2\right\}^{\frac{1}{2}}$$

The ribbon microphone is an example of a type which depends for its operation on the pressure-gradient principle. The ribbon, in the form of a thin flexible strip of metal, is supported in a magnetic field provided by a permanent magnet, and exposed to acoustic pressures on both faces.

The sound causes the ribbon, which is a conductor, to move in the magnetic field and so induces an e.m.f. in the ribbon. The pole-pieces of the magnet are in close proximity to the ribbon (Fig. 3.8)

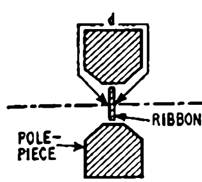


Fig. 3.8.—Cross-sectional representation of ribbon microphone, showing effective path length d *for very low frequencies*

and their shape and dimensions determine the length of the acoustic path *d* from front to rear of the ribbon.

There is no accurate mathematical expression for the distance *d*. It is approximately equal to the shortest air path from the front to the back of the ribbon. As can be seen, the path length depends on the cross-section of the pole-pieces and if the cross-section is large, the path length will be large.

The ribbon microphone used by the BBC has an effective path length at the low frequencies of approximately 1·5 in. For sinusoidal waves of the lowest frequencies, the force on the ribbon corresponds closely in magnitude and phase to the rate of change of pressure and leads the pressure and particle velocity by 90° in a plane wave (Fig. 2.6). With electrodynamic microphones of this type, the velocity of the conductor lags on the applied force by approximately 90° so that for low frequencies the generated e.m.f. is 90° out of phase with the force and has the same phase as the particle velocity.

3.3. The Phase-shift Principle

The phase-shift principle of operation can be regarded as an extension of the pressure-gradient principle, for the diaphragm of the microphone is also exposed to acoustic pressures on both front and rear surfaces. As in pressure-gradient operation, the force developed depends on the difference in the phase of the pressures on either side of the diaphragm but the phase-shift microphone differs from the pressure-gradient microphone in that the acoustic path connecting the front and rear surfaces of the diaphragm is in two distinct parts: first, an external path around the body of the microphone corresponding to the path in pressure-gradient operation and second, a path passing through the body of the microphone itself.

Fig. 3.9 (a) is a cross-sectional representation of a phase-shift microphone. Acoustic pressures can act on both faces of the diaphragm, for the apertures which pass through the body of the

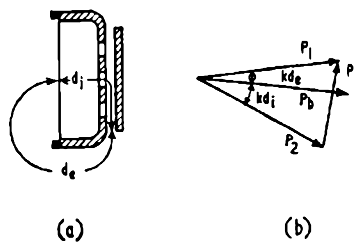


Fig. 3.9.—(a) Cross-sectional representation of a phase-shift microphone; (b) vector diagram of the resultant pressure p *obtained from the phase-shift microphone*

microphone are of such a size as to permit the sound waves to reach the back of the diaphragm with negligible loss. Thus the pressures on either side of the diaphragm are of the same amplitude but differ in phase, the difference being proportional to the path length.

The acoustic path d connecting the front and rear surfaces of the diaphragm consists of the external path d_e around the microphone (Fig. 3.9 (a)) and the internal path d_i which traverses the body of the microphone, passing via the apertures to the back of the diaphragm. Although the physical length of the external path may differ from that of the internal path, the size of the cavity and apertures may be so arranged as to make the phase shifts introduced by each path equal.

Sounds incident on the front of the microphone produce a pressure p_1 (Fig. 3.9 (b)) on the front surface of the diaphragm and a pressure p_b at the back of the microphone, of the same amplitude as p_1 but lagging by kd_e radians. Since the attenuation in the internal path is negligible, the pressure p_2 at the back of the diaphragm is equal in amplitude to p_1 and p_b but lags on p_b by kd_i radians and on p_1 by

$k(d_e + d_i)$ radians. The resultant pressure p is the vector difference of p_1 and p_2 and is given by

$$p = 2p_1 \sin k\frac{(d_e + d_i)}{2}$$

If the path length $d = (d_e + d_i)$ is small in comparison to the wavelength, then

$$\sin k\frac{(d_e + d_i)}{2} \simeq k\frac{(d_e + d_i)}{2}$$

hence
$$p = 2p_1 k\frac{(d_e + d_i)}{2}$$

The total force F on the diaphragm is

$$F = 2Ap_1\frac{\pi f}{c}(d_e + d_i)$$

where A is the area of the diaphragm and

$$k = \frac{2\pi}{\lambda} = \frac{2\pi f}{c}$$

Thus the force is proportional to frequency and to path length; a result identical with pressure-gradient operation but differing from it in that the internal path length d_i is independent of the angle of incidence of the sound. The external path, however, varies with the angle of incidence θ and its effective length is $d_e \cos\theta$ (*see* Section 9.43). Consequently the force on the diaphragm is not independent of the angle of incidence. The force F_θ at angle θ is

$$F_\theta = 2Ap_1\frac{\pi f}{c}(d_i + d_e \cos\theta)$$

In practical microphones of this type, the designer tries to ensure that the phase shift introduced by the internal path is equal to the phase shift produced by the external path.

If $d_e = d_i$,

$$F_\theta = 2Ap_1\frac{\pi f}{c}d_e(1 + \cos\theta)$$

The significance of this equation and the desirable properties possessed by such a microphone are discussed in Chapter 9.

4

Electro-acoustics

4.1. Introduction

IN THE DESIGN of a microphone, whether for constant velocity or for constant amplitude, the choice of pressure operation or differential-pressure operation depends on certain physical quantities which influence the velocity/frequency characteristic of the moving system. The velocity of the moving system for a given driving force depends on mass, compliance and friction, which are present in any mechanical or acoustical system. The study of the effects of mass, compliance and friction is facilitated if the system is represented by an equivalent electrical circuit, based on analogies between electrical and mechanical quantities.

4.2. Electromechanical Analogies

Force acting on a mechanical system is analogous to *voltage* (*electromotive force*) in an electrical system. Force exerted on a movable body or system causes motion, which is analogous to the movement of electrons in a conductor due to the application of a voltage; hence the *velocity* of a mechanical system is analogous to *current* in a circuit.

The mass of a body is the amount of matter contained therein, and determines the weight of the body. *Mass* in a mechanical system is analogous to *inductance* in an electrical circuit; the inertia of the mass opposes changes of velocity in a mechanical system, just as inductance opposes changes of current in an electrical circuit.

If the mass m has attained a velocity u under the action of a force, the energy stored is $\frac{1}{2}mu^2$.

When a current has been established in an inductance, the energy in the circuit is $\frac{1}{2}LI^2$.

The compliance of a mechanical system may be defined as the amount of deflection or displacement caused by a unit value of applied force. It is the reciprocal of stiffness, and a large value of compliance indicates that the system is easily flexed. *Compliance* in a mechanical system is analogous to *capacitance* in an electrical circuit; it has the property of accepting and storing the energy expended in the process of deflection. If the deflecting force is withdrawn, the stored energy is released, in a manner similar to the discharge of a capacitor.

Resistance (or " damping ") in a mechanical system is analogous to resistance in an electrical circuit. Mechanical resistance is generally caused by friction, but in microphones of the moving-coil type it can be produced by eddy-current damping.

To maintain a mechanical system in a state of vibration, there must always be an applied force sufficient to overcome mechanical resistance. The force required is proportional to the product of the velocity and the resistance, just as the voltage applied to an electrical resistance is proportional to the product of current and resistance.

Let us now consider the relationship between these analogies and certain fundamental laws of linear motion, and see how this relationship can be applied to determine the velocities and amplitudes of simple harmonic motion which occurs in the moving systems of microphones. (A derivation of the analogies from the equations for simple harmonic motion is given in Appendix 1.)

4.2.1. MASS AND INDUCTANCE

If we have a mass m, free from friction, and accelerating because of an applied force F, the law of motion, by Newton's Second Law, is $F = ma$, where a is the acceleration. The acceleration is the rate of change of velocity, and we may write

$$F = m \times \text{rate of change of velocity}$$

$$F = m\frac{\mathrm{d}u}{\mathrm{d}t} \qquad (1)$$

If a voltage V is applied to an inductance L, the relationship between voltage V and current i is given by the expression

$$V = L \times \text{rate of change of current}$$

$$V = L\frac{\mathrm{d}i}{\mathrm{d}t} \qquad (2)$$

Equations 1 and 2 are of identical form, and we may conclude that

force is analogous to voltage;
mass is analogous to inductance;
velocity is analogous to current.

4.2.2. COMPLIANCE AND CAPACITANCE

If a spring, having a coefficient of stiffness S, has one end fixed and its length is compressed by application of a force to the free end, the amount of movement of the latter point may be termed the displacement, x. The force equation is then $F = Sx$, and since stiffness is the reciprocal of the compliance C_m, we may write

$$F = \frac{x}{C_m} \tag{3}$$

For an electrical capacitor C, charged with a quantity of electricity Q by the application of a voltage V, we have the relationship

$$V = \frac{Q}{C} \tag{4}$$

By comparison of Equations 3 and 4 we see again that force is analogous to voltage, and may conclude that

displacement is analogous to quantity of charge;
compliance is analogous to capacitance.

4.2.3. MECHANICAL RESISTANCE AND ELECTRICAL RESISTANCE

The motion of a mechanical system is subject to various energy losses, which cause the system to have a mechanical resistance R_m. The force required to maintain a constant velocity u in the system is given by

$$F = u.R_m$$

The voltage V, required to maintain a current I, in a resistance R is given by

$$V = IR$$

The analogy between mechanical and electrical resistance is clear.

Table 4.1. ANALOGIES BETWEEN ELECTRICAL, MECHANICAL AND ACOUSTICAL QUANTITIES

ELECTRICAL			MECHANICAL			ACOUSTICAL		
Quantity	*Symbol*	*Unit*	*Quantity*	*Symbol*	*Unit*	*Quantity*	*Symbol*	*Unit*
Pressure	V	volt	Force	F	dyne	Pressure	p	dyne per cm^2
Current	I	ampere	Velocity	u	centimetre per second	Volume current	U	cm^3 per second
Power	P	watt	Power	P_m	erg per second	Power	P_a	erg per second
Charge	Q	coulomb	Displacement	x	centimetre	Volume displacement	X	cm^3
Resistance	R	ohm	Resistance	R_m	dyne second per centimetre (mechanical ohm)	Resistance	R_a	dyne second per cm^5 (acoustical ohm)
Capacitance	C	farad	Compliance	C_m	centimetre per dyne	Capacitance	C_a	cm^5 per dyne
Inductance	L	henry	Mass	m	gramme	Inertance	M	gramme per cm^4
Reactance	X	ohm	Reactance	X_m	mechanical ohm	Reactance	X_a	acoustical ohm
Impedance	Z	ohm	Impedance	Z_m	mechanical ohm	Impedance	Z_a	acoustical ohm
Electromagnetic energy	W	joule	Kinetic energy	W_m	erg	Kinetic energy	W_a	erg
Electrostatic energy	W	joule	Elastic energy	W_m	erg	Potential energy	W_a	erg

4.2.4. OTHER ANALOGIES

Apart from the main analogies which have been given, there are other analogies between all the various properties of electrical, mechanical and acoustical systems, and these are given in Table 4.1.

An explanation of the acoustical units is given in Appendix 6, together with formulæ for acoustical elements and the conversion to equivalent mechanical values.

4.3. Dynamic Conditions in a Mechanical System

In our study of microphone operation, we are concerned with the r.m.s. values of velocity and amplitude due to simple harmonic motion at various frequencies in the audio range. We can, by means of the analogies, calculate the mechanical impedance at these frequencies. The frequency characteristic of the velocity due to a constant applied force can be obtained by plotting the reciprocal of the impedance. If required, the frequency characteristic of amplitude can be obtained from that of velocity, by means of the relationship $x = u/\omega$.

4.4. Mechanical Impedance of a Simple System

A simple mechanical system is represented in Fig. 4.1 (a). A cylindrical mass m is capable of movement along the axis of a tube. It is subject to mechanical resistance R_m, due to contact with the

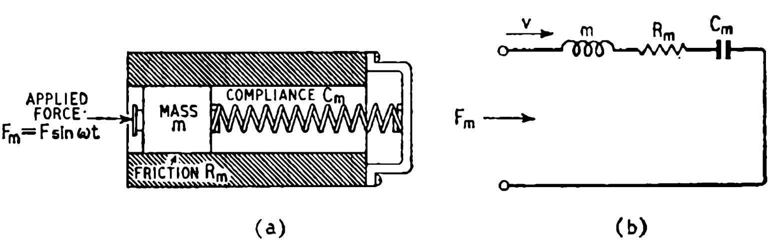


Fig. 4.1.—(a) Simple mechanical system; (b) equivalent electrical circuit

surface of the tube, and is connected by a spring, of compliance C_m, to a fixed outer bracket.

An alternating force $F_m = F \sin \omega t$ is applied to the system and a reciprocating movement is produced, which is a simple harmonic motion because F_m is sinusoidal. The compliance C_m is subjected to a velocity equal to that of the mass m, since one end of the spring is attached to the mass and the other is in a fixed position. The mechanical resistance R_m must be associated with the same velocity as the mass because it is due to friction between the mass and the wall of the tube.

Since there is a common velocity for the three elements of the system, we must represent them by elements connected in series, when drawing the equivalent circuit (Fig. 4.1 (b)).

For an electrical circuit, with inductance, capacitance, and resistance, all in series, we have

$$I = \frac{V}{Z} = \frac{V}{R + \mathrm{j}\left(\omega L - \dfrac{1}{\omega C}\right)}$$

where $\omega = 2\pi f$, and V and I are r.m.s. values.

For the series equivalent circuit of a mechanical system, we may write

$$u = \frac{F_m}{Z_m} = \frac{F_m}{R_m + \mathrm{j}\left(\omega m - \dfrac{1}{\omega C_m}\right)}$$

or we may write

$$u = \frac{F_m}{\sqrt{\left\{R^2{}_m + \left(\omega m - \dfrac{1}{\omega C_m}\right)^2\right\}}}$$

where F_m and u are r.m.s. values, and all quantities are in the units given in the table of analogies.

We can see that, for a constant value of the force F_m, the velocity u is proportional to the reciprocal of the impedance Z_m.

Example

A moving-coil microphone has a magnetic flux density B of 15,000 lines/cm^2, and the effective length l of the moving conductor is 100 cm. At 1,000 c/s, the moving system has the following effective series values of mass, compliance and resistance: $m = 0{\cdot}2$ g, $C_m = 0{\cdot}000023$ cm/dyn, $R_m = 50$ dyn sec/cm $= 50$ mechanical ohms.

Find the r.m.s. velocity and amplitude of vibration, and the open-circuit voltage generated in the coil, when the diaphragm is subjected to a sinusoidal alternating force of 25 dyn (r.m.s.) at 1,000 c/s.

$$\text{Velocity} = \frac{F_m}{Z_m}$$

$$= \frac{F_m}{\sqrt{\left\{R^2{}_m + \left(\omega m - \dfrac{1}{\omega C_m}\right)^2\right\}}}$$

$$F_m = 25$$

$$R_m = 50$$

$$\omega = 2\pi \times 1{,}000 = 6{,}280$$

$$\omega m = 6{,}280 \times 0{\cdot}2 = 1{,}256$$

$$\frac{1}{\omega C_m} = \frac{1}{6{,}280 \times 0{\cdot}000023} = 7$$

$$\therefore u_{rms} = \frac{25}{\sqrt{\{50^2 + (1{,}256{-}7)^2\}}}$$

$$= \frac{25}{1{,}250}$$

$$= 0{\cdot}02 \text{ cm/sec}$$

$$\text{Amplitude } x = \frac{u}{\omega}$$

$$u = 0{\cdot}02;\ \omega = 6{,}280$$

$$\therefore x_{rms} = \frac{0{\cdot}02}{6{,}280}$$

$$= 0{\cdot}00000318 \text{ cm}$$

$$\text{Generated voltage } V = Blu \times 10^{-8}$$

$$B = 15{,}000$$

$$l = 100$$

$$u = 0{\cdot}02$$

$$\therefore V_{rms} = 15{,}000 \times 100 \times 0{\cdot}02 \times 10^{-8}$$

$$= 0{\cdot}0003 \text{ volts}$$

4.5. Mechanical Impedance of a Complicated System

The moving system of a microphone may be much more complicated than the system shown in Fig. 4.1 and the equivalent circuit may therefore differ considerably from the simple series circuit. When drawing the equivalent circuit for a complicated system, we must be careful to observe the following cardinal principle: only those elements having a common velocity in the mechanical system may be represented by elements connected in series in the equivalent circuit.

It follows that for a mechanical system with several elements subjected to different velocities, the equivalent circuit must have a number of parallel branches; the total current through the circuit represents the velocity at the point of application of the force.

Fig. 4.2 represents a complicated mechanical system. Two cylindrical masses, m_1 and m_2, are located within a tube and coupled

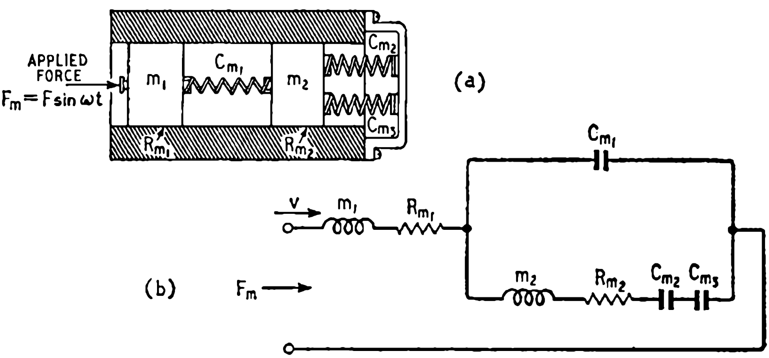


Fig. 4.2.—(a) Complicated mechanical system; (b) equivalent electrical circuit

by a spring C_{m1}. This assembly is connected by two springs C_{m2} and C_{m3} to a fixed outer bracket. Friction between the wall of the tube and the masses m_1 and m_2 causes mechanical resistance, represented by R_{m1} and R_{m2} respectively.

The total velocity of the system (velocity at the point of application of the alternating force) is the velocity of the mass m_1, with associated resistance R_{m1}; in the equivalent diagram m_1 and R_{m1} are shown in series with the remainder of the circuit.

The velocity of the spring C_{m1}, at any instant, is the difference between the velocities of m_1 and m_2; consequently C_{m1} and m_2 are shown in two separate branches of the equivalent circuit, together making up the total velocity corresponding to the velocity of m_1 and R_{m1}.

The springs C_{m2} and C_{m3} have a common velocity with the mass m_2 and its frictional resistance R_{m2}; all four elements are shown in series in the equivalent diagram. The effective compliance for the lower branch of the circuit is equal to

$$\frac{(C_{m2} \times C_{m3})}{(C_{m2} + C_{m3})}$$

The equivalent diagram makes it clear that there are not only different magnitudes of velocity in the various parts of a complicated system, but, to an extent depending on the frequency, there are differences of phase between the motions of the various elements.

4.6. Equivalent Circuits for Composite Mechanical-acoustical Systems

When drawing an equivalent circuit for a composite mechanical-acoustical system, the principles governing the series or parallel connections are the same as those for the mechanical system.

For the purpose of calculations based on the equivalent diagram, it is necessary to convert the acoustical quantities into equivalent

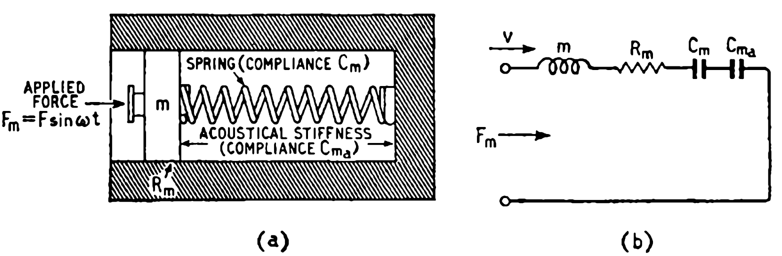


Fig. 4.3.—(a) Simple mechanical-acoustical system; (b) equivalent electrical circuit

mechanical values, as described in Appendix 6. If, for example, the force acting on a system is due to a sound wave, we have

$$F_m = pA$$

where p = effective sound pressure,
A = area of application of pressure.

Fig. 4.3 (a) represents a composite system, with mechanical properties of mass, compliance, and friction and a large additional stiffness due to the small compliance of an enclosed volume of air. The application of the alternating force F_m causes the mass m to have a velocity which is common not only to C_m and R_m, but also to the acoustical compliance, C_a, whose value in equivalent mechanical units we will call C_{ma}. In the equivalent circuit (Fig. 4.3 (b)), we

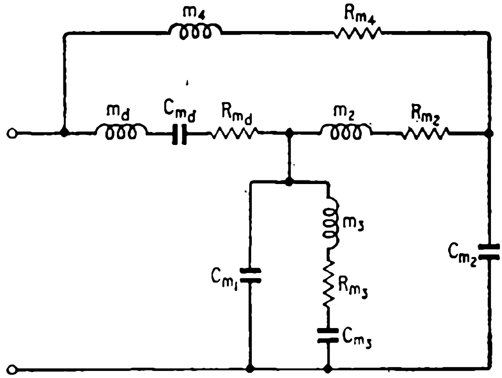


Fig. 4.4.—Equivalent circuit for mechanical-acoustical system: S. T. & C. microphone 4017A

show all four elements in series, because of their common velocity. The effective compliance is given by

$$\frac{(C_m \times C_{ma})}{(C_m + C_{ma})}$$

and, if C_{ma} is very small, the effective compliance is very small. By placing the mechanical and acoustical compliances in series, the frequency of resonance of the mechanical system is raised; with capacitor microphones this principle is sometimes used to raise the frequency of resonance to a point above audibility.

The moving parts of a microphone may have mass, compliance and friction in such proportion as to have a serious effect on the response/frequency characteristic, unless some correction of the mechanical impedance/frequency characteristic is arranged. For this purpose, it is usual to employ acoustical impedances coupled to the moving system, and the composite mechanical-acoustical system so formed may be very complicated; as, for example, that of the S. T. & C. microphone Type 4017A, represented by the equivalent circuit of Fig. 4.4. The quantities m_d, C_{md} and R_{md}

are due to actual mechanical elements; all others are due to acoustical elements.

This system has a fairly constant mechanical impedance over a large part of the audio-frequency range. The only purely mechanical elements in the system are represented by m_d, C_{md} and R_{md} the constants for the diaphragm and moving-coil assembly; the rest of the network is due to various acoustical impedances.

In such equivalent circuits, it is usual for the constants which are due to acoustical elements to be represented by the ordinary mechanical symbols m, C_m and R_m; but the values, if shown, must be mechanical equivalents of the actual acoustical values (*see* Appendix 6).

4.7. Resonance in Mechanical Systems

4.7.1. ANALOGY BETWEEN MECHANICAL AND ELECTRICAL RESONANCE

Resonance in a series electrical circuit occurs when the effective reactance

$$\omega L - \frac{1}{\omega C}$$

is zero. The current is then limited only by the resistance R, and is equal to V/R. The precise frequency of resonance is given by

$$f_o = \frac{1}{2\pi\sqrt{LC}}$$

but the peak in the current/frequency characteristic is not sharp if, at frequencies on either side of resonance, the resistance R is large compared with the reactance

$$\omega L - \frac{1}{\omega C}$$

Resonance in a simple mechanical system such as that of Fig. 4.1 occurs when the series mechanical reactance

$$\omega m - \frac{1}{\omega C_m}$$

is zero. The velocity is then limited only by the mechanical resistance R_m and is equal to F_m/R_m. The precise frequency of resonance is given by

$$f_o = \frac{1}{2\pi\sqrt{mC_m}}$$

but the peak in the velocity/frequency characteristic is not sharp if, at frequencies on either side of resonance, the resistance R_m is large compared with the series mechanical reactance

$$\omega m - \frac{1}{\omega C_m}$$

Resonance in the moving system of a microphone is, in general, undesirable. Since, at any given frequency, amplitude is proportional to velocity, a peak in the velocity/frequency characteristic is accompanied by a peak in the amplitude/frequency characteristic. Thus with any microphone, whether of the constant-velocity or constant-amplitude class, the excessive velocity due to resonance causes a peak in the response/frequency characteristic.

It is not entirely satisfactory to compensate electrically for the effects of mechanical resonance, because the relatively large amplitude of motion in the moving system is likely to cause harmonic generation, or transient distortion. Electrical circuit arrangements are sometimes introduced to compensate for minor imperfections in the motional characteristic, but it is essential for the latter to be reasonably free from the effects of resonance.

The effects of resonance may be made negligible by one of three methods. The method used depends on whether the microphone is in the constant-velocity or the constant-amplitude class and whether pressure-operated or pressure-gradient operated. The three possible methods are as follows:—

1. *Resistance control:* resonance within the working range, but heavily damped.
2. *Mass control:* resonance at a frequency much lower than the lower limit of the working frequency range.
3. *Compliance control (sometimes called stiffness control):* resonance at a frequency much higher than the upper limit of the working frequency range.

4.7.2. RESISTANCE CONTROL AND CONSTANT-IMPEDANCE SYSTEMS

The aim of this arrangement is to keep the mechanical impedance Z_m at an almost constant value throughout the frequency range. It is impossible to do this when there is only the one condition of resonance due to m and C_m, unless R_m is very large compared to the net reactance $\omega m - (1/\omega C_m)$ even at the extremes of the range. A very large value of R_m results in low sensitivity of the microphone, and, to avoid the use of a large amount of damping, acoustic reactances and resistances may be coupled to the moving element.

By such means, the mechanical impedance is kept substantially constant throughout the frequency range and the velocity produced by a constant driving force is then independent of frequency.

If the microphone is a constant-velocity type (e.g., moving-coil), the driving force must be constant, and is therefore obtained by pressure operation. An example is the S. T. & C. microphone Type 4017A, with pressure operation of the composite system of Fig. 4.4.

For a constant driving force, the velocity of a resistance-controlled system is constant and the displacement amplitude is inversely proportional to frequency. The combination of pressure operation and constant mechanical impedance is not therefore suitable for microphones of the constant-amplitude class. If a microphone of this class (such as a capacitor or crystal microphone) has a mechanical impedance substantially independent of frequency, then differential-pressure operation must be arranged, so that driving force is proportional to frequency and the displacement amplitude is therefore constant.

4.7.3. MASS CONTROL

If the product mC_m is made large enough, the frequency of resonance given by

$$f_o = \frac{1}{2\pi\sqrt{(mC_m)}}$$

is much lower than the lowest frequency of the working range. At frequencies greater than about three times the frequency of resonance, the net reactance

$$\omega m - \frac{1}{\omega C_m}$$

is not very different from the mass reactance ωm. If, at the lowest frequency of the range, R_m is small compared with ωm, the mechanical impedance Z_m is almost entirely reactive, and is practically proportional to frequency over the working range. Thus the velocity is given fairly accurately by

$$u = \frac{F_m}{\omega m}$$

and because ωm is a mass reactance, we say that the system is mass controlled.

Since the velocity is inversely proportional to frequency for a constant driving force, such a system is not suitable for pressure operation. The driving force must be proportional to frequency to maintain constant velocity; consequently pressure-gradient operation is necessary. An example is the BBC-Marconi ribbon microphone (Section 8.2).

4.7.4. COMPLIANCE CONTROL

By arranging for the effective value of the product mC_m to be very small, we can ensure that the frequency of resonance is well above the highest frequency of the working range. We cannot achieve this solely by choice of materials and dimensions for the moving element, because the requirements of extremely small mass and compliance are, to some extent, in conflict. We can, however, arrange for the mass to be very small, and a low value of the effective compliance C_m can be obtained by coupling an acoustical capacitance in series with the actual compliance of the moving element, in the manner represented in Fig. 4.3.

At frequencies below that of resonance, the net reactance of a series circuit is negative, i.e., the capacitive reactance predominates. For the mechanical system, we should say that the compliance reactance predominates. If the frequency of resonance is much higher than the highest frequency of the working range, the net reactance

$$\omega m - \frac{1}{\omega C}$$

at the latter frequency is almost equal to the compliance reactance

$$\frac{1}{\omega C_m}$$

if R_m is relatively small, the impedance through the working range is given, fairly accurately, by

$$Z_m = \frac{1}{\omega C_m}$$

and we say that the system is compliance controlled.

The velocity, for a constant driving force, is then practically proportional to frequency, as for a perfect constant-amplitude

Table 4.2. MODE OF CONTROL FOR DIFFERENT CLASSES OF MICROPHONES

MODE OF CONTROL	PRESSURE OPERATED Pressure independent of frequency				PRESSURE-GRADIENT OPERATED Pressure proportional to frequency			
	Velocity	*Amplitude*	*Type of microphone*	*Polar diagram*	*Velocity*	*Amplitude*	*Type of microphone*	*Polar diagram*
Resistance controlled	constant	$\propto \frac{1}{f}$	moving-coil	omni-directional	$\propto f$	constant	electrostatic twin diaphragm	bi-directional
Mass controlled	$\propto \frac{1}{f}$	$\propto \frac{1}{f^2}$	—	—	constant	$\frac{1}{f}$	ribbon	bi-directional
Compliance controlled	f	constant	crystal carbon condenser	omni-directional	$\propto f$	$\propto f^2$	—	—

system. Therefore the amplitude due to a constant driving force is substantially constant and the system is suitable for a pressure-operated, constant-amplitude microphone.

The direct-actuated crystal microphone (Fig. 1.2) is an example of pressure operation of a compliance-controlled system. The complete enclosure of the back of the crystal element causes the effective compliance to be very small, due to the stiffness of the pocket of trapped air.

In Table 4.2 the more important information given in Sections 4.7.2, 4.7.3 and 4.7.4 is summarised, and examples are given of common microphones that employ the principles outlined there. There are other microphones which combine pressure operation and pressure-gradient operation; the mechanical system of these is usually complex and their inclusion in this table would lead to confusion.

5

Diffraction and Dimensional Effects

5.1. Introduction

IN THIS CHAPTER we shall try to show how the physical dimensions of the diaphragm and the shape of the microphone housing affect the mean pressure acting on the diaphragm; these factors have an important bearing on the frequency response and directional characteristics of the microphone.

5.2. Diaphragm Dimensions

Most types of microphones are fitted with a diaphragm to provide a coupling between the moving element and the acoustic pressure. The force exerted on the diaphragm is proportional to the product of sound pressure and diaphragm area; it acts upon the moving system as a whole including the additional mass, compliance and mechanical resistance due to the diaphragm itself. The mechanical impedance of the diaphragm should be as low as possible for a given area, because the main consideration is to produce a velocity as high as possible for a given sound pressure. The diaphragm must be sufficiently rigid to ensure a piston-like movement; there must be flexibility at the junction of the diaphragm and casing. Little advantage is gained by using a large diaphragm, with its proportionately large mass, for the increase in force thus obtained is not accompanied by a proportionate increase in velocity, because of the greater mass reactance of the diaphragm; if the latter is large enough to be responsible for most of the mechanical impedance, the velocity cannot be increased to any appreciable extent by further increase in diaphragm area. There are, moreover, very serious objections to the use of a large diaphragm, and the choice of dimensions must take into account the three following acoustical effects:—

(i) Diffraction.
(ii) Phase difference across the diaphragm.
(iii) Cavity resonance.

5.3. Diffraction

5.3.1. EFFECTS OF DIFFRACTION

Diffraction is the term applied to the distortion of the wavefront when an obstacle is present in the sound field; this distortion is most marked when the obstacle is large compared with the wavelength of the sound, for the distribution of pressure on the diaphragm will be very different from that of free space.

When a sound impinges on an obstacle in its path, the normal free-field conditions are disturbed, for a secondary wave is produced and is scattered from the surface of the obstacle. If the obstacle is large compared with the wavelength, it acts as a reflecting surface, and a portion of the scattered wave is reflected or returned against the original wave. The magnitude of the reflected wave is approximately equal to the incident wave; these two waves produce a standing wave in front of the obstacle, the maximum amplitude of the standing wave being twice that of the original plane wave. This is the phenomenon of *pressure doubling.*

That portion of the scattered wave which is produced behind the obstacle is concentrated in such a way as to interfere destructively with the original sound field there, reducing the sensitivity in that region and producing a sharp-edged shadow.

It often happens in sound problems that the object is small compared with the wavelength; pressure doubling and the sharp-edged shadow are then absent because the scattered wave is small. In intermediate cases, where the sizes of the object and of the wavelength are comparable, a variety of interesting phenomena occurs; these will be discussed later.

The diffraction will be negligible at the lower audio frequencies for a microphone whose dimensions are of the order of two or three inches; when the frequency is increased, diffraction becomes important, and if the diaphragm faces the sound source, the pressure on it will be greater than that in free space, whilst those portions of the microphone that are remote from the sound source will be in the region of shadow or reduced pressure.

So far, our argument has been confined to sound at normal incidence, the front of the object being the surface nearest to the source of sound. If the microphone housing is not spherical, diffraction will vary also with the angle of incidence. Diffraction, therefore, not only affects the response/frequency characteristics of a microphone, but has an important bearing on its directional properties; furthermore, both response/frequency characteristics and directional properties are influenced by the physical shape of the object.

When studying the effects of diffraction, it is customary to consider the pressure p at a particular point of interest on the surface of an object, in relation to the pressure p_0 of the sound wave, at the same point, when unobstructed by an obstacle; we will call p_0 the *free-space pressure*.

We can now examine the effects of diffraction for objects of different shapes, bearing in mind that for any given shape, the effects vary as D/λ varies (D = frontal dimensions of the object and λ = the wavelength of the sound) and also as the angle of incidence varies.

The effects may be shown graphically for several angles of incidence, with the pressure ratio p/p_0 plotted as a function of D/λ. The graphs shown in the following sections are based on data for plane-wave conditions, but the effects are very much the same whether the waves are plane or spherical.

5.3.2. DIFFRACTION CAUSED BY A SPHERICAL OBJECT

The calculated diffraction by a sphere for various values of D/λ at different angles of incidence is indicated in Fig. 5.1. In this figure, the pressure ratio p/p_0 is given for a single point, A, on the

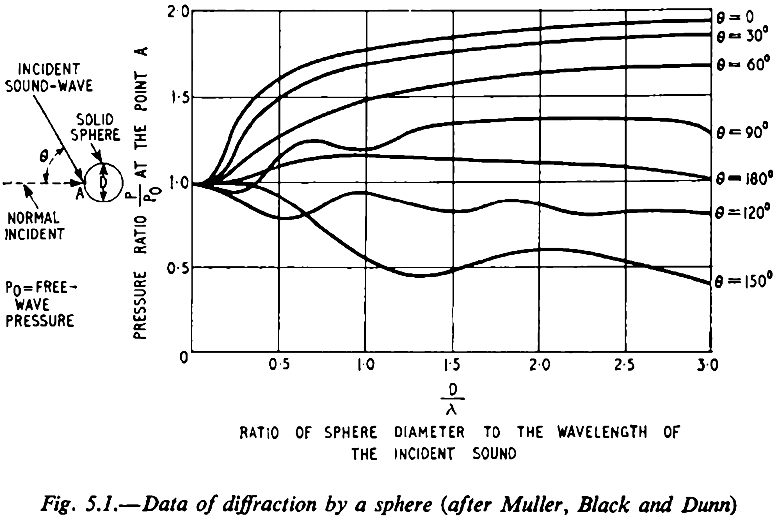


Fig. 5.1.—Data of diffraction by a sphere (after Muller, Black and Dunn)

surface of the sphere which, at normal incidence, is nearest to the sound source.

It will be seen that, for the smaller angles of incidence (0–60°) p/p_0 increases rapidly as D/λ attains a measurable value; it continues to rise, but more steadily, after $D/\lambda = 1$, until when D/λ reaches 3·0,

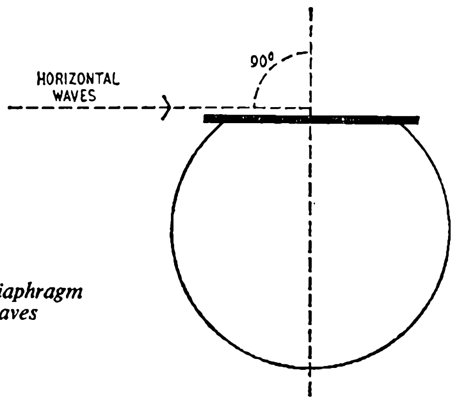


Fig. 5.2.—Microphone with diaphragm in plane of horizontal waves

the pressure attains almost double the free-wave value. This is what is meant by the term *pressure doubling*; it is an important consideration, for we shall see later that, when the surface is flat instead of spherical, the pressure may increase to more than twice the free-wave pressure for the same variations in D/λ.

The curve for $\theta = 90°$ is particularly interesting, for at this incidence the pressure remains reasonably uniform for values of D/λ above 1·5 but without pressure-doubling effect. This feature has considerable practical significance: in broadcasting, the sound sources are normally in the horizontal plane and if a microphone is placed so that its diaphragm is always in the plane of the horizontal waves (Fig. 5.2), the effective value of θ will be 90° for these waves, whatever the location of the sound source. Such a microphone, therefore, would have the same diffraction for all angles of incidence; a practical example is discussed in Section 7.17.

The values of p/p_0 for incidence greater than 90° are also of interest. The ratio falls as low as 0·4 for $\theta = 150°$ and $D/\lambda = 3·0$, but the curve for $\theta = 180°$ shows that when A is the point most remote from the sound source, the pressure at A is not less than

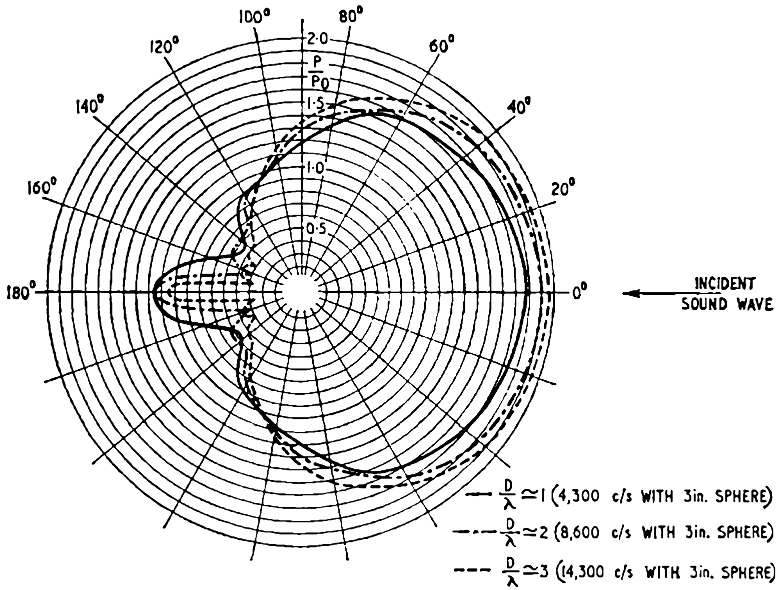


Fig. 5.3.—Polar diagram showing pressure distribution over a sphere subjected to a plane wave

the free-wave pressure for all values of D/λ between 0–3·0; indeed over most of this range the ratio p/p_0 is greater than 1·0.

So far we have been concerned with the pressure at a given point on the surface of the spherical object. We must now turn our attention to the distribution of pressure over the surface of the object, for it is this distribution which determines the most suitable size for a microphone diaphragm.

Because of the circular symmetry of the sphere, pressure distribution over the surface can be shown by means of a polar diagram based on the data supplied in Fig. 5.1. Such a diagram is drawn in Fig. 5.3 in which values of p/p_0 are plotted as concentric circles and radial lines are drawn at angles varying from 0–180°; these diagonal lines form polar angles over the surface of the sphere, and are correlated to the angles of incidence shown in Fig. 5.1. We may therefore use the data from Fig. 5.1 to plot pressure values at different points on the surface of the sphere where the polar angles intercept the circles representing p/p_0 values. A smooth curve drawn through these points gives us a pictorial representation of the pressure distribution over the surface; in Fig. 5.3 three such curves are shown for three different values of D/λ.

Let us now consider how the diffraction effects shown by the polar diagram (Fig. 5.3) affect the performance of a spherical microphone, particularly with relation to the size of the diaphragm.

At the front surface of the sphere, the pressure variation with polar angle is fairly uniform over a large central area; the average pressure on a diaphragm forming part of the spherical surface is nearly equal to the pressure at its centre, even if the diaphragm is a major portion of the front surface.

The peculiar distribution of pressure at the rear surface shows that the variation of response with various angles of incidence greater than 90° will depend to some extent on the size of the diaphragm which, if large, will tend to smooth out the sharp variations, reducing the mean pressure on the diaphragm to a value where p/p_0 is below unity. Fig. 5.3 shows that with a three-inch sphere there is, even at quite high audio frequencies, an appreciable area at the centre of the rear surface where the pressure is substantially equal to the free-wave pressure. Thus, if the diameter of the diaphragm is less than about one-third that of the spherical microphone casing, the effective operating pressure at 180° incidence is similar to that of the free wave up to fairly high frequencies. A polar diagram showing the directional characteristics of a spherical microphone with a relatively small diaphragm would be similar to Fig. 5.3; even if the small diaphragm is flat, the diffraction effect is similar to that of a sphere, except at frequencies so high that the wavelength is comparable with the diameter of the small diaphragm.

The conclusions to be drawn are, therefore, that for a spherical microphone with a large diaphragm area, pressure will be fairly uniform for a wide range of incidence, the value of p/p_0 remaining above unity for angles up to 90°; the peak pressure around 180° will be flattened, so that the mean pressure on the diaphragm will be less than unity. If, however, the diameter of the diaphragm is less than about one-third the diameter of the sphere, the mean pressure at angles of incidence in the region of 180° will be above unity.

5.3.3. DIFFRACTION CAUSED BY A CYLINDRICAL OBJECT

The pressure ratio for the centre of the end surface of a cylinder is shown in Fig. 5.4 as a function of the ratio D/λ, for various angles of incidence.

It should be noted first, that at a frequency where $D/\lambda = 1$, the pressure ratio is approximately 3·0 for normal incidence compared with 1·0 at frequencies where $D \leqslant \lambda$; this corresponds to a difference of approximately 10 dB in microphone output for different frequencies at normal incidence. Secondly, when $D/\lambda = 1$, and $\theta = 120°$ to $150°$, the pressure ratio is as low as 0·6, compared with 3·0 for the same D/λ when $\theta = 0$; this represents a difference in

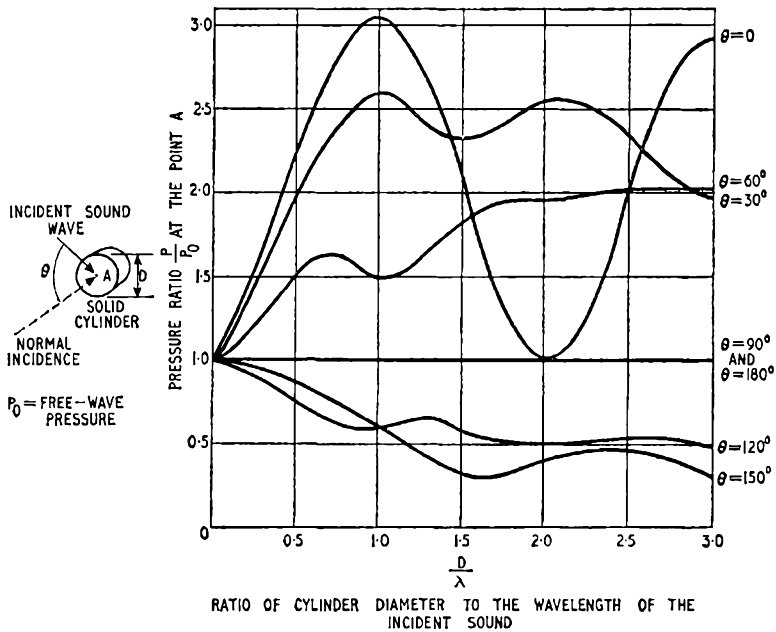


Fig. 5.4.—Data for diffraction by a cylinder

pressure ratio of 5 : 1, or approximately 14 dB in microphone output for one frequency at different angles of incidence.

Thus, for a relatively small diaphragm at the centre of the end surface of a cylindrical microphone, the response/frequency characteristic may vary by as much as 10 dB; for the same diaphragm, when $D/\lambda = 1$, the response/direction characteristic may vary by 14 dB. If these figures are compared with Fig. 5.3, it will be seen that the pressure distribution on a sphere is much more uniform as D/λ varies, and for a given frequency where $D/\lambda = 1$, the maximum difference in pressure ratio between $\theta = 0$ and $\theta = 180°$ is approximately 3·0, which is equivalent to a variation in response/direction characteristic of 10 dB.

Experimental investigations of pressure distribution indicate that the diameter of the diaphragm of a cylindrical microphone should be a large fraction of the outer diameter (unless the latter is too small to cause appreciable obstruction of sound waves). The pressure distribution across the front surface of a cylinder, with sound at normal incidence, is represented approximately in Fig. 5.5, for various values of D/λ.

In the range of D/λ from 0 to 1 the pressure distribution is fairly even over a central circular area, of diameter half that of the cylinder; beyond this limit the pressure falls away to approximately free-wave value at the outer edge. The use of a diaphragm covering most of the front surface will therefore minimise the effect of the rise of pressure up to $D/\lambda = 1$. In the range $D/\lambda = 1$ to 3 the centre point of

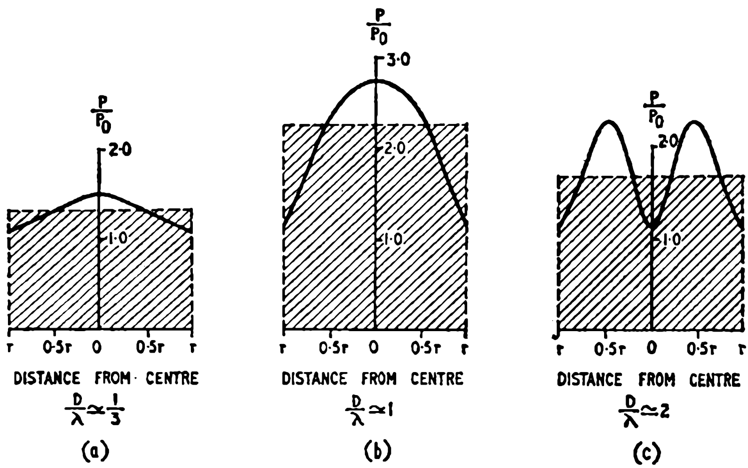


Fig. 5.5.—Approximate representation of pressure distribution on the flat end surface of a cylinder of radius r, *with sound at normal incidence*

the front surface is, so to speak, in a crater of low pressure, with the pressure rising sharply to a ring-like crest, at a distance from the centre equal to half the radius of the cylinder and then falling again towards free-wave pressure at the edge. In this range of D/λ the advantage of a relatively large diaphragm is quite definite, since the average pressure on a large diaphragm will be much more independent of frequency than is the pressure at the centre point.

The horizontal straight line shown in Fig. 5.4 for 90° and 180° incidence indicates two important features.

(a) 90° *incidence.* The diffraction caused by a surface at right angles to the wavefront is, theoretically, zero. Because of this, at normal incidence the magnitudes of the pressures at front and rear surfaces are fairly independent of the thickness of the obstacle; in fact, the pressure ratios for a thin rigid disc are taken as similar to those of a cylinder of the same diameter, but having considerable length.

(b) 180° *incidence.* As has been shown for the sphere, there is a central area at the rear surface of the obstacle where the pressure is of the same order as the free-wave value. The diameter of this area decreases with increase of frequency, and at large values of D/λ it is roughly equal to the wavelength, the pressure ratio at the centre point being constant at unity for all frequencies.

Secondary Radiation

The central area of the rear surface where the pressure is approximately that of free space has been termed the acoustic ***bright spot*** (Fig. 5.6); the existence of a similar effect in the shadow of a disc obstructing light rays was predicted by Poisson and later observed by Arago. The area of the acoustic bright spot varies with frequency and although its diameter is of the order of a wavelength immediately behind the disc, it increases in diameter as the distance from the disc is increased until it eventually terminates the sound shadow.

It has been suggested that this bright spot is due to the secondary radiation from the edge of the obstacle. It is assumed that the radiation is similar to that which would be obtained from a series of point sources situated along the edge; and for the disc, sphere or cylinder the radiation directed towards the centre arrives in phase in that region, producing the bright spot. Obviously an interference pattern results from the combination of the original sound field and the edge radiation.

Thus the sound shadow behind the disc is not uniform in intensity but varies in a regular manner giving rise to " bright " and " dark " rings. The length of the shadow is proportional to the frequency and the square of the diameter, i.e., proportional to D^2/λ. The nature of the sound shadow, in the space at the rear of a cylindrical obstacle, may be assumed similar to that of the disc

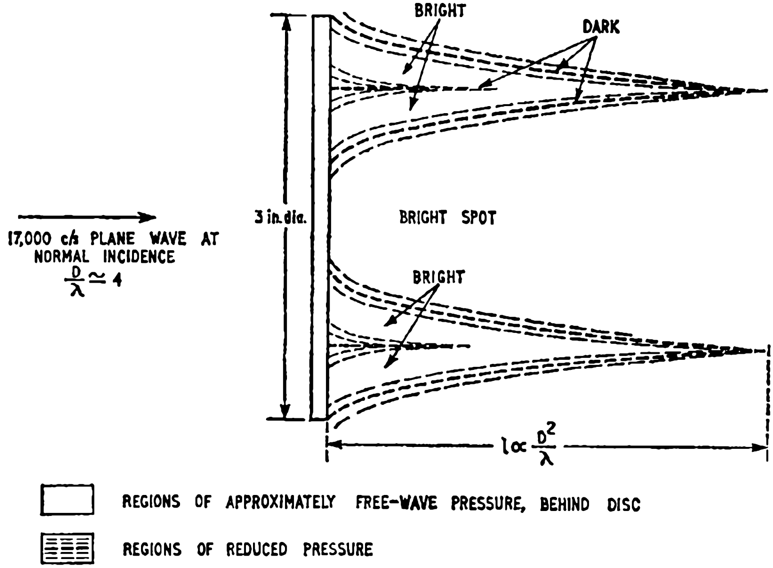


Fig. 5.6.—Approximate cross-sectional representation of sound shadow behind a rigid disc

(Fig. 5.6). The special diffraction properties of the disc are discussed in Appendix 7.

5.3.4. DIFFRACTION CAUSED BY RECTANGULAR OBJECT

The data shown for the cube (Fig. 5.7) may be taken as representative for other rectangular obstacles, the dimension W being the lesser of the two frontal dimensions, if they are different. The thickness dimension has little effect on the pressure ratios at the centres of front and rear surfaces, and the data shown for normal incidence may be assumed to apply to a rigid rectangular baffle, however thin. The thickness dimension does of course introduce an equivalent phase angle, corresponding to the propagation time for the distance from front to rear (as also for the cylinder and other shapes).

Fig. 5.7 shows that the pressure for normal incidence is approximately trebled at $W/\lambda = 1$, and this may be taken as characteristic of an obstacle with a flat surface at the front. In the diagrams for both cylinder and cube it is noticeable that the undulations of the curves show a tendency to decrease in amplitude as D/λ or W/λ becomes greater than 3·0. It is reasonable to assume that the undulations of the curves are further reduced as λ decreases, the ratio p/p_0 tending to an ultimate steady value of 2·0 at very high frequencies.

In general the curves of Figs. 5.4 and 5.7 show that there is little to choose between the cube and the cylinder for the shape of a microphone. The curve for 180° incidence (Fig. 5.7) shows that up to $W/\lambda = 1$ there is a definite " bright " area in the central part of the rear surface, although the pressure in that region decreases steadily with further increase of W/λ. This brightness is explained by the symmetry of the rectangular boundary, various distances from the centre being common to various groups of points on the outer edge, so that the secondary radiation from the edges has some cumulative effect at the centre.

The magnitude of the pressure in the central region of the rear surface is an important factor in the functioning of a ribbon microphone, where the front and rear surfaces of the ribbon occupy small central areas of what is in effect a rectangular baffle, formed by the pole-pieces and ribbon. The large frontal pressure, rising to

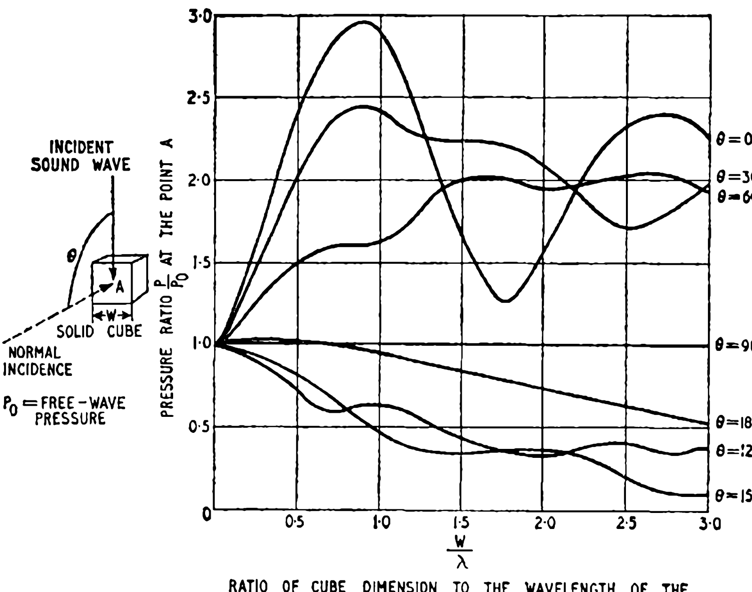


Fig. 5.7.—Data for diffraction by a cube

thrice free-space value at $W/\lambda = 1$, in conjunction with unity pressure at the rear and the incidental conditions of phase, has a profound influence on the range of frequency response. For values of W/λ greater than 1 the frontal pressure does not decrease sufficiently to equal the pressure at the rear: for example, in Fig. 5.7, at $W/\lambda = 1{\cdot}65$, the pressure ratio at the front is 1 : 1·3, and at the back 1 : 0·8,

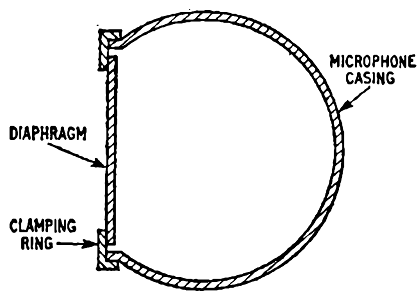


Fig. 5.8. — Spherical microphone with large flat diaphragm

so that whatever the conditions of phase, the differential pressure cannot fall to zero. Consequently there is no " extinction frequency " at which output is zero.

5.3.5. DIFFRACTION AND MICROPHONE DIMENSIONS

In concluding the subject of diffraction, the following point must be emphasised:—

1. The frontal dimension of the microphone is of great importance and must be kept as small as possible, if the effective pressure on the diaphragm is to be approximately equal to the free-wave pressure throughout the audio-frequency range.
2. The front surface of a pressure-operated microphone should be rounded as far as possible, to avoid the larger increase of pressure which occurs with a flat surface. The sphere is the best shape for a microphone, as far as diffraction effects are concerned, the variation of effective pressure with variation of sound incidence being less than for other shapes. If the diaphragm diameter is made a large fraction of the spherical casing diameter (in order to minimise the effects of sound incidence variation), then the diaphragm should be of a spherical shape; with a large flat diaphragm, such as represented in Fig. 5.8, much of the advantage of a spherical shape would be lost, the diffraction effects corresponding more closely to those of the cylinder.

3. The shape of the outer rim, edge, or boundary, is of less importance, but it does influence the pressure distribution on the rear surface. This frequently results in a greater microphone output for an incidence of 180° than for, say, 150°.
4. The graphs for the various shapes show that the pressure ratio at 90° incidence is unity for the cylinder and cube and little greater than unity for the sphere. Because of this, a microphone of almost any shape with the diaphragm facing vertically upward (Fig. 5.2) will preserve the free-space pressure of horizontal waves (usually the most important components of the sound field). This arrangement is satisfactory if the diaphragm is fairly small, but with a large diaphragm the high-frequency response to the horizontal waves may be seriously reduced by the effect of phase difference across the diaphragm; this will now be discussed.

5.4. Phase Difference Across a Diaphragm

When a flat diaphragm is subjected to a plane wave at normal incidence, the effective pressure on the diaphragm is equal to the sound pressure if diffraction is ignored. With angles of incidence other than zero or 180° the effective diaphragm pressure is less than the sound pressure, because there are differences of phase across the diaphragm surface.

The phase-difference effect with a flat diaphragm and plane-wave propagation is illustrated in Fig. 5.9, which represents instantaneous conditions of maximum pressure at the centre of the diaphragm.

In Fig. 5.9 (a) the wavelength is several times the diaphragm diameter, and the pressure at the outer edges is approximately equal to the pressure at the centre, as indicated by the ordinates drawn as arrows in the sinusoidal-wave diagram. For these conditions of frequency and incidence, the loss of effective pressure is very slight, but at a higher frequency, where the wavelength may be comparable with the diaphragm diameter, the loss of driving force on the diaphragm becomes considerable. This is shown in Fig. 5.9 (b) where the centre of the diaphragm is at maximum positive pressure, and the outer regions of its surface are at maximum negative pressure. The effect grows more severe as the wavelength becomes small relative to the diaphragm dimension.

The loss of effective pressure caused in this way is the same for any two opposite directions of propagation, as indicated by the double-headed arrows drawn for the axes of propagation in Fig. 5.9; thus, for instance, it is the same at incidence − 135° and + 135°, as at incidence + 45° and − 45°. The loss increases from zero at normal and 180° incidence to a maximum at 90° incidence, the phase difference across the diaphragm at a given frequency being proportional to the projection of the diaphragm dimension on to the axis of propagation, i.e., proportional to the sine of the angle of incidence, as well as to frequency and diaphragm dimension.

The nature of the effect is similar for a diaphragm which is other than circular, although its magnitude may be slightly different.

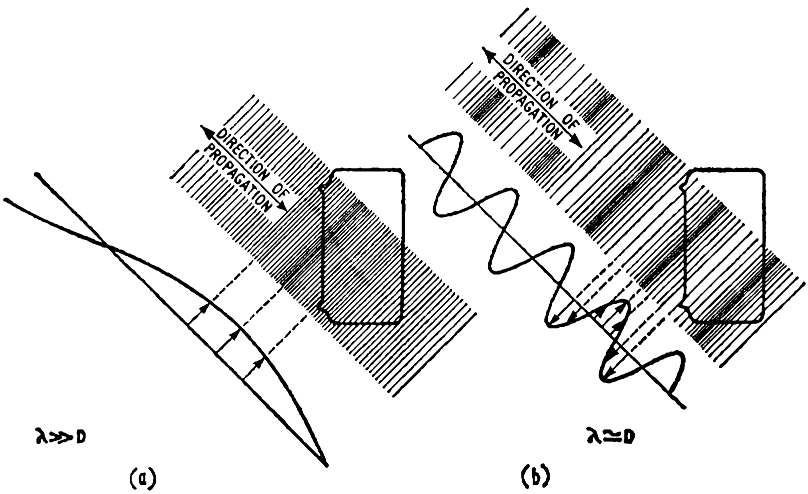


Fig. 5.9.—Effect of phase difference across diaphragm

With a very narrow and relatively long diaphragm, such as that of a ribbon microphone, the phase-difference effect is different for different planes. If a ribbon microphone is mounted in the conventional manner, with the length of the ribbon vertical, the phase-difference effect is quite negligible throughout the audio range for any angle of horizontal wave, but is appreciable for waves having a frequency of, say, 12,000 c/s or more (λ less than ribbon length) and whose direction of propagation is approximately along the length of the ribbon (e.g., waves reflected from the ceiling or floor of a studio). For planes intermediate between horizontal and vertical, there is some phase-difference effect, depending on angle of incidence and frequency.

The effective pressure on a circular diaphragm, expressed as a percentage of the normal incidence pressure, is shown for the condition of 90° incidence, as a function of the ratio D/λ (Fig. 5.10). The graph shows how serious is the reduction of operating force for transverse sounds having a wavelength comparable with the diameter of the diaphragm; for example, with a two-inch diaphragm there is a 6-dB discrimination against a 7,000-c/s signal ($D/\lambda = 1$) at 90° incidence.

The effects of diffraction variation and phase difference across the diaphragm combine to make the high-frequency response of a microphone greater for normal incidence than for 90° incidence. The diffraction variation increases the response to sound sources in front of the microphone and phase difference reduces the response to sound from the side, although in this sense their effects are cumulative; for some microphones there are intermediate angles of incidence at which the increase of pressure due to diffraction variation just compensates for the reducing effect of phase difference, making

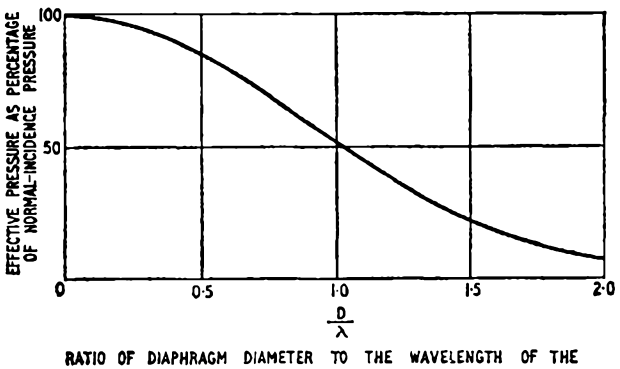


Fig. 5.10.—Phase-difference data for incidence of 90°

the operating force almost independent of frequency. This provides an explanation for the "oblique" technique practised by radio announcers using the earlier types of microphone.

With almost any pressure-operated microphone the reproduction of unpleasant sibilants in a speaker's voice may be reduced by positioning the microphone so that the angle of incidence is approximately 90°.

5.5. Cavity Resonance

A common arrangement for clamping the edge of a diaphragm is shown in Fig. 5.11. The clamping ring serves to locate the diaphragm in its correct position and also prevents access of sound to the rear surface.

The presence of the clamping ring in front of the diaphragm causes a cylindrical cavity, the air of which has inertance, compliance, and resistance. With usual diaphragm dimensions, the frequency of the main resonance of the cavity is within the audio range and

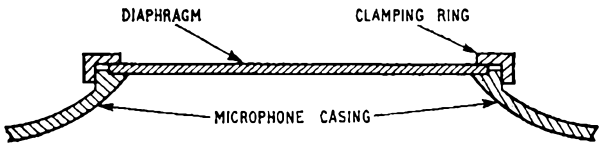


Fig. 5.11.—Diaphragm clamped by annular ring

there is a pronounced increase in pressure on the diaphragm at this frequency; there are also other frequencies of resonance, as with waves in a pipe, but their effects are slight and may be ignored.

Resonance occurs when the wavelength is approximately equal to the circumference of the cavity, i.e., when the wavelength is from 2 to 3 times the diameter.

The rise of effective pressure at the diaphragm, in proportion to the acoustical Q, depends on the depth d of the cavity (Fig. 5.12), and if the cavity resonance effect is to be negligible the depth must be a very small fraction of the diameter ($\frac{1}{10}$ or less), as shown in Fig. 5.11. This type of resonance was a most potent cause of bad performance from early microphones with deep cavities.

The cavity resonance effect is largely independent of the angle of incidence of the sound, and may therefore be put to advantage in the design of some microphones to extend the range of response at the higher frequencies.

The output of a constant-velocity type pressure-operated microphone decreases beyond some upper frequency limit, because whatever arrangements are made to secure constant mechanical impedance, the mass reactance of the moving system eventually predominates, consequently reducing velocity and output at the high frequencies.

The diaphragm may however have an appreciable cavity at the front, depth and diameter being carefully chosen so that the effect of mass reactance is offset, up to a very high frequency, by the increase of pressure due to cavity resonance. Any arrangement of this nature requires a particularly small diaphragm, for Fig. 5.12

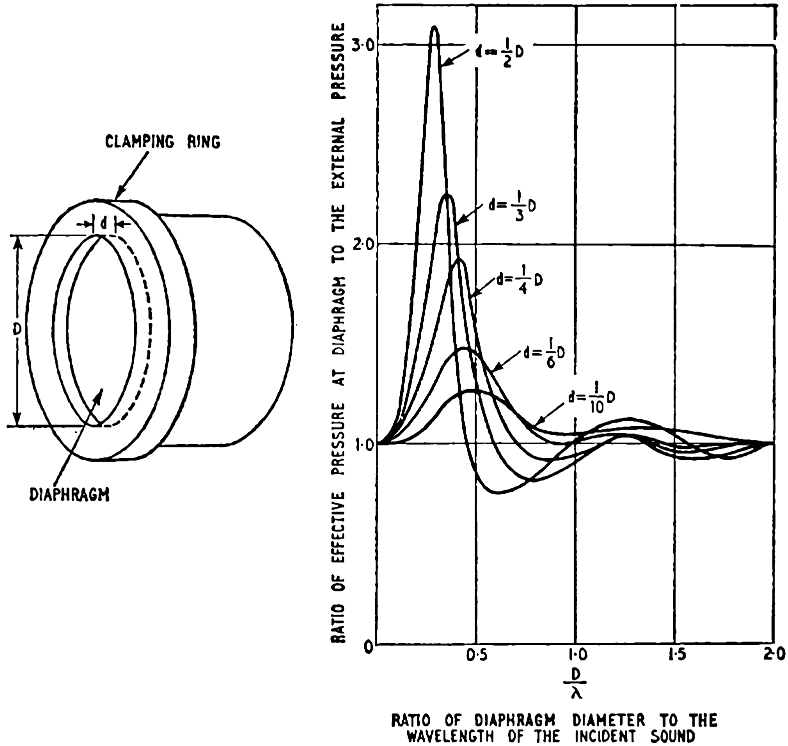


Fig. 5.12.—Resonance data for cylindrical cavities of moderate depth

shows that the effective pressure will only increase up to $D/\lambda = 0{\cdot}5$ (or less, according to depth). The diameter must therefore be from $\frac{1}{3}\lambda$ to $\frac{1}{2}\lambda$ at the highest frequency required.

5.6. Dimensions of High-quality Pressure-operated Microphones

In designing a microphone to have a response/frequency characteristic independent of the angle of sound incidence, it is necessary to exercise a strict control over the dimensions of the outer casing, the diaphragm and the diaphragm cavity. To limit the phase-difference effect to 6 dB at 15 kc/s (50 per cent loss of the pressure occurs at $D/\lambda = 1$) the diameter of the diaphragm must be less than one inch, and for diffraction effects to be moderate the dimension of the outer casing (preferably spherical) should be less than two inches. The depth of the frontal cavity should be very small for a constant-amplitude type, of the order of 0·05 in. or less; for some constant-velocity types making use of cavity resonance effects, the depth of cavity may be much greater.

For acoustical research work requiring sound-pressure detectors which are completely omni-directional and cause negligible obstruction of sound waves, small crystal microphones made from a block of lithium salt have been developed with frontal dimensions of an eighth to one-quarter of an inch.

6

Directional Response Characteristics

The last chapter showed that the physical dimensions of the diaphragm and the size and shape of the housing had an important effect on the performance of the microphone, especially on its response to sounds arriving at an angle to the axis. The variations of output with angle of incidence can be represented in two ways: first, by plotting the response of the microphone at different frequencies for a particular angle of incidence (examples of these

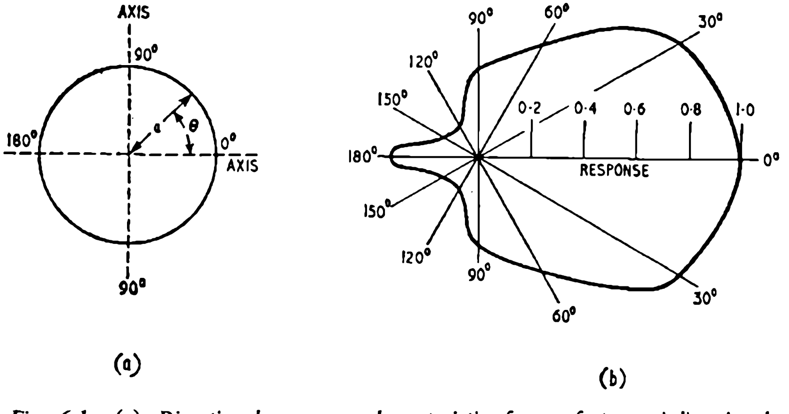


Fig. 6.1.—(a) Directional response characteristic for perfect omni-directional microphone; (b) directional characteristic for cylindrical microphone with small diaphragm located on the axis

curves will be found in Section 3.2.1), and secondly, by means of a polar graph or directivity characteristic. In this type of diagram, the magnitude of the output at various angles of incidence is plotted as a distance from a central origin, which is the point of intersection of the axes. Thus for a truly omni-directional microphone, the directivity characteristic is a circle (Fig. 6.1 (a)).

The variations of output are usually represented by means of a percentage or fractional scale, the reference level being some convenient arbitrary value for normal incidence. A decibel scale may be used with advantage for certain purposes.

6.1. Omni-directional Characteristic

A microphone which employs the pressure principle of operation does not discriminate against sounds which arrive at an angle to the axis, provided the dimensions of the microphone are small in comparison to the wavelengths of the sounds. Since the microphone favours sound from all directions, it is said to be omni-directional (or non-directional); that is to say its output is constant for all angles of incidence θ.

The directional characteristic is a circle (Fig. 6.1 (a)) whose radius is proportional to the output of the microphone. If the size of the diaphragm or case which contains the microphone is large, i.e., comparable with the wavelength, the directional characteristic is altered by diffraction and out-of-phase effects. For a cylindrical microphone with a small central diaphragm, the directional characteristic at the higher frequencies approximates to that shown in Fig. 6.1 (b). This is the polar representation of the data given in Fig. 5.4 for the diffraction of a cylinder at $D/\lambda = 1$.

A microphone having an axial response independent of frequency would give acceptable quality out of doors, but if its directional characteristic varied markedly with frequency, its performance in a studio or concert hall would be poor. The reason for this is that the acoustic energy which is " picked up " by a microphone in a studio can be divided into two parts. The first is that which arrives by the direct path between the sound source and the microphone. This is called direct energy or direct sound because it is transmitted over the shortest possible path between the source and the microphone, and is illustrated in Fig. 6.2.

The second is that which arrives by reflection from one or more of the boundary surfaces of the studio. This is referred to as indirect sound or reverberant energy. The indirect sound is the contribution the studio acoustics make to the broadcast programme and the proportion of direct to indirect energy is important, for it influences our enjoyment and enhances presentation.

The quantity of direct sound picked up by a microphone depends on its distance from the sound source but the quantity of the indirect sound received depends on the reverberation time of the studio and on the directional characteristic of the microphone. For every programme, or type of programme, there is an optimum ratio of direct to indirect sound and it is possible to change the amount of the direct sound and so alter the ratio by moving the microphone away from, or closer to the sound source.

The quality of the direct sound picked up depends on the axial frequency response of the microphone and if this is independent of

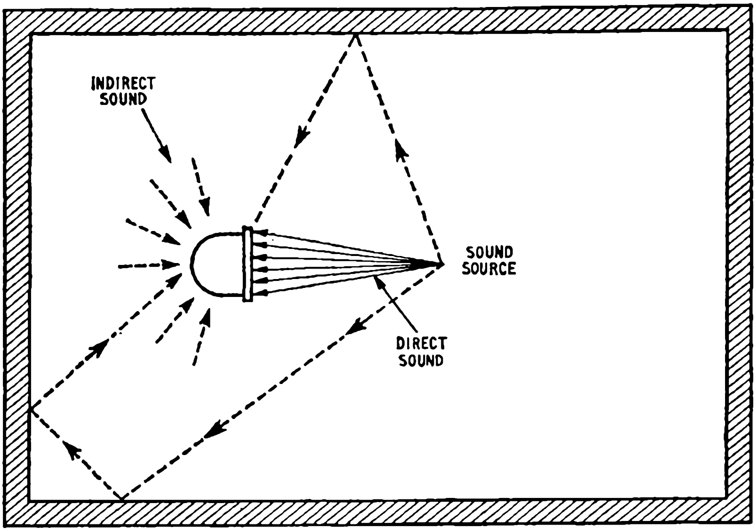


Fig. 6.2.—Sound energy received over direct sound path and indirectly by reflection from the studio surfaces

frequency, the quality of the direct sound will be satisfactory. The quality of the indirect sound depends on the two factors already mentioned, the acoustics of the studio or room and the directional characteristic of the microphone. If the directional characteristic varies markedly with frequency, the microphone is not likely to prove satisfactory for use indoors for the quality of the indirect sound will be poor.

Thus over the audio-frequency range the polar response as well as the axial response of a studio microphone should be independent of frequency. This is difficult to achieve in a practical design because of the wide frequency range (over eight octaves) over which the microphone is expected to operate.

6.2. Bi-directional Characteristic

In Chapter 3 it was shown that the force F on the diaphragm or ribbon of a pressure-gradient microphone is given by

$$F = 2p_1\pi\frac{f}{c}d\cos\theta$$

Since the output E of a microphone is proportional to the force,

$$E = afd\cos\theta$$

where a is a constant. It will be shown later that pressure-gradient microphones are designed so that the output E is independent of f, in which case

$$E \propto \cos\theta$$

since a and d are constant. This equation is the directional characteristic of the microphone and if it is plotted for all values of θ up to 90° on each side of normal incidence, a circle is produced which passes through the origin. Continued plotting for angles greater

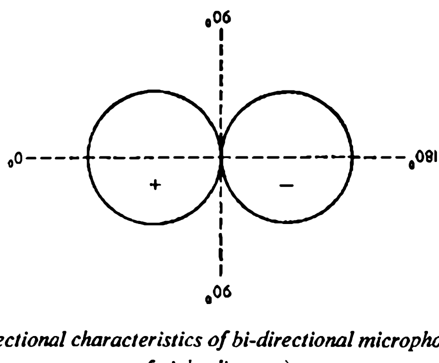


Fig. 6.3.—Directional characteristics of bi-directional microphone (figure-of-eight diagram)

than 90° on each side gives another circle occupying the remainder of the diagram (Fig. 6.3).

The positive and negative signs indicate that the output of a pressure-gradient microphone undergoes a phase change of 180° when the microphone is turned from a position facing the sound source to a position facing away from the source. As θ approaches 90° the output falls, and at $\theta = 90°$ is 0. For angles greater than

90° the output reverses in phase and increases up to a maximum at 180°.

A directional characteristic of this shape is sometimes called a *figure-of-eight diagram* and it indicates that the microphone is "live" on two faces, front and rear, where $\theta = 0$ and $\theta = 180°$. The difference in phase of the two halves of the diagram is important only if the microphone is to be used in combination with others.

Bi-directional microphones in broadcasting are generally accepted as having a working angle of 100° (50° on either side of the axis) in the forward or backward direction. Within this solid angle, the output falls only slightly for sounds off the axis, up to the limit of the working angle. The loss in dB which results if a sound on the axis is moved to the limit of the acceptance angle, i.e., 50° off the axis, is

$$20 \log \frac{\cos 0°}{\cos 50°} \text{ dB loss}$$

$$= 20 \log \frac{1}{0{\cdot}6428}.$$

$$= 3{\cdot}8 \text{ dB loss}$$

The directional characteristic of practical bi-directional microphones differs slightly from that in Fig. 6.3 and at the highest audio frequency, the characteristic is influenced to some extent by diffraction and phase-difference effects.

The improvement in the quality of studio broadcasts in the "thirties" was in part attributable to a better understanding of room acoustics and to the advent of the bi-directional microphone whose directional response was sensibly constant over its working range. Balance and control engineers found the "live" and "dead" quadrants of the characteristic helpful in both dramatic and musical programmes.

6.3. Uni-directional Characteristic

6.3.1. TWO-ELEMENT TYPE

In 1933 a new microphone appeared which employed in one unit a pressure element and a pressure-gradient element. The elements were of equal sensitivity, the directional characteristic being a combination of the omni-directional characteristic of the pressure element and the bi-directional characteristic of the pressure-gradient element. The combined directional characteristic was a cardioid of revolution with the axis normal to the plane of the ribbon of the pressure-gradient element. Microphones having a characteristic of this type are live on one face only and are known as uni-directional or *cardioid* microphones. The principle on which they are based is similar to that used in direction-finding equipment, which combines the output of a vertical aerial with the output of a loop or frame to determine the sense of the incoming signal.

In the microphone the pressure and pressure-gradient elements are connected in series and so arranged that the voltages of the two elements are in phase for sounds arriving on the $\theta = 0°$ axis. Since

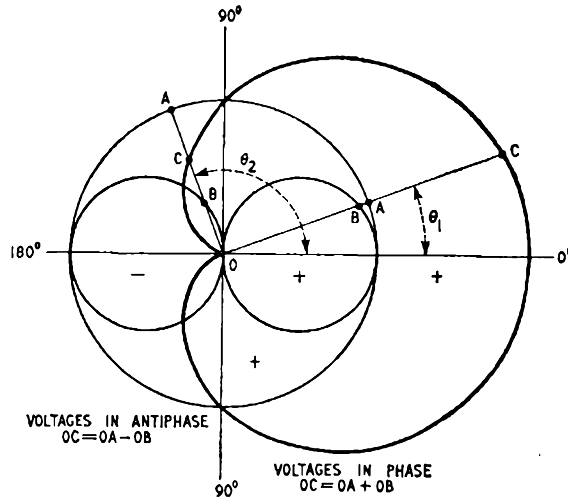


Fig. 6.4.—Uni-directional or cardioid characteristic obtained from a balanced combination of pressure and pressure-gradient units

the output of the pressure element e_p is independent of the angle of incidence, its directional characteristic is represented in Fig. 6.4 by the circle radius OA. The output of the pressure-gradient element $e_{pg} \cos \theta$ varies with the angle of incidence and its directional characteristic is represented by the figure-of-eight diagram in Fig. 6.4.

The directional characteristic of the combination can be obtained by adding the directional characteristics of the separate elements.

At angle θ_1 the combined output OC is the output of the pressure element OA plus the output of the pressure-gradient unit OB, that is,

$$OC = OA + OB$$

At angle $\theta_2 > 90°$ the outputs of the two elements are in anti-phase and

$$OC = OA - OB$$

Repeating the process for various values of θ produces the heart-shaped diagram which is the directional characteristic of the combination.

The same result can be obtained in another way. The combined output e_c of the two elements for any angle of incidence θ is given by

$$e_c = e_p + e_{pg} \cos \theta$$

If $e_p = e_{pg}$ when $\theta = 0°$ then

$$e_c = e_p (1 + \cos \theta)$$

This is the equation of a cardioid and if e_c is plotted as a polar graph for various values of θ, a heart-shaped diagram similar to that in Fig. 6.4 is obtained.

A microphone with a cardioid polar response is said to be unidirectional since it accepts sound coming from the front and discriminates against sounds coming from the sides or from the back.

If the sensitivity of the pressure element in the combination is less than the sensitivity of the pressure-gradient element, the unidirectional property is lost and to a limited extent the microphone is live at the back. Fig. 6.5 (a) shows how the directional characteristic is affected if the sensitivity of the pressure unit is reduced by 6 dB and Fig. 6.5 (b) shows the result of reducing the output of the pressure-gradient unit to 6 dB below that of the pressure unit. With most microphones, there is some arrangement by which the sensitivities of the two units can be altered. The adjustments may be acoustical, involving the alteration of slots or apertures in the microphone casing, or they may be effected electrically by means of a potentiometer or switched attenuator. Thus relatively fine adjustments of the characteristic can be obtained between the extremes of the circle and the figure-of-eight.

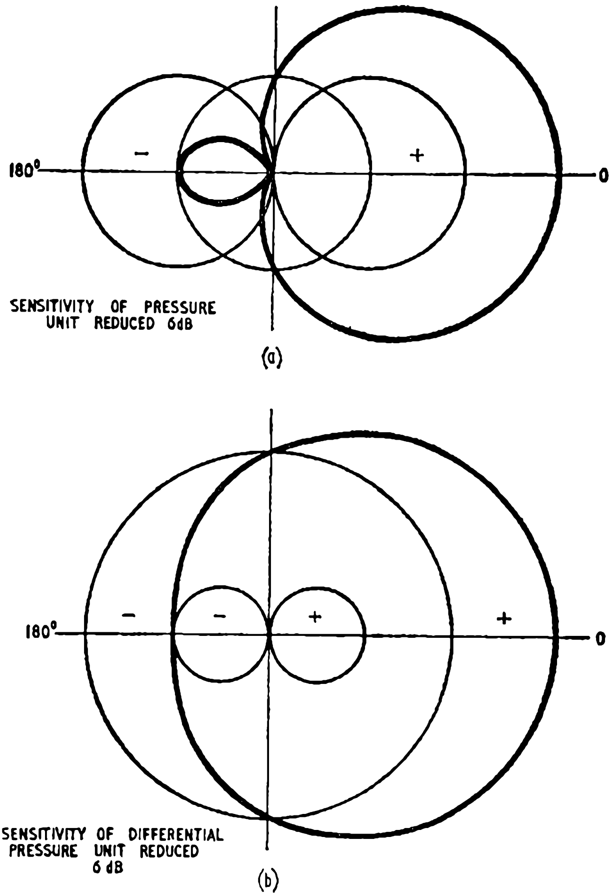


Fig. 6.5.—Two possible directional characteristics for an adjustable cardioid or uni-directional microphone

6.3.2. SINGLE-ELEMENT TYPE

Uni-directional microphones can be divided into two groups or classes. The first, just described, employs two elements of equal sensitivity, a pressure-gradient element and a pressure element, and combines their outputs to derive the desired characteristic. In the second group, only one element is used. The directional characteristic is obtained by applying to the diaphragm of the microphone forces obtained from pressure-gradient operation together with forces obtained from pressure operation. This is the phase-shift method of operation which was described in Chapter 3.

The force F_θ on the diaphragm of a phase-shift microphone is given by

$$F_\theta = 2Ap_1 \, \pi \frac{f}{c} (d_i + d_e \cos \theta)$$

If the external and internal paths introduce equal shift, then

$$F_\theta = 2Ap_1 \, \pi \frac{f}{c} \, d_e (1 + \cos \theta)$$

Thus the force varies with the angle of incidence and the equation for F_θ is of the form

$$F_\theta = a\,(1 + \cos \theta)$$

where $a = 2Ap_1 \, \pi \frac{f}{c} d_e$ is a constant with respect to θ. Hence the directional characteristic is a cardioid.

The uni-directional characteristic is maintained only if the phase shift introduced by the two paths d_e and d_i is equal. This condition is difficult to satisfy over a wide frequency range. If the phase shift introduced by the internal path exceeds that of the external, that is $d_i > d_e$, the directional characteristic is similar to that in Fig. 6.5 (a). Conversely if $d_e > d_i$, the characteristic tends towards an omnidirectional pattern, as in Fig. 6.5 (b).

6.4. Directivity Ratio, Random Efficiency and Directivity Index

These are quantities which indicate the directional discrimination of microphones.

In considering the response of a microphone to indirect sound, it is assumed that the sound waves arrive simultaneously from all directions. This is reasonably true of reverberant sounds but does not necessarily apply to noise, especially where it emanates from a discrete source or from a particular direction. The *directional efficiency* is the energy output due to the simultaneous sounds at all angles expressed as a fraction of the energy which would be obtained from an omni-directional microphone of the same axial sensitivity. As directional efficiency is based on energy, the ratio of the corresponding voltages is the square root of the directional efficiency.

For example, a uni-directional microphone with a cardioid characteristic has a directional efficiency of 1/3 and the voltage response to the indirect sound is $1/\sqrt{3}$ of that obtained from an omni-directional microphone. The directional efficiencies of bi-directional and uni-directional microphones are identical, i.e., 1/3.

In general, uni-directional and bi-directional microphones can, for the same ratio of direct to indirect sound, be used at a distance from the sound source $\sqrt{3}$ times as great as with an omni-directional microphone. Since these microphones discriminate by a factor of 3 against noises arriving from all directions, their use in noisy conditions often results in an improved signal-to-noise ratio. Moreover the microphone can be orientated so that the region of zero response points directly at any discrete source of noise and thus reduces or completely eliminates direct pick-up from such a source. In these circumstances, the improvement in the signal-to-noise ratio is much greater than the factor of 3 which is based on the assumption that the noise is random in direction.

Finally the large angle within which a uni-directional microphone accepts sound without noticeable discrimination makes it possible to use a single microphone to cover widely separated instruments or artists. Microphones of this type are particularly useful on film and television sets where their angle of null response allows the technical operations associated with the production to be carried out without interference with the programme.

7

Omni-directional Microphones

A DESIRABLE FEATURE in early microphones was a large output coupled with high sensitivity, and although there were many physical phenomena that could be used to convert acoustic energy into electrical energy, only the carbon-granule microphone possessed the attributes of simplicity and sensitivity. However, the development of the valve amplifier made possible the use of microphones of much lower sensitivity, and electrodynamic microphones, both moving-coil and ribbon, with their uniformity of response and absence of internal noise, speedily replaced the carbon microphones in use in the broadcasting service. The advent of the miniature valve and miniature components gave a great impetus to microphone development and made available for the first time pre-amplifiers of small size. As a result, electrostatic and phase-shift microphones became practical realities.

7.1. Pressure-operated Moving-coil Microphones

Moving-coil microphones have been used extensively by the BBC for a number of years, particularly in outside broadcasts.

The microphone is virtually a small alternator in which the conductors in the form of a coil are caused to vibrate in a magnetic field provided by a permanent magnet. A light diaphragm to which the coil is rigidly attached on the underside (Fig. 7.1) is set in motion by the acoustic pressures and an alternating voltage e is induced in the coil by virtue of its movement in the magnetic field.

$$e = Blu$$

where B = flux density in the air gap,
l = length of the wire in the coil,
u = velocity of the coil.

Since B and l are constants it follows that

$$e \propto u$$

The microphone will be uniformly sensitive with respect to frequency if the velocity of the mechanical system is independent of the frequency of the acoustic pressures. To keep the mass of the mechanical system to a minimum, aluminium ribbon or wire is used for the coil, the diaphragm being made of a thin sheet of Duralumin with a domed central portion for rigidity. The coil is fixed to the

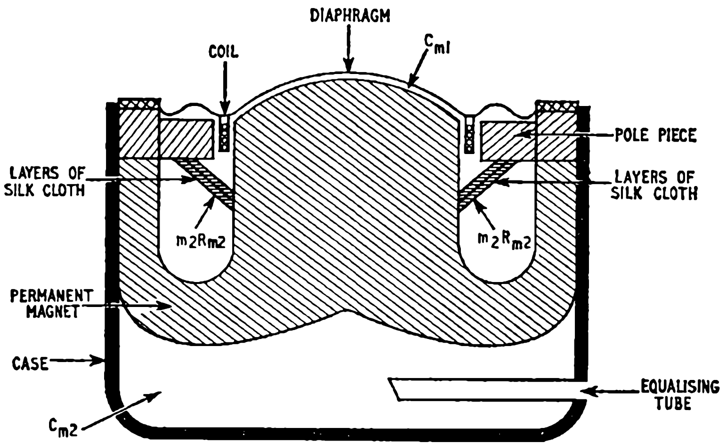


Fig. 7.1.—Schematic cross-section of a moving-coil microphone

diaphragm immediately under the dome. Concentric corrugations in the " flat " part of the diaphragm increase its flexibility and allow the coil to move with a piston-like motion in the air gap without tilting (Fig. 7.1 and Plate 7.1).

In the equivalent electromechanical circuit (Fig. 7.2) the mass of the diaphragm and coil is represented by the inductor m_d, the compliance of the system by C_{md} and the friction and mechanical damping by R_{md}. If an alternating force F_m is applied to the system, the velocity u is

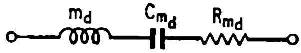
$$u = \frac{F_m}{Z_m}$$

$$= \frac{F_m}{R_{m_d} + \mathrm{j}\left(\omega m_d - \dfrac{1}{\omega C_{m_d}}\right)}$$

The velocity is not constant, but is proportional to the reciprocal of the impedance Z_m. The velocity/frequency curve (Fig. 7.3) is similar in shape to the current/frequency curve of an electrical circuit of this type and has a pronounced peak at resonance when the mechanical impedance is purely resistive.

Since the open-circuit voltage is proportional to the velocity, the frequency response characteristic of the microphone has the same general shape as curve A in Fig. 7.3. If the damping R_m is increased sufficiently, the mechanical impedance becomes sensibly independent of frequency but the velocity decreases and the output of the microphone falls, as in curve B in Fig. 7.3. Thus uniformity of response can only be achieved at the expense of sensitivity; moreover it is difficult

Fig. 7.2.—Electromechanical equivalent circuit of diaphragm



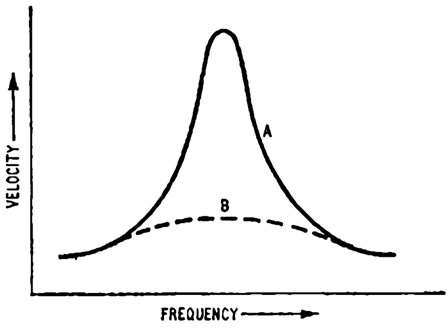


Fig. 7.3.—Frequency response of diaphragm B: with, and A without, damping

to increase R_m since it is associated with the flexing of the circular corrugations and depends on the type of material used for the diaphragm.

7.1.1. INDIRECT DAMPING

Instead of attempting to increase R_m directly, additional mechanical elements are added to form a network whose impedance is approximately constant over the audio-frequency range. These elements can take several forms, one of which is illustrated in Fig. 7.1. Layers of silk cloth are placed in the body of the microphone and so arranged that if the diaphragm attempts to move with high velocity at resonance, an alternating current of air is forced through the holes in the woven silk. The viscous forces which arise as a result of this movement oppose the motion of the diaphragm.

7.1.2. OPERATION

Fig. 7.4 is the electromechanical equivalent circuit of the microphone shown in Fig. 7.1. C_{m_1} is the compliance of the air immediately behind the diaphragm, and since the stiffness of the air in this pocket

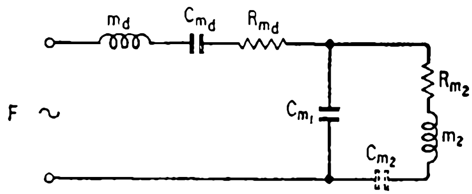


Fig. 7.4.—Equivalent circuit of resistance-controlled diaphragm

increases the stiffness of the diaphragm system, C_{md} and C_{m_1} are arranged in series; m_2 and R_{m_2} represent respectively the mass of air moving through the silk and the viscous damping associated with its movement. C_{m_2} is the compliance of the large volume of air in the case and is so small in comparison to the other reactances that it will be neglected in this analysis.

The branch m_2, R_{m_2} is in parallel with C_{m_1} and forms an anti-resonant circuit. By proper choice of the components, a large value of resistance can be introduced into the circuit to compensate for the fall in the impedance of the series circuit at resonance. Above resonance the diaphragm impedance rises, due to the increase in the mass reactance, while the impedance of the parallel circuit falls, since it behaves as a compliance reactance. Cancellation occurs and the impedance remains sensibly constant. Reactance cancellation also takes place below resonance, for the compliance reactance of the diaphragm which then predominates is reduced by the predominant mass reactance of the parallel circuit.

The series equivalent of the parallel circuit is

$$R_s = \frac{R_{m_2}}{(1 - \omega^2 m_2 C_{m_1}) + (\omega C_{m_1} R_{m_2})^2}$$

$$X_s = \frac{\omega m_2 (1 - \omega^2 m_2 C_{m_1}) - (\omega C_{m_1} R_{m_2})^2}{(1 - \omega^2 m_2 C_{m_2})^2 + (\omega C_{m_1} R_{m_2})^2}$$

The mechanical constants m_d, C_{m_d}, R_{m_d}, C_{m_1}, m_2, R_{m_2} can be so chosen that the absolute value of the mechanical impedance of the whole system is reasonably uniform over a wide frequency range.

Curve A in Fig. 7.5 is a typical pressure frequency response curve of a moving-coil microphone with an undamped diaphragm having its main resonance about the middle of the audio-frequency band. Curve B shows the improvement in the response when the mechanical

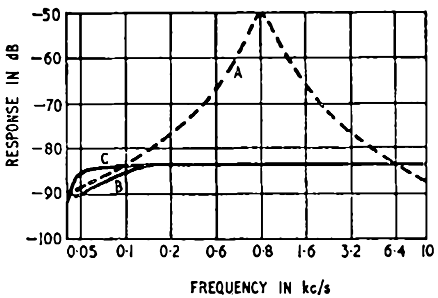


Fig. 7.5.—Response of damped and undamped diaphragm. Reference level—1 V/dyn/cm²

system is indirectly damped by the silk cloth. A comparison of Curve B in Fig. 7.5 with Curve B in Fig. 7.3 shows also that at the high-frequency end of the characteristic, indirect damping of the diaphragm circuit is superior to direct damping.

7.1.3. EQUALISER TUBE

The deviation from the constant velocity characteristic at the low-frequency end of the response curve is due to the increase in the compliance reactance of the diaphragm with decrease in frequency. Although the mass reactance in the parallel circuit does provide some compensation, it is not sufficient to cancel the large increase in the diaphragm's compliance reactance at low frequency, and the output of the microphone falls.

Some improvement in the response can be effected by the addition of an equalising tube (Fig. 7.1) connecting the volume of air in the case to the outer atmosphere. At medium and high frequencies, the mass reactance of the air in the tube is large and prevents acoustic pressures from being transferred to the air in the case and so to the back of the diaphragm. At low frequencies the reactance of the air in the tube falls and acoustic pressures enter the case but the length and diameter of the tube are so chosen that the pressures are shifted by 180°, thus increasing the force on the diaphragm and hence the output at low frequency (Curve C in Fig. 7.5). In improving the low-frequency response, the phase-shifting action of the equalising tube is similar to the phase shift introduced by the port in a phase-inverter type of loudspeaker cabinet.

7.1.4. PERFORMANCE

The response curve in Fig. 7.5 indicates that the open-circuit sensitivity of a moving-coil microphone is lower than that of the electrostatic or crystal types. However, the internal impedance of the microphone is low, of the order of 20 Ω, and a step-up transformer is necessary to couple the microphone to the grid of the first valve. If the turns ratio of the transformer is 1 : n the effective open circuit voltage is increased by a factor n and the impedance by n^2.

Usually, two transformers are employed. The purpose of the first is to match the microphone to a line (300 Ω). This results in a loss in level of 6 dB due to the termination, but a 10 dB gain is achieved by the step-up action of the transformer. This arrangement permits the output of the microphone to be transmitted over fairly long distances without serious attenuation or loss of response. At the receiving end of the line, a second transformer steps up the voltage still further and matches the line to the high resistance of the grid of the first amplifier stage.

Having discussed the operation of the moving-coil microphone generally, we are now in a position to examine in more detail its main components.

7.1.5. DIAPHRAGMS

The diaphragms of moving-coil microphones are in the form of thin stiff plates rather than membranes. The essential difference between the membrane and the plate is that while they both possess mass, the elastic restoring force of the membrane is due entirely to its tension, while that of the plate depends on its dimensions and on the material from which it is made, no tension being applied. Pressing out the central portion of the diaphragm into a dome increases the stiffness to such an extent that it acts as a discrete or lumped mass. The harmonic modes of this portion of the diaphragm are then in the region of 15,000 c/s and lie outside the working range of the microphone.

The domed disc is not the only way of achieving a light stiff system. For example, the diaphragm in Fig. 7.6 (a) is of balsa wood, 2·5 mm thick, enclosed between two thin sheets of aluminium foil. Enamelled aluminium wire is used for the coil which is wound on a

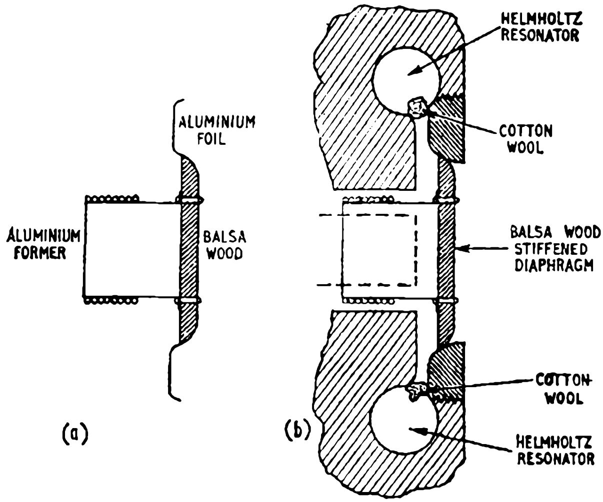


Fig. 7.6.—(a) Microphone diaphragm stiffened with balsa wood; (b) schematic cross-section of moving-coil microphone with eddy current and Helmholtz resonator damping

thin aluminium former riveted to the diaphragm. The complete assembly is waxed and combines high rigidity with lightness. The outer surround of the aluminium foil is clamped at its edge in a stretched condition by the action of two clamping rings (not shown in the diagram).

The most commonly used materials for the diaphragms of moving-coil microphones are aluminium alloys such as Duralumin but plastics and paper have also been used successfully.

7.1.6. DIAPHRAGM DAMPING SYSTEMS

We have seen that silk screens can be used in microphones to provide an acoustic resistance to limit the velocity of the diaphragm at resonance. Silk is a particularly useful material for this purpose, for the ratio of resistance to inertance is high. The magnitude of the resistance depends on the number of holes in the material and on their size and shape. The acoustic resistance R_a for a square centimetre of sheer silk is

$$R_a \simeq 4n$$

where n is the number of layers of silk.

The acoustic resistance is due to the viscosity of the air, which can be likened to friction between adjacent layers. If air is caused to flow through a narrow slot or small tube, the velocity of the air in the tube varies from a maximum at the centre to zero at the tube wall. Thus the various layers of air in the tube must slide over each other in their motion and this gives rise to a viscous loss. The smaller the diameter of the tube, the higher is the resistance.

The air moving in the tube possesses mass and therefore there is a reactive component associated with the acoustic resistance. This reactance decreases as the diameter of the tube decreases, and by suitable choice of diameter the ratio of inertance to resistance may be made to have almost any desired value.

The slot is another useful form of acoustic impedance and has been used in many successful designs to damp the main resonance of the diaphragm system of moving-coil microphones.

The diaphragm of the microphone shown in Fig. 7.7 is of the conventional type already described, and the coil associated with it is of aluminium ribbon wound on edge. The diaphragm-and-coil assembly has mass m_d, compliance C_{md} and resistance R_{md} such that

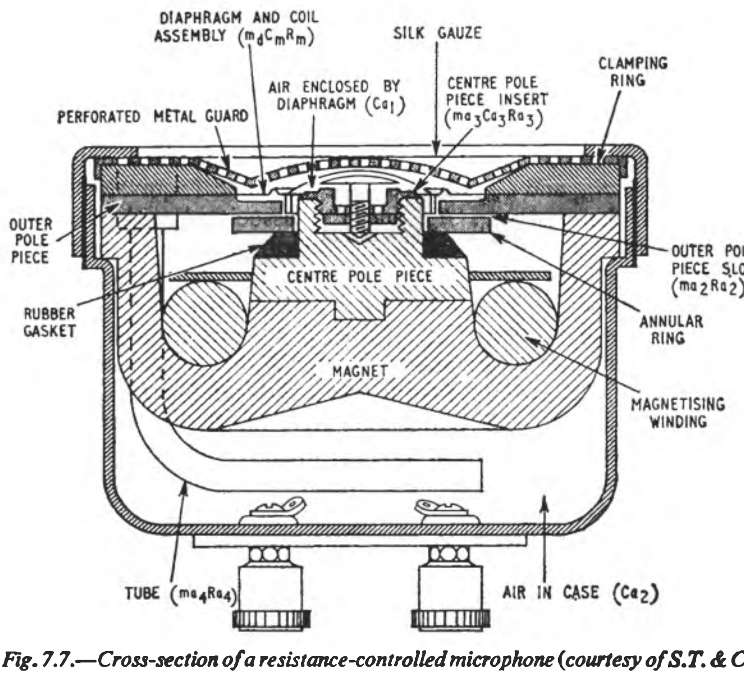


Fig. 7.7.—Cross-section of a resistance-controlled microphone (courtesy of S.T. & C.)

if no other elements were present, the velocity/frequency characteristic would have a sharp peak about 2,000 c/s. This resonance is avoided by the provision of slots and cavities behind the diaphragm; because of the acoustical impedances introduced, the velocity for a constant applied force is reasonably uniform through the frequency

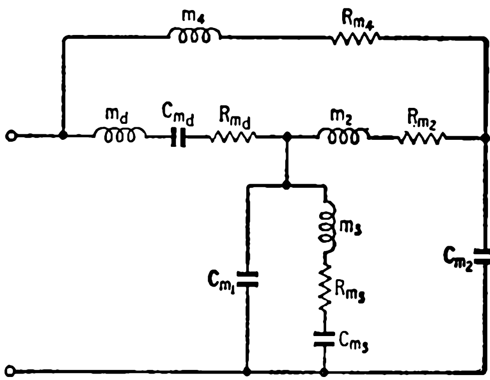


Fig. 7.8.—Equivalent circuit of a resistance-controlled microphone

range. The equivalent circuit for the composite system of acoustical and mechanical elements is shown in Fig. 7.8.

The main resonance of the diaphragm is smoothed out by the effect of m_2, R_{m_2} and C_{m_2} which are the mechanical equivalents of the acoustical elements m_{a_2}, R_{a_2} and C_{a_2} associated with the outer pole-piece slot and the air in the microphone casing. In the absence of the other acoustical correction circuits, the velocity would decrease below 200 c/s, with sharp irregularities above 2,000 c/s caused by m_2, R_{m_2} and C_{m_2}. These deviations from a constant-velocity characteristic are shown by the dotted curves of Fig. 7.9 which is a graph of output for a constant applied pressure, plotted against frequency. The centre pole-piece insert represented in the equivalent circuit by m_3, R_{m_3} and C_{m_3} smooths out the variations in the response above 2,000 c/s, leaving only the minor variations shown by the full-line curve.

The function of the equalising tube m_{a_4}, R_{a_4} is to improve the output of the microphone at the low frequencies. At frequencies lower than 200 c/s there is a partial resonance between the positive reactance of the air in the tube and the negative or compliance reactance of the air in the case. The acoustical resistance associated with the tube is small and an appreciable pressure is produced in the casing, the phase being such that the velocity of the diaphragm is augmented and the low-frequency response maintained, as shown in Fig. 7.9.

The microphone just described was used extensively by the BBC in the 1930s and for a number of years afterwards, particularly on outside broadcasts. It is now obsolete but it is worthy of study because it is an excellent example of the application of the principles we have been discussing.

A third method, used in an early microphone, of limiting the velocity of the diaphragm at resonance, is illustrated in Fig. 7.6. The diaphragm of this microphone is stiffened with balsa wood (Section 7.1.5) and the coil is wound on a thin aluminium former. The main resonance of the diaphragm and moving-coil system is controlled in two ways: first, by the eddy-current damping produced by the movement of the aluminium coil former in the magnetic field and

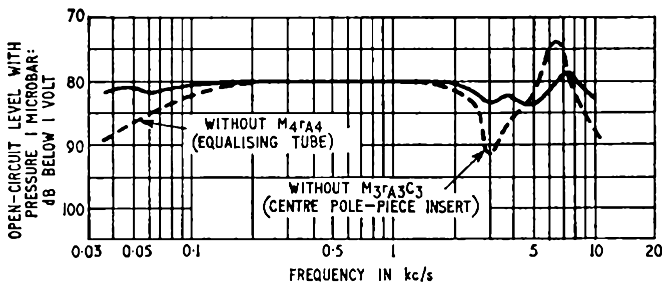


Fig. 7.9.—Pressure/frequency calibration showing the effect on the response of the acoustical correcting networks (courtesy of S.T. & C.)

secondly, by the acoustic resistance provided by the Helmholtz resonator situated immediately behind the diaphragm (Fig. 7.6 (b)). The annular cavity of the resonator has a narrow ring-shaped neck and is tuned to resonate at the same frequency as the diaphragm-and-coil assembly. Cotton wool is used to damp the resonator and increase the acoustic resistance.

7.1.7. MICROPHONE HOUSINGS

In discussing diffraction effects in Chapter 5, we saw that the size and shape of the microphone case can influence its axial response and affect its directional characteristic. In general the shapes of microphone housings are so irregular that it is impossible to predict theoretically their effect on a sound field. Such a situation is obviously undesirable, and Plate 7.2 and Fig. 7.10 show the general construction of a practical type of microphone housing whose diffraction effect can be predicted in advance.

The microphone is designed for use with the diaphragm uppermost. Thus sounds in the horizontal plane irrespective of their angle of

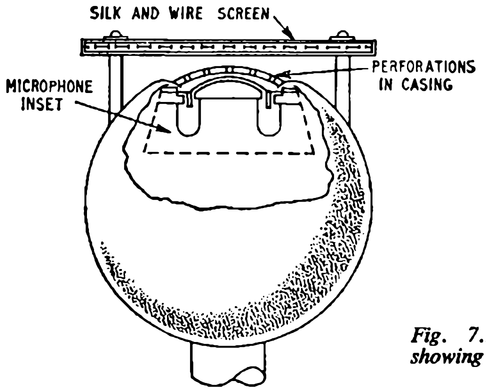


Fig. 7.10.—Schematic cross-section showing the position of the microphone inset

incidence on the microphone have free access to the diaphragm. The diaphragm must be small in comparison to the wavelength of the highest frequency, so that there is no appreciable loss in response due to phase differences across it. At the top of the microphone there is a disc-like screen mounted on small pillars so that its centre is approximately $\frac{1}{8}$ in. from the perforations in the casing above the diaphragm.

If the diaphragm is exposed directly to sound waves normal to its surface, the size and shape of the housing enhance the output at about 2,400 c/s, for at this frequency $D/\lambda \simeq 0{\cdot}5$ for a sphere of $2\frac{1}{2}$ in. diameter. From the graph in Fig. 5.1 the ratio of p/p_0 is 1·6 and the increase in output is of the order of 2 dB (20 log 1·6). The good omni-directional characteristic of this microphone in the vertical plane is due to the properties of diffraction and attenuation possessed by the screen, which is made of silk gauze enclosed within layers of fine wire mesh and bounded by a rigid metal rim.

At frequencies below 1,000 c/s, sound at any angle has free access to the diaphragm. At higher audio frequencies, sound impinging on the microphone from above is obstructed by the screen. Some sound passes through the silk screen due to the high pressure on the top surface but is attenuated in the process. As a result, the sound pressure which reaches the diaphragm is that of the free-wave field.

High-frequency sound impinging on the microphone from below is partially reflected by the underside of the screen on to the diaphragm, maintaining the effective diaphragm pressure, despite the obstruction presented to the wave by the microphone casing.

7.1.8. INCONSPICUOUS MOVING-COIL MICROPHONE

Occasionally in television it may be necessary for the microphone to appear in shot and it is desirable that it should be housed in an inconspicuous case that does not detract from the microphone's performance. Moving-coil microphones intended for close speech have been produced with the microphone element in the base of the stand, the sound pressures reaching the diaphragm through a vertical tube several feet long.

The inlet to the tube is small, the primary object being to reduce diffraction effects and so maintain an omni-directional characteristic at all frequencies. Unfortunately the response is obtained at the expense of sensitivity, but this is not necessarily serious since the microphone is usually intended for close-range operation. The tube must be correctly terminated, otherwise multiple reflections would occur, making the response curve irregular and ragged. Like eigentones in a studio, the irregularities become more closely spaced as frequency increases and if the tube is insufficiently damped, both frequency response and transient response may be adversely affected.

7.2. Electrostatic Microphones

Electrostatic microphones made a brief appearance in British broadcasting in the 1930s but there were various operational difficulties associated with their use. Heavy and bulky valve amplifiers were required and in some cases a special battery supply was necessary. The insulation in the high-impedance circuit was also a frequent source of trouble. The modern electrostatic microphone with pre-amplifier is small and compact and compares favourably with other types of microphones for reliability and consistency of performance.

The microphone depends for its action on the variation of capacitance between a tightly stretched metallic membrane situated in close proximity to a rigid metal back plate. The back plate (Fig. 7.11 (a)) is insulated from the remainder of the microphone and a polarising voltage is applied between it and the metallic diaphragm. Sound pressures acting on the diaphragm cause it to vibrate about its

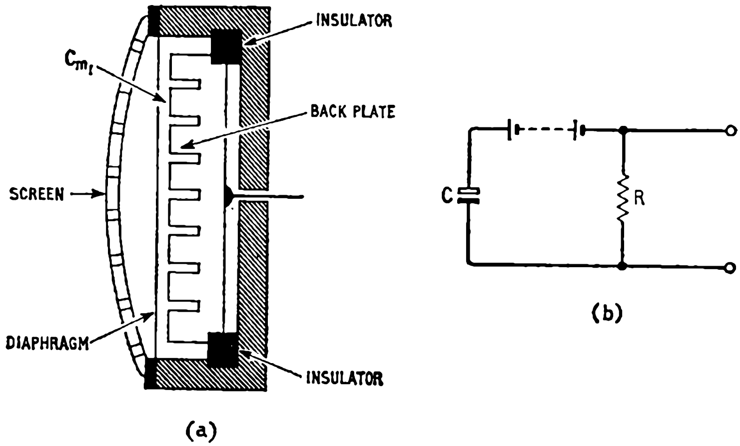


Fig. 7.11.—(a) Schematic cross-section of an electrostatic microphone; (b) simplified circuit showing the polarising supply

mean position and so vary the capacitance C between the back plate and the diaphragm, as shown in Fig. 7.11 (b).

In the absence of acoustic pressures the voltage V across the capacitor is given by

$$V = \frac{Q}{C}$$

where Q is the charge stored in C supplied from the polarising source via the resistor R.

Sound pressures acting on the diaphragm vary the spacing s between it and the back plate by an amount $\pm \Delta s$. The spacing at any instant is $s \pm \Delta s$ and the capacitance at any instant is $C \mp \Delta C$

$$C \mp \Delta C = \frac{a}{s \pm \Delta s}$$

where a is a constant.

If the resistance R is large, the charge Q is virtually constant: then

$$C \mp \Delta C = \frac{a}{s \pm \Delta s}$$
$$= \frac{Q}{V \pm \Delta V}$$

Since a and Q are constants,

$$\Delta V \propto \Delta s$$

that is, the variations in voltage across the capacitor which constitute the output of the microphone are proportional to the displacement of the diaphragm about its mean position.

7.2.1. CONSTANT-AMPLITUDE SYSTEM

If the microphone is to be uniformly sensitive with respect to frequency, the mechanical system must be so arranged that the displacement of the diaphragm is independent of the frequency of the acoustic pressures. The electrostatic microphone is therefore classed as a constant-amplitude type.

In the electromechanical equivalent circuit in Fig. 7.12 (a) the effective mass of the diaphragm is represented by m_d, the compliance

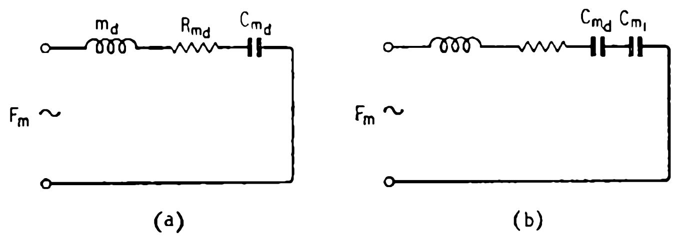


Fig. 7.12.—(a) Equivalent circuit of diaphragm of electrostatic microphone; (b) equivalent circuit showing the compliance of the air space in series with the diaphragm impedance

by C_{md} and the mechanical damping of the system by R_{md}. Assuming an alternating force F_m is applied to the diaphragm, its velocity u is given by

$$u = \frac{F_m}{Z_m}$$

$$= \frac{F_m}{R_{md} + \mathrm{j}\left(\omega m_d - \dfrac{1}{\omega C_{md}}\right)}$$

If the frequencies of the acoustic pressures are well below the natural frequency of the system, the mass reactance ωm_d and the mechanical damping R_m can be neglected in comparison to the large compliance reactance $\dfrac{1}{\omega C_{md}}$.

Hence at frequencies below the natural frequency of the diaphragm, the velocity u is given by

$$u = F_m \bigg/ \frac{1}{j\omega C_{md}}$$

If the phase relationship between velocity and driving force is unimportant then u is given by

$$u = F_m \, \omega \, C_{md}$$

The displacement x of the diaphragm is then

$$\begin{aligned} x &= \frac{u}{\omega} \\ &= \frac{F_m \, \omega \, C_{md}}{\omega} \\ &= F_m \, C_{md} \end{aligned}$$

Since C_m is a constant and if F_m is independent of frequency, the displacement x is independent of frequency and the condition for uniform response is satisfied.

7.2.2. DIAPHRAGM RESONANCE

The output of the microphone is independent of frequency provided the mechanical system is compliance controlled, and this condition is satisfied if the fundamental resonance of the diaphragm system is high and lies outside the frequency range for which the microphone is intended.

The fundamental frequency of the diaphragm depends on its mass and tension and is given by

$$f_o = \frac{0{\cdot}382}{r} \sqrt{\frac{T}{m}}$$

where r = radius of the diaphragm in cm,
m = mass per unit area in g/cm^2,
T = tension in dyn/cm.

The frequency of resonance can be increased by increasing the tension T or by decreasing the radius r. An increase in T or a decrease in r will result in a reduction of the amplitude of movement of the diaphragm and hence in a drop in the output. As before, uniformity of response can only be obtained by a reduction in sensitivity.

Obviously the diaphragm must be light, otherwise the excessive tension which would be required to obtain a high resonance frequency would result in fracture. With the foil materials available at present, the natural frequency of a practical diaphragm cannot exceed 8,000 c/s by the action of tension alone. However the small volume of air between the diaphragm and back plate raises the natural frequency of the system. Movement of the diaphragm compresses or rarifies the air in the pocket, thus adding perceptibly to the stiffness of the system as a whole. The compliance C_{m_1} of the air space is effectively in series with the compliance C_{md} of the diaphragm itself, as shown in Fig. 7.12 (b).

7.2.3. STIFFNESS AND SENSITIVITY

To obtain a large output for small movements of the diaphragm the changes in capacitance must be large: close spacing between the electrodes is therefore essential. As we have seen, this close spacing increases the stiffness of the system and may result in loss of

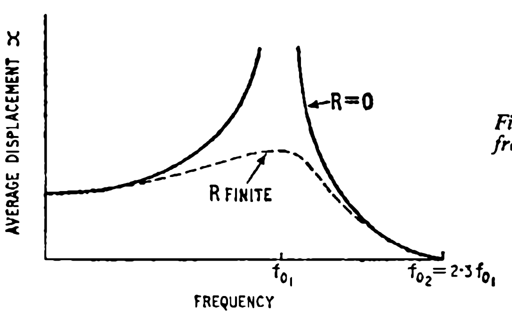


Fig. 7.13.—Displacement/frequency curves with and without damping

sensitivity. If the back plate has a smooth unbroken surface, the volume of air enclosed is very small and the stiffness and resonance frequency may be unnecessarily high. Stiffness must be reduced if sensitivity is to be maintained, but this should not be done by increasing the spacing between diaphragm and back plate but rather by providing additional compression spaces. In the older types of microphones, deep holes of small diameter were drilled in the back plate to increase the volume of air in the compression space, but today, grooves or separate compression chambers can also be used.

Reducing the stiffness of the system in order to maintain sensitivity usually results in moving the natural frequency of the diaphragm into or near the upper end of the audio-frequency range. The output increases as the resonance frequency is approached, the amplitude of movement of the diaphragm being limited only by the mechanical damping (Fig. 7.13). Increasing the damping reduces the peak and extends the range of uniform output up to and beyond the first natural frequency of the diaphragm.

The deep holes and grooves in the back plate not only decrease the stiffness of the system but are so arranged that the viscous resistance

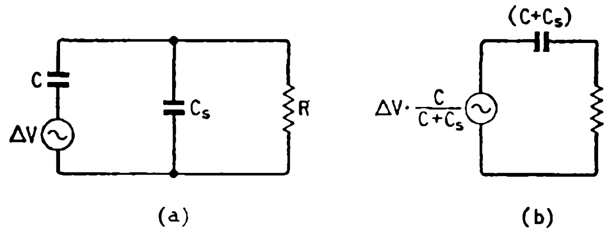


Fig. 7.14.—(a) Equivalent circuit of microphone and stray capacitance; (b) Thévenin equivalent of (a)

associated with the movement of air in the holes or grooves damps the diaphragm resonance. Alternatively, a ring-shaped cavity behind the diaphragm with a suitably designed aperture acting as a Helmholtz resonator can be used to provide the acoustic damping required and so limit the excursions of the diaphragm.

At frequencies beyond resonance, the response of the microphone falls, for the mass reactance of the diaphragm predominates and increases with frequency. The displacement x is then

$$x = \frac{F_m}{\omega^2 m_d}$$

that is, $x \propto \frac{1}{f^2}$. Above resonance, the response falls rapidly at a rate of 12 dB per octave.

7.2.4. THE LOWER LIMIT OF THE FREQUENCY RESPONSE

When the microphone is placed in its case or housing, stray capacitance is introduced by virtue of the leads which connect it to the amplifier and by the proximity of the case. This stray capacitance C_s, which is in parallel with that of the microphone itself (Fig. 7.14 (a)), reduces the output and in conjunction with the resistor R sets a limit to the low-frequency response. The microphone is equivalent to a generator in series with a capacitor. Using Thévenin's theorem, the circuit of Fig. 7.14 (a) can be replaced by the simplified equivalent of Fig. 7.14 (b).

The current through the resistor R is given by

$$i = \frac{\Delta V \dfrac{C}{C + C_s}}{R + \dfrac{1}{j\omega\,(C + C_s)}}$$

The voltage across R is

$$V_R = \frac{\Delta V \dfrac{C}{C + C_s}}{R + \dfrac{1}{j\omega\,(C + C_s)}}\,R$$

$$V_R = \Delta V \frac{C}{C + C_s}\;\frac{1}{1 + \dfrac{1}{j\omega R\,(C + C_s)}}$$

The voltage across R, which is the output of the microphone, is made up of a constant

$$\Delta V \frac{C}{C + C_s}$$

multiplied by a factor

$$\frac{1}{1 + \dfrac{1}{j\omega R\,(C + C_s)}}$$

which involves frequency.

If the frequency of the sound is high,

$$\frac{1}{j\omega R\,(C + C_s)}$$

is small, and the output of the microphone is constant at

$$\Delta V \frac{C}{C + C_s}$$

The effect of the stray capacitance C_s is to reduce the output by a factor

$$\frac{C}{C + C_s}$$

This is what would be expected from the action of the two capacitors as a potential divider.

At low frequencies

$$\frac{1}{j\omega R\,(C + C_s)}$$

is large and the output of the microphone falls (Fig. 7.15), and is 3 dB down at the frequency which makes $\omega R\,(C + C_s) = 1$, i.e.,

$$\frac{1}{1 + \dfrac{1}{j\omega R\,(C + C_s)}} = \frac{1}{1 - j} = \frac{1}{\sqrt{2}}$$

7.2.5. CONSIDERATIONS GOVERNING THE VALUE OF R

If we accept a 3-dB drop in the response at the frequency which marks the lower limit of the operating range of the microphone, we

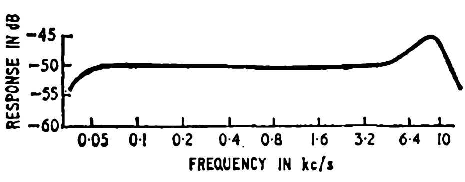


Fig. 7.15.—Typical response characteristic showing reduced sensitivity at low frequency. Reference level—1 V/dyn/cm²

can calculate the value of R, provided C and C_s are known. If the lowest frequency to be reproduced is 30 c/s, the static capacitance C of the microphone is 80pF and the stray capacitance C_s is 20pF, we have

$$R\,(C + C_s) = \frac{1}{\omega} \text{ or } 100R = \frac{10^{12}}{2 \times 3{\cdot}14 \times 30}$$

This gives a value for R of over 50 MΩ.

Such a large value requires that the insulation of the microphone and the components in the grid circuit of the pre-amplifier be of a high order. A special valve or a specially selected gas-free specimen will be needed for the first stage.

7.2.6. DYNAMIC RANGE

Having discussed the frequency range over which the microphone is uniformly sensitive, it is now necessary to consider the factor which tends to limit its dynamic range. The range of intensities is limited at the lower level by the self-noise of the microphone. This noise is due to the thermal agitation of the electrons in the load resistor and in the material used to insulate the back plate from the diaphragm.

Electrostatic microphones are comparatively insensitive and if a satisfactory dynamic range is to be obtained, they must operate with a high value of load resistance. The increased output which results offsets the noise produced within the resistor itself and up to a point the dynamic range can be improved by increasing the value of the load resistor.

The higher level of the dynamic range is set by the acceptable distortion in the signal. In this connection, it should be remembered that the principle of operation of the microphone is non-linear, for the output voltage is inversely, not directly, proportional to the change in capacitance. The percentage of second harmonic produced by this effect is:

$$\text{percentage of second harmonic} \simeq 50\,\frac{\Delta V}{V}$$

where ΔV is the peak open-circuit voltage produced by the microphone and V is the steady polarising potential. The second harmonic distortion obtained is small and may only amount to as little as 1 per cent even for sounds of high intensity.

7.2.7. VIBRATIONAL MODES OF THE DIAPHRAGM

The output ΔV depends on the average displacement of the diaphragm and this is not necessarily proportional to pressure except for very small deflections. Furthermore, the displacement of the diaphragm at the resonant frequencies may be quite large, yet the average displacement may be small, for under certain circumstances different elements of the diaphragm may move in anti-phase. At frequencies below and up to the fundamental frequency f_{01} (Fig. 7.16 (a) and (a′)), all parts of the diaphragm move in the same direction at the same instant in time. The average displacement is

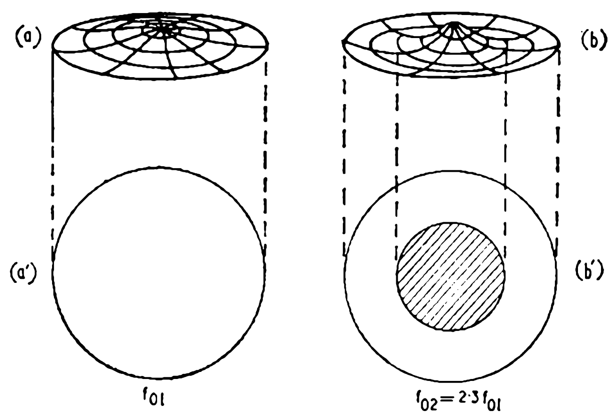


Fig. 7.16.—Modes of vibration of stretched circular membrane. (a) First natural frequency: all parts of the membrane are displaced in the same phase. (b) Second natural frequency with nodal circle. Shaded portions of the membrane move in opposite phase to the unshaded

large. As the frequency of the driving force is increased, a nodal circle appears at the outer edge of the diaphragm, and as the frequency is further increased, the nodal circle decreases in size and moves away from the edge towards the centre of the diaphragm. This circular central portion of the membrane (Fig. 7.16 (b′)) moves in anti-phase to the ring-shaped outer portion and the average displacement decreases and is zero at the second overtone $f_{02} = 2{\cdot}3f_{01}$. Higher vibrational modes of this type are characterised by more than one nodal circle. Other modes can also be induced in which the nodal points lie on diameters and complex vibrations combining both the nodal circles and nodal diameter modes are possible.

7.2.8. OPERATING CONDITIONS

Because of the high impedance of the microphone, careful shielding is necessary if electrostatic pick-up of hum is to be avoided. Furthermore, a long run of screened cable cannot be used to connect the microphone to the amplifier. This is because the capacitance of ordinary shielded cable is of the order of 30 pF/ft and even a short length will seriously reduce sensitivity, since the capacitance of the cable adds to the stray capacitance C_s across the microphone. This is discussed in Section 7.2.4. It is therefore necessary to provide at least one stage of amplification in the immediate vicinity of the microphone.

Dampness or high humidity can cause leakage across the insulating surfaces and special precautions may be necessary. For example, the space between the diaphragm and the back plate of the microphone is so small that dust and moisture droplets, which might reduce insulation or cause unwanted noise, are prevented from entering. In some microphones this space is hermetically sealed but the air within the space is maintained at atmospheric pressure by the provision of an auxiliary diaphragm of thin rubber which permits changes in the atmospheric pressure to be conveyed to the air in the case. High temperatures should also be avoided, for they tend to reduce the resistivity of the insulation and thus may seriously affect the tension of the diaphragm. In spite of this difficulty, electrostatic microphones have been produced which operate quite satisfactorily even when they are exposed to the heat from the high-power lights commonly used in television studios.

7.3. Piezo-electric Effect of Crystals

We have already seen that electrodynamic microphones make use of the movement of a conductor or conductors in a magnetic field to transform acoustical pressure into electrical voltages. Another important group of microphones employs crystals with piezo-electric properties as their transducing element. The piezo-electric effect is a phenomenon manifested by some crystals when subjected to mechanical strains or to electric fields: the term is derived from the Greek word " piezein " meaning " to press ". Under certain conditions, mechanical strains can produce opposite electric charges on two different faces of a given crystal. With crystal microphones, the strains are produced by air pressures. Conversely, electric fields applied to different faces of a crystal can cause it to expand along one axis and contract along another.

If a slice of quartz cut from a crystal in a certain way has a force applied to its ends (Fig. 7.17), placing the crystal in tension, electrical charges will appear on the surface A of the crystal and charges of

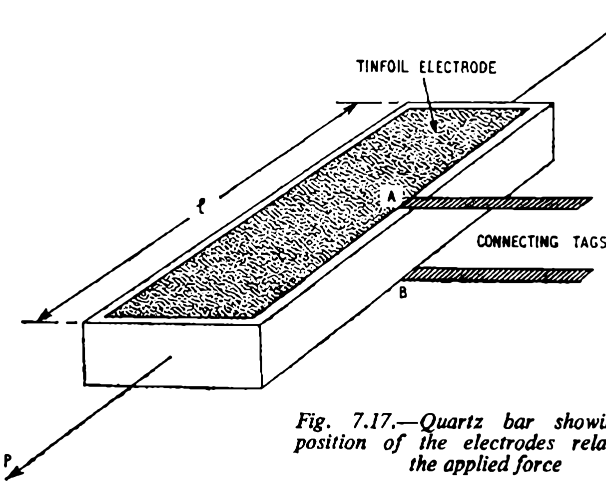


Fig. 7.17.—Quartz bar showing the position of the electrodes relative to the applied force

opposite polarity on the under surface B. If the force p is reversed so that the crystal is now in compression, the polarity of the charges on the two faces will also be reversed. If the force is of an alternating nature, an alternating voltage can be obtained from the two faces if suitable electrodes, usually tin foil, are cemented to the surfaces. This is called the *direct piezo effect* and was first observed by the brothers J. and P. Curie, as early as 1880. In 1881 Lippman, by a purely philosophical argument based on the conservation of charge, predicted the inverse effect, i.e., if an alternating voltage is applied to the electrodes the crystal will vibrate in sympathy with it, expanding and contracting along the length dimension l. This inverse effect was verified experimentally in the same year by the Curie brothers who used a microscope to observe the extensions and contractions of the crystal. The discovery was for many years regarded merely as a scientific curiosity and found little practical application, although the Curies had shown that it was possible to measure either force or voltage by using the direct or indirect effect.

During the 1914-18 war, Professor Langevin at the suggestion of the French Government turned his attention to the problem of locating submarines under water. After many experiments he finally produced an under-water "loudspeaker" working on the inverse piezo-electric effect. The device consisted of a number of quartz plates bonded to two steel sheets, and when it was suitably energised, powerful longitudinal waves at a frequency of 40,000 c/s could be produced in water. The war was over before the apparatus was perfected but the equipment was successfully employed as the first sonic depth finder.

Some ceramics and certain crystals other than quartz possess piezo-electric properties. Of these, the one most commonly used in

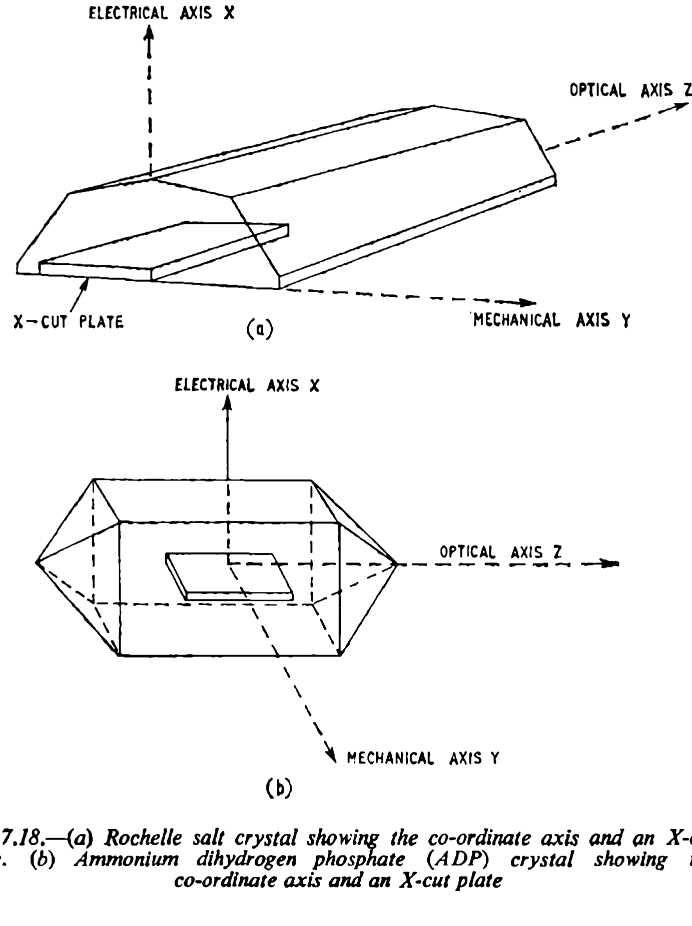


Fig. 7.18.—(a) Rochelle salt crystal showing the co-ordinate axis and an X-cut plate. (b) Ammonium dihydrogen phosphate (ADP) crystal showing the co-ordinate axis and an X-cut plate

the manufacture of crystal microphones is Rochelle salt. It is cheaper than quartz and superior to it in its piezo-electric effect, the charge for a comparable displacement being about one thousand times greater than in quartz. Unfortunately Rochelle crystals deteriorate in the presence of moisture or in conditions of high humidity, and if they are subjected to temperature in excess of 130°F, the piezo-electric effect is destroyed.

Plates cut from synthetic crystals of ammonium dihydrogen phosphate (ADP crystals) are less sensitive than Rochelle salt but will withstand temperatures as high as 212°F and can be employed as the transducing element in hydrophones. For all normal conditions in which microphones operate, the Rochelle crystals are perfectly satisfactory. It is interesting to note, however, that in very recent years, ceramic elements of lead zirconate have been used for gramophone pick-ups. These have a sensitivity comparable to that of Rochelle salt and are far less susceptible to heat or humidity.

Rochelle salt is a double tartrate of potassium and sodium and its crystals belong to the orthorhombic spenoidal group. The common form is the half-crystal shown in Fig. 7.18 (a). The axis along the length of the crystal is the Z or optical axis. That through the apex of the prism is the electrical or X axis, while the third axis, the Y or mechanical axis, is at right angles to the other two.

An X-cut plate, sometimes referred to as a *Curie cut*, is one whose major surfaces are in the YZ plane, the thickness dimension being parallel to the electrical or X axis. The location of such a plate relative to the Rochelle and ADP crystals is shown in Fig. 7.18. If an X-cut plate from a Rochelle or ADP crystal has one edge firmly fixed and a force P is applied to the opposite edge, the plate will distort as depicted in Fig. 7.19 (a) and (b). As a result of the distortion the diagonal AC will either increase or decrease in length, depending on the direction of the force. A potential difference (p.d.) will be developed between the opposite faces of the plate and the polarity of this potential difference will reverse if the force is reversed.

The physical distortion which this type of force produces is called a shear strain and results from the various layers of the material sliding over each other, as shown in Fig. 7.19 (c), under the action of the shear force P. It is for this reason that X-cut plates are sometimes called " shear " plates.

In a practical microphone it might be difficult or inconvenient to subject the crystal element to shear, and a longitudinal or expander bar might prove a more suitable arrangement. Such a bar could be obtained from an X-cut plate by cutting from it a rectangular section,

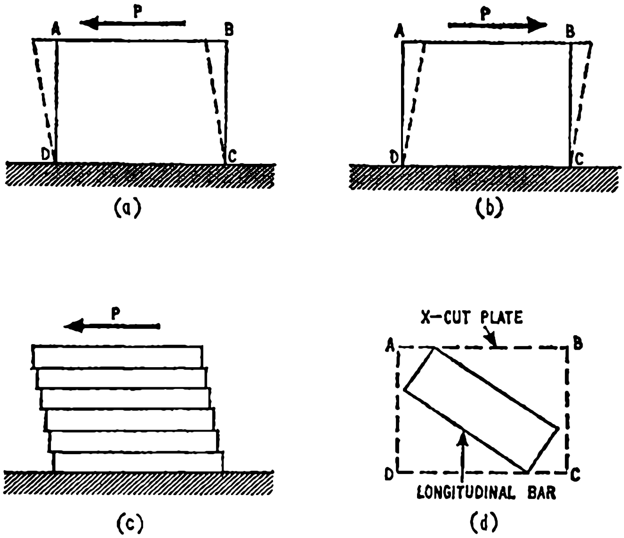


Fig. 7.19.—(a) and (b) Distortion of a plate under shear stress; (c) shear force causes layers of material to slide over each other; (d) method of cutting an expander bar from an X-cut plate

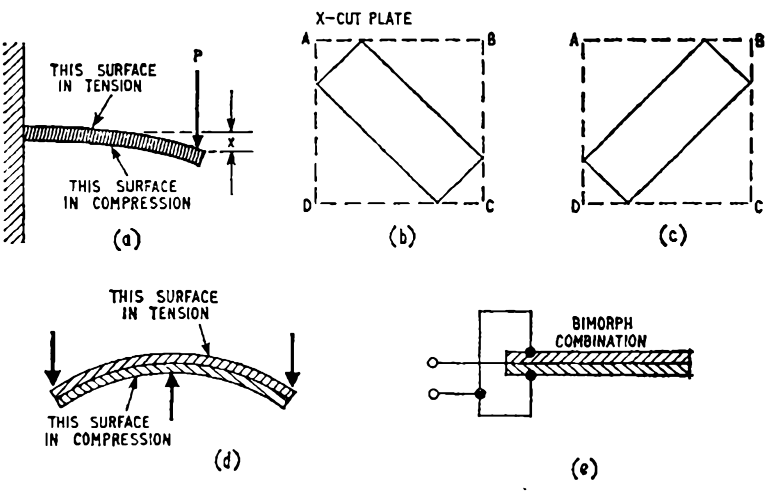


Fig. 7.20.—(a) Cantilever showing tension and compression surfaces; (b) and (c) expander bars, for bimorph combination, cut from opposite diagonals of X-cut plates; (d) bimorph under bending stress showing compression and tension surfaces; (e) method of connecting electrodes in a bimorph combination

as in Fig. 7.19 (d), whose main dimension is parallel to one or other of the diagonals AC and BD. When this bar is subject to tension or compression along its length, piezo-electric charges are produced.

The principal disadvantage of the expander bar is its high mechanical impedance which results in a serious mismatch in air, and as a consequence it is not often employed as a transducing element in a microphone. It can be used to advantage, however, in liquids, since their specific acoustic impedance is large. In such a medium a satisfactory match is obtained and energy can be transferred efficiently.

The mechanical impedance of the bar can be reduced if it is operated as a cantilever. When loaded at the free end, the cantilever deflects as shown in Fig. 7.20 (a), putting one surface of the bar in tension and the other in compression. A longitudinal bar loaded in this way produces no electrical output, for the piezo-electric potentials generated by tension cancel those resulting from compression.

If however two bars cut from X-cut plates on opposite diagonals, as in Fig. 7.20 (b) and (c), are firmly cemented together, a composite bar is formed and is called a *bimorph*. When the combination is bent, as shown in Fig. 7.20 (d), one bar is wholly in tension and the other wholly in compression. Appropriate connection to the electrodes, as in Fig. 7.20 (e), results in an electrical output proportional to the strain. The bimorph arrangement is the equivalent of two sources of voltage connected in parallel. This combination of crystals, first used by the Curies, is called a *bimorph bender* and has many applications.

7.4. Crystal Microphones

A cross-sectional view of a diaphragm-actuated crystal microphone employing a bimorph bender as a cantilever is shown in Fig. 7.21 (a). The diaphragm acts as a coupling unit between the low impedance of the air and the relatively high impedance of the crystal. If the diaphragm and case of the microphone are large compared to the wavelength, diffraction effects may adversely affect the performance of the microphone.

It is possible to dispense with the diaphragm and still obtain an acceptable output if two bimorph benders are combined to form a single sound cell (Fig. 7.21 (b)). In this way diffraction effects are reduced for the dimensions of the bimorphs can be of the order of 0·5 in^2 and 0·05 in. thick. When a sound cell of this type is subjected to an acoustic pressure, as in Fig. 7.21 (c), the deformation of the crystals is such as to place the external plates in each bimorph in compression and the internal plates in tension. The voltages obtained from the two bimorph elements can be connected in series or in parallel as desired.

Microphones that give a large output have been made using as many as 24 of these sound cells connected in various series and parallel groups.

The bimorph bender is not the only arrangement of crystals available to the designer. Twister bimorphs are in common use and

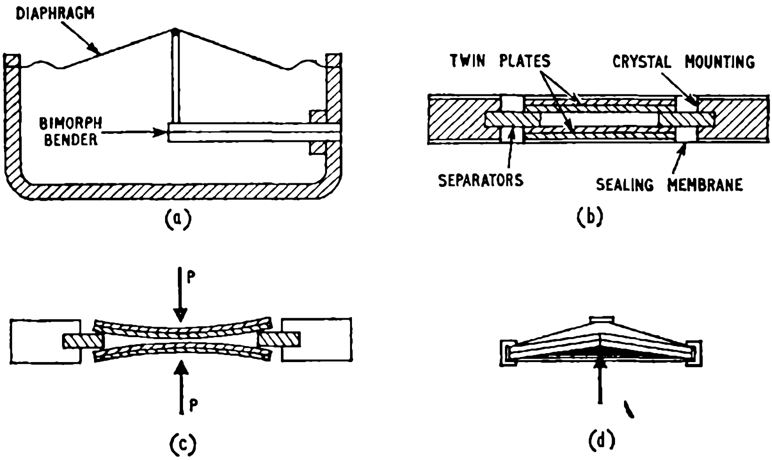


Fig. 7.21.—(a) Schematic cross-section of a crystal microphone; (b) cross-section of crystal sound-cell; (c) sound-cell subjected to acoustic pressure; (d) two X-cut plates combined to make a bender bimorph

are made by combining two X-cut plates, suitably orientated to give an electrical output. The method of loading and fixing the combination is illustrated in Fig. 7.21 (d).

A discussion on the advantages and disadvantages of the various combinations of crystals and the methods of loading them is beyond the scope of this book. What follows is a simple treatment since it involves displacement in one direction only and does not make use of any lever system or other mechanical advantage.

An expander bar of length l, width w and thickness t, is fixed at one end and a force P is applied at the other end, as shown in Fig. 7.22, producing a displacement x. It will be seen that the electrode system is virtually a capacitor C_c with the crystal as the dielectric.

The potential difference V between the electrodes is

$$V = \frac{Q}{C}$$

The charge Q depends on the piezo-electric constant K, on the area lw of the electrodes, and on the displacement x. Consequently,

$$Q = Klwx$$

Now x depends on the force P and on the mechanical properties of the crystal.

Within the elastic limits of a material, force is proportional to displacement, i.e., stress $\propto$ strain. Consequently the ratio of stress to strain is a constant: it is called the modulus E of the material and is defined thus:

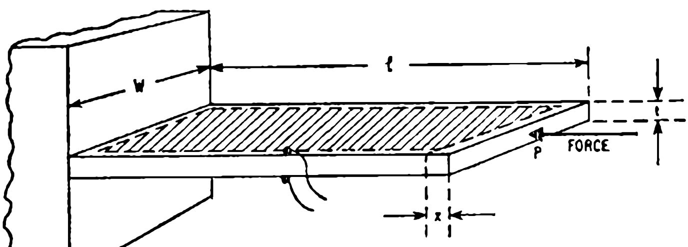


Fig. 7.22.—Expander bar under compressive stress showing method of mounting and main dimensions

$$E = \frac{\text{stress per unit area}}{\text{strain per unit length}}$$

If p is the stress per unit area, i.e.

$$p = \frac{P}{wt}$$

$$\text{then} \quad E = \frac{p}{x/l}$$

where $\frac{x}{l}$ is the strain per unit length:

$$\text{hence} \quad x = \frac{pl}{E}$$

Substituting in the equation for charge Q we have

$$Q = Klw\,\frac{pl}{E} = \frac{Kpl^2w}{E}$$

If the permittivity of the crystal is ε

$$C_c = 4\pi\varepsilon l\frac{w}{t}$$

hence
$$V = \frac{Kpl^2w/E}{4\pi\varepsilon l\dfrac{w}{t}} = \frac{Kplt}{4\pi\varepsilon E}$$

It will be seen that as far as sensitivity is concerned, the area of the electrodes wl appears to be unimportant and the sensitivity is dependent on the two factors $\frac{K}{\varepsilon E}$ and lt.

The first factor involves not only the piezo-electric constant but also the permittivity of the crystal and its modulus. It should be pointed out that in piezo-electric crystals the modulus is not independent of the direction of the stress as it is in many other materials. We see also that the sensitivity is improved if lt is large. There is however a limit to the value of l, especially if the force P is alternating at high frequency; l may then be comparable with the wavelength of the longitudinal vibration in the crystal. In this case the bar would behave as a transmission line open-circuited at the fixed end. Standing waves of strain would be produced and the charge under the electrodes would not be uniform and if l were large compared with λ, the polarity of the charge would change along the length of the bar.

If now we make t large, the capacitance C_c will be small, especially if we have decided to make the area under the electrodes small, since it does not contribute to the sensitivity. This would mean that the internal impedance $\frac{1}{\omega C_c}$ of the microphone would be high and if sensitivity were to be maintained, the length of the cable connecting the microphone to the amplifier would have to be restricted accordingly. Such difficulties do not occur in practice, for a typical value for C_c is of the order of 0·002μF and consequently the internal impedance of crystal microphones is much lower than that of electrostatic microphones which must be connected directly to a cathode follower or pre-amplifier.

In discussing the factors which affect sensitivity, it was shown that it is difficult to obtain high sensitivity and a low output impedance because the thickness t of the expander bar must be large to maintain sensitivity and small if the output impedance is to be low.

If instead of an expander bar a bender bimorph is used loaded as a cantilever (Fig. 7.20 (a)), the deflection x is much greater for the same applied force.

For a cantilever loaded at the free end and having the same dimensions as the bar, the deflection is:

$$x = \frac{Pl^3}{3EI}$$

where I is the moment of inertia.

For a bimorph of rectangular cross-section

$$I = \frac{wt^3}{12}$$

$$\text{hence} \quad x = \frac{4Pl^3}{Ewt^3}$$

In terms of the dimensions, the deflection x is

$$x \propto \frac{l^3}{wt^3}$$

From this proportionality it will be seen that a large deflection and hence high sensitivity can be obtained if l is large and t is small. These are just the conditions that must be observed if a low output impedance is desired. One conflicting factor is the width w, but it is offset by the fact that in the deflection equation l and t are third-power terms.

So far in the discussion we have been mainly concerned with the strain produced by a static load and have disregarded the mass and compliance reactance of the system. When the applied force is alternating as it is in a microphone, these reactances are all important.

The performance of the crystal microphone as a transducer can be more clearly appreciated from the equivalent circuit of Fig. 7.23.

In this circuit m is the effective mass, C_m is the compliance of the mechanical system and C_c is the electrical capacitance. The resistor r, shown dotted, is the bulk resistivity of the crystal and is important

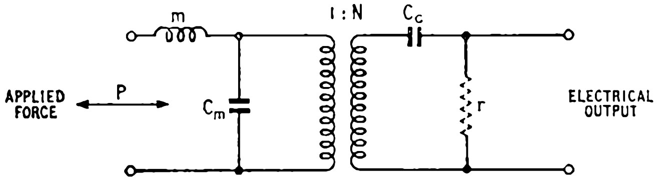


Fig. 7.23.—Equivalent circuit of a crystal microphone. Component values for Rochelle salt crystals are obtanable from Table 7.1

only at low frequencies. The turns ratio of the transformer takes into account the mechanical advantages of any lever system employed in the design.

In Table 7.1 will be found the value of the various components in the equivalent circuit for X-cut expander bars and square-torque bimorph Rochelle salt crystals.

Any imperfection in the mounting of the crystal will affect the compliance of the system. Imperfect mounting may permit movement in a direction where the crystal should be rigidly fixed and may restrain its motion in a direction where it should be free.

It has been shown that the crystal microphone is analogous to a capacitor which acquires a charge by virtue of the strains produced in it and that this charge is proportional to the displacement x. Hence the voltage $V \propto x$.

If the output voltage of the microphone is to be independent of frequency, it follows that x must also be independent of the frequency of the applied force P.

If P is of the form $p = P \sin \omega t$, the velocity u of the mechanical system is

$$u = \frac{P}{Z_m}$$

and the amplitude is
$$x = \frac{P}{Z_m \omega}$$

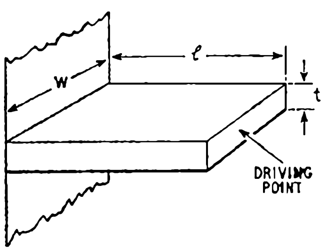
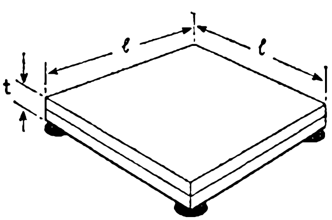
$$\text{then } x = \frac{P}{\left\{R_m + \mathrm{j}\left(\omega m - \frac{1}{\omega C_m}\right)\right\}\omega}$$

where m, R_m and C_m are the mass, resistance, and compliance of the mechanical system respectively.

In order to obtain a constant amplitude in the mechanical system, the resistance and mass reactance of the system must be negligible

Table 7.1.

ROCHELLE SALT CRYSTALS[a]

	X-cut expander bar	*X-cut square-torque bimorph*
		
m	$786\, lwt$	$246\, l2/t$
C_m	$31{\cdot}4 \times 10^{-12}\, l/wt$	$2{\cdot}40 \times 10^{-10}\, l^2/t^3$
C_0	$\varepsilon lw/t$	parallel connection $2{\cdot}8\varepsilon l2/t$
N	$0{\cdot}093/w$	$0{\cdot}128/t$
ε	50×10^{-10}	$1{\cdot}4 \times 10^{-10}$
F	Force in newtons (1 newton = 10^5 dyn)	
C_0	Crystal capacitance in farads	
m	Effective mass in kilograms	
C_m	Compliance in metres per newton	
lwt	Dimensions in metres	
ε	Value for room temperature 15°C	

[a] From information supplied by Brush Development Co.

in comparison with the compliance reactance. In which circumstances the displacement x is

$$x = \frac{P/\omega}{\omega C_m}$$

Thus ω disappears from the equation and the displacement $x = PC_m$ is independent of frequency.

The condition that the compliance reactance must be large in comparison to r and ωm is satisfied if the frequency of resonance of the mechanical system is high and the working range of the microphone is well below this frequency. In practical microphones resonance occurs at frequencies between 3,000 and 6,000 c/s. The response at low frequencies is virtually independent of frequency but rises as the fundamental frequency of the mechanical system is approached. The rise is usually smooth, especially for directly actuated sound-cell microphones, and satisfactory equalisation can easily be obtained by using a simple resistance-capacitance filter network. A typical constant-pressure frequency response characteristic is shown in Fig. 7.24.

If non-linear distortion is to be avoided, it is essential that the compliance of the moving system should be independent of the

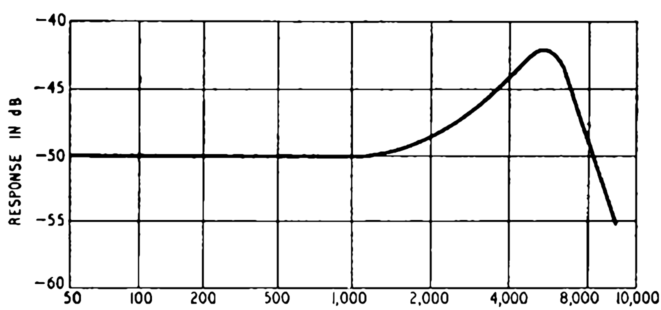


Fig. 7.24.—Constant pressure/frequency response of a crystal microphone. Reference level—1 V/dyn/cm²

amplitude of movement. Non-linear distortion can also result from certain hysteresis effects exhibited by Rochelle salt crystals. The effect which is at a maximum in the temperature range − 18° to + 24°C, is due to the lag between the voltage produced across a small load resistance and the applied force. Fortunately at ordinary sound levels the hysteresis effect is small, and if the sound level is high the hysteresis can be reduced to negligible proportions by using a high value of load resistor, or a cathode follower.

To avoid loss and keep distortion and extraneous noise to a minimum, not only should the microphone operate into a load resistor of suitable value, but the cable connecting microphone to amplifier should be short in length, well screened and of low capacity. A numerical example will illustrate how the capacity of the cable affects the output of the microphone.

Example

A microphone of impedance 80,000Ω at 1,000 c/s is connected to an amplifier by (1) 50 ft of coaxial cable having a capacity of 5pF/ft: (2) 50 ft of twin screened cable having a capacity of 40pF/ft. Find the voltage loss in dB in each case, assuming the impedance of the microphone is entirely capacity reactance.

The capacitance C_1 of the microphone is

$$C_1 = \frac{1}{\omega X_c}$$
$$= \frac{10^6}{2 \times 3{\cdot}14 \times 1{,}000 \times 80{,}000}$$
$$= 0{\cdot}002\mu\ F$$

The capacitance C_2 of the coaxial cable is $50 \times 5\text{pF} = 0{\cdot}00025\mu\text{F}$
The capacitance of the twin screened cable is $50 \times 40\text{pF} = 0{\cdot}002\mu\text{F}$

$$\text{dB loss} = 20 \log \left(1 + \frac{C_2}{C_1}\right)$$

With coaxial cable,

$$\text{dB loss} = 20 \log \left(1 + \frac{0{\cdot}00025}{0{\cdot}002}\right)$$
$$= 20 \log 1{\cdot}125$$
$$= 1 \text{ dB}$$

With twin screened cable,

$$\text{dB loss} = 20 \log \left(1 + \frac{0{\cdot}002}{0{\cdot}002}\right)$$
$$= 20 \log 2$$
$$= 6 \text{ dB}$$

7.4.1. OPERATING CONDITIONS: TEMPERATURE AND HUMIDITY

The crystal elements in piezo-electric microphones are specially coated to safeguard the microphone against abnormal conditions of temperature and humidity. High temperatures cause the crystal to lose water of crystallisation and become dehydrated while in very damp conditions the crystal may partially or entirely dissolve. These changes in the structure of the crystal are accompanied by changes in resistance and capacitance as set out in Table 7.2.

Table 7.2.

CHANGES IN THE STRUCTURE OF A CRYSTAL ELEMENT

Resistance	*Capacitance*	*Condition*
Increased	Decreased	Dehydrated
Decreased	Increased	Partially dissolved
Decreased	No change	Surface leakage

The changes in resistance and capacitance given in the table do not apply to ammonium *di*hydrogen phosphate (ADP) crystals, for they have no water of crystallisation and hence do not dehydrate like Rochelle salt. Furthermore, they may be operated at temperatures approaching 212°F without danger of deterioration and without marked changes in their properties. For these and other reasons they have replaced Rochelle salt in conditions of high humidity or for use under water.

8

Bi-directional Microphones

8.1. Introduction

The microphones used by the BBC until 1933 were all of the pressure-operated type, but because of their large size relative to the wavelengths, their directional characteristics were not independent of frequency. At low frequency they were omni-directional but as the frequency increased, the characteristic altered progressively and at high frequencies they behaved virtually as uni-directional microphones. The variations in the directional characteristic made the positioning of the microphone with respect to the sound source critical in a studio. If the microphone was so positioned as to make the ratio of direct to indirect sound an optimum for the high frequencies, then it was incorrectly placed for the low frequencies. The sound quality as heard via the microphone under these conditions was boomy and low-pitched because of the excessive bass reverberation picked up as a result of the low-frequency omni-directional characteristic.

A certain amount of indirect or reverberant energy is desirable in a studio broadcast, the actual amount depending on the type of programme. If the reverberation is excessive, however, the characteristic of the music can be seriously impaired and the diction of the instruments of the orchestra lost. This is one of the factors which determines the maximum working distance between performers and the microphone. In general, microphones with omni-directional characteristics are placed closer to the sound source than the bi-directional or uni-directional types.

The introduction in 1934 of the pressure-gradient ribbon microphone brought about a marked improvement in the quality of broadcast programmes. The ribbons and pole-pieces of these microphones were small in comparison to the wavelength and for the first time in the history of broadcasting, the directional characteristic in the horizontal plane was appreciably independent of frequency.

The difficulties experienced in positioning the microphone in the studio were greatly reduced.

The amount of reverberation picked up by these microphones with their bi-directional characteristics was only one-third of that picked up by the omni-directional types. In practice this meant that for the same ratio of direct to indirect sound the working distance between performer and microphone could be increased by $\sqrt{3}$, or almost doubled.

The frequency response of the microphones, especially in the high-frequency region, was more uniform than that of the best moving-coil microphone of the period and where the acoustics of the studio were suitable, distant balance techniques could be employed without loss of quality.

8.2. Pressure-gradient Ribbon Microphones

The pressure-gradient microphones in daily use in the BBC are of the electrodynamic type in which a ribbon acts both as the diaphragm and as the conductor. The electrodynamic voltage e generated in a conductor of length l moving with velocity u in a magnetic field of strength B is given by

$$e = Blu$$

The voltage e is proportional to the velocity and the response of the microphone is independent of frequency, if the velocity of the ribbon or conductor in the magnetic field is constant and independent of the frequency of the sound pressures.

Fig. 8.1 depicts an early form of a pressure-gradient microphone. An extremely thin ribbon of beaten aluminium foil supported at its ends is situated in a strong magnetic field provided by a permanent magnetic system. To facilitate adjustment, the ribbon is corrugated, concertina-fashion, at right angles to its length and suspended in the magnetic field between the pole-pieces. The clamping and tensioning arrangements at top and bottom of the ribbon can be seen in Fig. 8.1, together with the pole-pieces which distribute the flux uniformly in the plane of the ribbon.

Since the ribbon is corrugated uniformly throughout its length, it tends to vibrate in a plane at right angles to its surface in the manner of a stretched string. There is thus a possibility of harmonic modes of vibration at multiples of the fundamental ribbon resonance, but if the resonance is very low and suitably damped, an acceptable frequency response results. In some microphones the corrugations

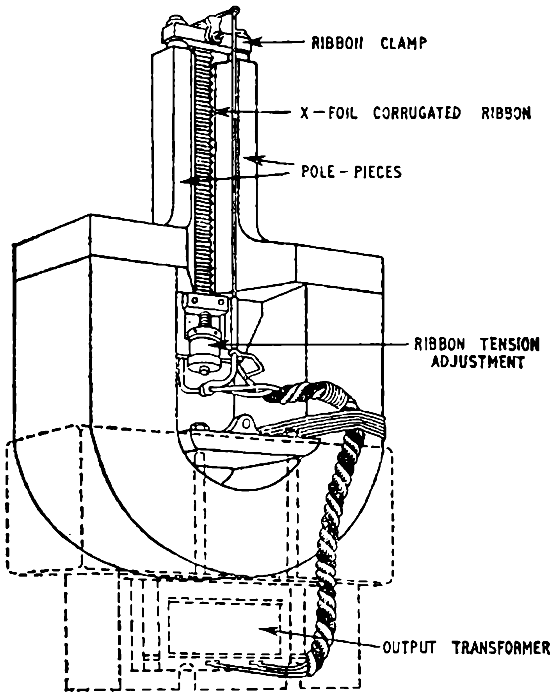


Fig. 8.1—Ribbon clamping and tensioning arrangements

in the ribbon are confined to the end portions only, the centre section being stiffened to ensure a piston-like motion, free from harmonic modes.

The response of a pressure-gradient microphone is dependent on the difference of the sound pressures acting on the front and rear surfaces of the diaphragm (Section 3.2); the resultant pressure p is given by

$$p = 2p_1 \pi \frac{f}{c} d$$

where p_1 is the acoustic pressure in the sound field, and d is the path length of the microphone, assumed to be small in comparison with the wavelength of the sound pressures.

The equation indicates that for constant sound pressures

$$p \propto f$$

The ribbon constitutes a mechanical system possessing mass m_d, compliance C_{md} and mechanical damping R_{md}. The velocity u of the system is given by

$$u = \frac{F_m}{Z_m}$$

$$= \frac{2Ap_1 \pi \frac{f}{c} d}{R_{md} + j\left(\omega m_d - \frac{1}{\omega C_{md}}\right)}$$

where A is the surface area of the ribbon on which the pressure acts.

The velocity of the ribbon is constant and independent of frequency, provided that the mechanical system is mass controlled, that is, the mass reactance ωm_d over the working range is large in comparison to R_{md} and $\frac{1}{\omega C_{md}}$. If this condition is observed, then the velocity is:

$$u = \frac{2Ap_1 \pi f d}{2\pi f m_d c}$$

$$= \frac{Ap_1 d}{m_d c}$$

Since A, p_1, d, c, and m_d are constants, the velocity of the system is constant and the induced voltage is independent of frequency.

Mass control of the mechanical system is achieved in practical designs by arranging for the fundamental resonance of the ribbon to be low and to be outside the audio-frequency range. The mass of a ribbon about 2½ in. long together with the air load on it is of the order of 0·0025 g. Because the tension in the ribbon is small, the compliance reactance is negligible above a few cycles per second, and as a consequence the fundamental resonance of the ribbon may be as low as 2 or 3 c/s. The resonance is damped by the acoustic resistance which results from the movement of the ribbon in the narrow slot formed by the pole-pieces (Fig. 8.2). This resistance, together with

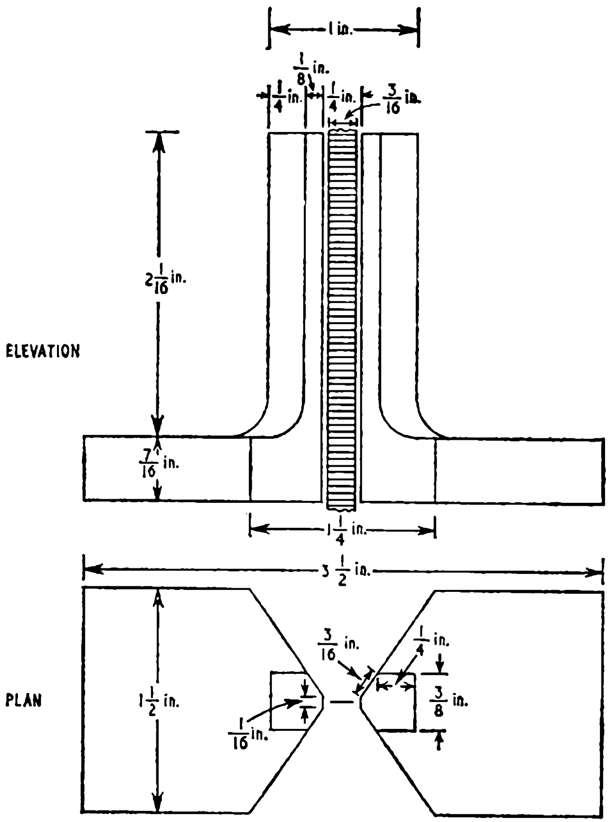


Fig. 8.2.—Ribbon microphone: pole pieces and ribbon

the electromagnetic damping, may form an appreciable part of the mechanical impedance below 50 c/s.

8.2.1. CONDITIONS AFFECTING THE MAGNITUDE OF THE FORCE F_m

The constant-velocity criterion holds so long as the applied force is proportional to frequency. The pressure-gradient principle of operation provides a force proportional to frequency, provided that the path length is small in comparison to the wavelength of the acoustic pressures. This is true at the low frequencies when d/λ is small, but as the quarter-wavelength condition is approached, the velocity of the system begins to decrease (Section 3.2). Above

$D/\lambda = \frac{1}{4}$ the velocity decreases rapidly with increase in frequency, and theoretically the output of the microphone falls and is zero at $d/\lambda = 1$.

Fig. 8.3 is the theoretical response of a pressure-gradient mass-controlled electrodynamic microphone and shows that at $d/\lambda = \frac{1}{4}$ the output is down by less than 1 dB. If a theoretical loss of about 1 dB at 10,000 c/s is acceptable, then it is possible to calculate d from the quarter-wave condition:

$$\begin{aligned} d &= \frac{c}{4f} \\ &= \frac{1{,}100 \times 12}{4 \times 10{,}000} \text{ in.} \\ &= 0{\cdot}33 \text{ in.} \end{aligned}$$

If the path length is 0·33 in., then the portion of the curve AB represents the theoretical response of the microphone: the quarter-wavelength condition occurring at B, i.e., at 10,000 c/s. As far as the frequency response is concerned, the design appears to be satisfactory, because the cut-off frequency at $d/\lambda = 1$ occurs at 40,000 c/s and is thus well outside the audio-frequency range. Unfortunately,

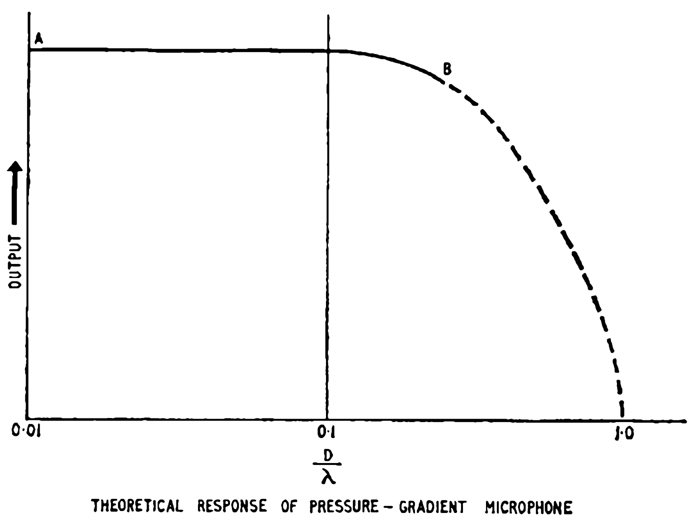


Fig. 8.3.—Theoretical responses of pressure-gradient microphone

the force F_m obtained by pressure-gradient operation is proportional to the path length and with a path length of only 0·33 in. the sensitivity of the microphone would be poor.

8.2.2. FACTORS AFFECTING SENSITIVITY

The sensitivity could be increased by 6 dB by doubling the effective length of the ribbon but the directional response of the microphone in the vertical plane would be adversely affected. Alternatively, an attempt might be made to increase the flux density in the gap but this would be difficult, for the magnetic system of a ribbon microphone is very inefficient in comparison to that of a moving-coil microphone. Because of the wide gap between the pole-pieces, only about 5 per cent of the flux leaving the magnet reaches the pole tips. If the driving force F_m could be increased, the velocity of the ribbon would be increased in proportion. This could be done by increasing the path length d. To increase the output by about 12 dB, the path length would have to be increased four times, bringing the cut-off frequency at $d/\lambda = 1$ into the working range of the microphone.

8.2.3. DRIVING FORCE TAKING INTO ACCOUNT THE DIFFRACTION EFFECTS

The graph in Fig. 8.3 shows that theoretically there is no output at the cut-off frequency; fortunately the graph is based only on the phase difference between the pressure p_1 at the front of the ribbon and the pressure p_2 at the back, and neglects any changes in the magnitude of the pressures produced at high frequency by diffraction effects.

At low frequency, the pressure p_1 is equal in magnitude to the pressure p_2 and the phase angle kd is small, as shown in Fig. 8.4 (a). At medium frequencies, there is little change in the magnitude of p_1 and p_2 but the angle of lag increases and the resultant pressure p increases in proportion (Fig. 8.4 (b)). At high frequency, the angle of lag is large and the resultant pressure p would be small if it were not for diffraction effects which decrease the pressure p_2 at the back and increase the frontal pressure p_1, as in Fig. 8.4 (c). Thus the driving force remains substantially proportional to frequency within the working range of the microphone.

The resultant pressure p is given for any condition by

$$p = \sqrt{(p_1{}^2 + p_2{}^2 + 2p_1p_2 \cos\theta)}$$

where $\theta = kd$.

To illustrate how the pressures p_1 and p_2 vary with frequency, we will assume that the diffraction effects around the rectangular frontal area of the microphone are similar to those produced by a square solid of width *w*. The curve p/p_0 plotted against w/λ in Fig. 8.5 shows how the pressures in front of and behind the square obstruction vary with frequency (that is with w/λ). The scale in kc/s has been compiled on the assumption that $w = 1$ in. The effect of the cavity produced by the chamfered pole-pieces, and the obstruction created

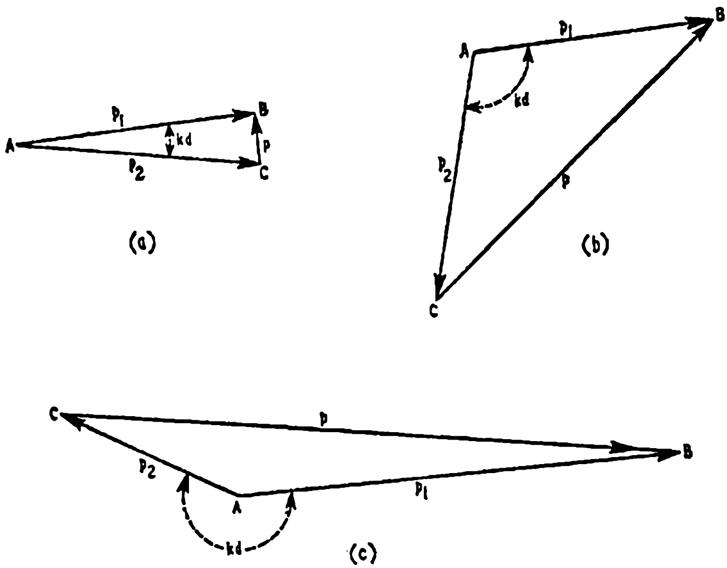


Fig. 8.4.—Ribbon microphone—driving force vectors. (a) Low frequency; (b) medium frequency; (c) high frequency. p_1 = *pressure on front surface;* p_2 = *pressure on rear surface*

by the permanent magnet itself, have been neglected in this illustration. It should be noted that the front and rear pressures are equal at the low frequencies only, that is, when *w* is very small compared to the wavelength. The pressure at the front is a maximum at $w/\lambda = 0{\cdot}8$ and decreases above $w/\lambda = 1$ to a minimum at $w/\lambda = 1{\cdot}7$.

Associated with the changes in pressure produced by the obstruction are changes in phase and these are shown in Fig. 8.5. At $w/\lambda = 1{\cdot}1$, the phase difference is 360° but the pressures front and rear are not equal and the combined effect of the pressure and phase characteristic is such that although there may be a dip in the response characteristic, there is no cut-off frequency in a practical microphone.

By careful attention to the acoustic design of the pole-pieces, a satisfactory output can be obtained even at $w/\lambda = 1{\cdot}1$ and the

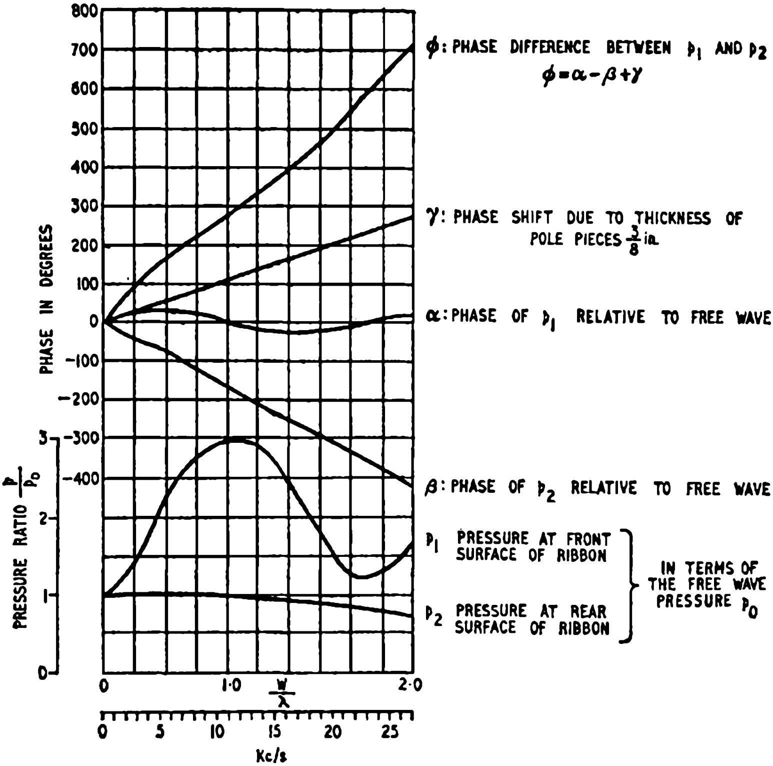


Fig. 8.5.—Magnitude and phase of pressures at front and rear of rectangular lamina

response of a well-designed pressure-gradient microphone in the high-frequency region is more uniform than that of most pressure-operated types.

8.2.4. LIGHTWEIGHT PRESSURE-GRADIENT MICROPHONES

The early pressure-gradient microphone, even judged by modern standards, had an acceptable frequency response but it was bulky, and because of the length of the ribbon the directional characteristic in the vertical plane was not independent of frequency. The improvements in magnetic materials which have taken place in the past 20 years have made possible pressure-gradient microphones of small size yet with adequate sensitivity. The large flux obtainable from modern magnetic alloys allows a shorter ribbon to be employed without adversely affecting output, and consequently the directional characteristic of the microphone in the vertical plane is much improved, especially at high frequencies.

8.2.5. THE MAGNETIC CIRCUIT

In certain programmes it is necessary, if good sound quality is to be achieved, to place the microphone close to the sound source and hence it may appear " in shot ". In these circumstances the microphone must be small and unobtrusive.

Small bi-directional microphones are available and owe their compact appearance not only to the newer high-efficiency magnetic materials, but also to the layout of their magnetic circuit. In general, two methods are employed in locating or positioning the magnet relative to the pole-pieces, and each design has its advantages and disadvantages.

In the early bi-directional microphones, the magnet was located below the pole-piece (Fig. 8.1). With this arrangement the sound

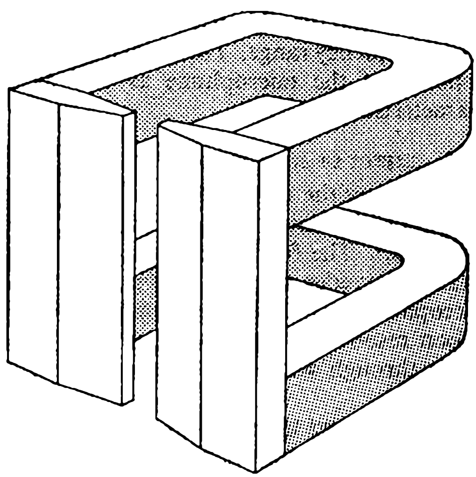


Fig. 8.6—Alternative magnet system for ribbon microphone

pressures have free access to the front and rear surfaces of the ribbon and so both halves of the figure-of-eight diagram in the horizontal plane are of equal area and similar shape. Many modern high-quality microphones use this layout because it produces a desirable directional characteristic in the horizontal plane but the positioning of the magnet below the pole-pieces affects the characteristic in other directions. The magnet acts as an obstruction to sounds arriving from below the horizontal and as a result the directional characteristic in the vertical plane is modified by the diffraction effects which occur at high frequencies.

Another practical method of arranging the magnetic circuit is to locate the permanent magnet or magnets behind the pole-pieces as shown in Fig. 8.6. This permits the frontal area of the microphone to be reduced in size and a neat and compact design results. Moreover, with this arrangement the efficiency of the magnetic circuit is high and a large electrical output relative to size can be obtained at small cost.

The magnetic circuit of Fig. 8.6 has some disadvantages when compared with that of Fig. 8.1. Sounds, especially those of high frequency, no longer have free access to the back of the ribbon because of the obstruction offered by the permanent magnet system. If the obstacle effect is pronounced, the directivity characteristic of the microphone will depart from the normal cosine curve, the departure being most marked at high frequency.

An added complication can arise if one or more of the magnets enclose behind the ribbon a volume of air capable of resonating in the audio-frequency range. This cavity resonance can effect the frequency response of the microphone as well as modifying its directional characteristic.

8.2.6. EFFECT OF SHORTENING THE RIBBON

Shortening the ribbon from about 2½ in. to 1 in. complicates the design of the microphone and presents the designer with fresh problems.[1] The fundamental frequency of the ribbon is raised and harmonic modes may be prominent in the operating range of the microphone. Mechanical damping is necessary to control the ribbon resonances and this may be obtained from the viscous flow of air through wire-mesh screens placed in close proximity to the ribbon.

Fig. 8.7 shows the ribbon vibrating in the second harmonic mode. The portion AB moving to the right compresses the air ahead of it and creates a rarefaction behind it, while the portion BC moving in the opposite direction has an area of high pressure in front and an area of low pressure behind. The baffle action of the pole-pieces prevents the rarefactions and compressions from neutralising each other on opposite sides of the ribbon but the pole-pieces do not obstruct the flow of air from the high-pressure area into the low-pressure area, provided that these areas are situated on the same side of the ribbon. The dotted arrows in Fig. 8.7 indicate the direction of air flow. As the ribbon vibrates, an alternating current of air is set up between adjacent sections moving in anti-phase. The air tends to flow in a direction along the length of the ribbon rather than normal to its surface, and to achieve effective damping a substantial

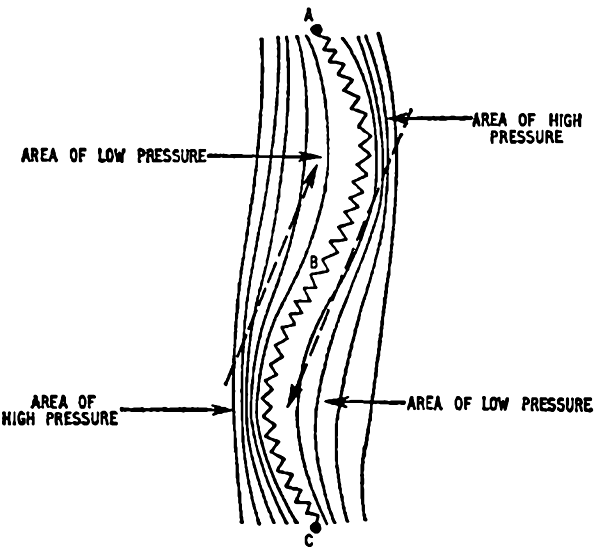


Fig. 8.7.—Alternating air flow between adjacent sections of ribbon vibrating in second harmonic mode

amount of this moving air must flow through the gauze screens. The need for the close spacing between screen and ribbon is thus apparent.

8.2.7. BASS COMPENSATION

Because of the damping required to control the higher vibrational modes, the ribbon system is no longer mass controlled at the low frequencies. A significant part of the impedance in this region is resistive and is due to the damping effect of the screens. Consequently the output of the microphone tends to fall at low frequency and some bass compensation is necessary if uniform sensitivity is to be achieved.

Braumühl and Weber in 1935 used a closely woven gauze screen in the form of a baffle to improve the low-frequency response of a bi-directional electrostatic microphone. A baffle of this type can be used with a ribbon microphone to offset the loss in sensitivity which occurs at low frequency as a result of damping the ribbon resonances. The baffle material, usually silk, bolting cloth or closely woven wire gauze, should have an acoustic impedance which is appreciably constant over the frequency range of the microphone. The baffle used in the BBC ribbon microphone consists of a single layer of bolting cloth supported on a coarse wire mesh. The baffle surrounds the pole-pieces and completely fills the circular opening, as can be seen in Plate 8.1.

Since the baffle is transparent to sound, the acoustic pressures pass through it to reach the back of the ribbon, following the normal path around the pole-pieces. Viscosity effects in the baffle material attenuate the acoustic pressures in their passage through it and so reduce the amplitude of the pressure acting on the rear surface of the ribbon.

If the material selected for the baffle has only a small reactive component, then the phase shift introduced by the baffle at low frequency is negligible and the phase angle between the pressures on the front and rear surfaces of the ribbon is determined solely by the path length and by the frequency.

The vector diagrams of Fig. 8.8 (a) and (b) show the resultant pressure on the ribbon at low and at high frequency and with and

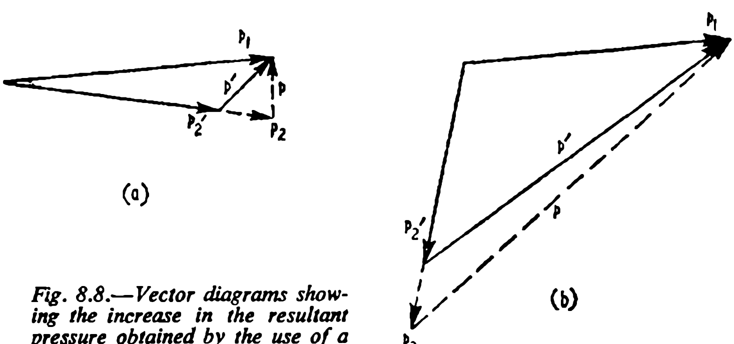


Fig. 8.8.—Vector diagrams showing the increase in the resultant pressure obtained by the use of a fabric baffle

without the baffle. The dotted vectors represent the conditions without the baffle and the full line vectors the conditions with the baffle.

The low-frequency conditions are represented in Fig. 8.8 (a) where in the absence of the baffle, p_1 and p_2 are the pressures at front and rear of the ribbon. The resultant pressure is p. The action of the baffle reduces the amplitude of the pressure p_2 at the back of the ribbon to p_2' without changing the phase angle. The resultant pressure p' is now greater than p and the sensitivity of the microphone at low frequency is improved.

Since the impedance of the baffle material is mainly resistive, high frequency and low frequency sounds experience the same attenuation in their passage through it; that is, the reduction in the amplitude of the pressures at high frequency is the same as the reduction in the pressure at low frequency. This condition is shown in Fig. 8.8 (b) where the amplitude of the pressure at the back of the ribbon is reduced from p_2 to p_2' by the action of the baffle. It will be seen that the resultant pressure on the ribbon is now reduced rather than increased but the reduction is not serious. Moreover the reactive component of the baffle impedance, which had negligible effect at low frequency, may at high frequency increase the phase angle between the front and rear pressures and so restore the high-frequency sensitivity. The frequency range over which the baffle is effective depends on whether the material is finely or coarsely woven and on how it is supported.

8.2.8. THE CASE OR HOUSING OF THE MICROPHONE

We have already seen that the case or housing of a pressure-operated moving-coil microphone can affect its frequency response and its directional characteristics. Although the case of a pressure-gradient microphone must be transparent to sound, it must also be sufficiently robust to protect the ribbon structure from rough handling and as a consequence it may present an appreciable obstacle to sound pressures. Furthermore, standing waves may be set up inside the case which will affect the response, increasing it at certain frequencies and decreasing it at others. In some microphones the dimensions of the case are chosen so that the standing-wave system can be used to smooth out irregularities in the response characteristic of the microphone at high frequencies.

If the ribbon is situated at a pressure node in the standing-wave system, the output is reinforced, for at this point in the wave train the pressure-gradient is a maximum, as shown in Fig. 8.9 (a). If the ribbon is placed centrally in the case, the first frequency of reinforcement occurs at the fundamental mode of vibration f_1 of the air in the case. If w is the width of the case from front to rear, $f_1 = c/2w$. No reinforcement will occur at the second harmonic mode $f_2 = c/w$, for the pressure gradient in the standing-wave system in the centre of the case is zero, as in Fig. 8.9 (b). Thus

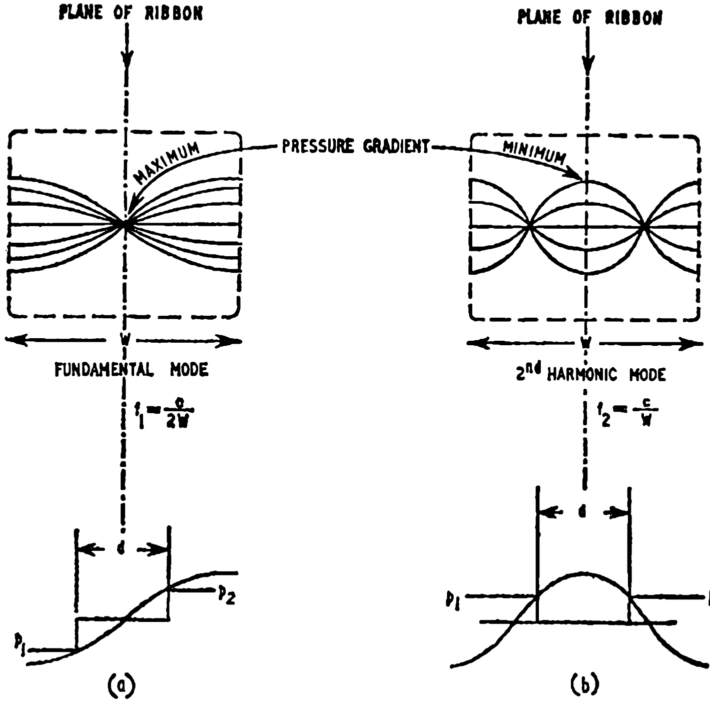


Fig. 8.9.—Pressure gradient at the nodal and anti-nodal points in a standing-wave system

reinforcement is limited to the fundamental and odd low-order harmonics, with a corresponding reduction in output at the even-order harmonics.

Reinforcement of the output may be necessary to smooth out an irregularity in the response characteristic at a high frequency, when the width of the case may be too great to achieve the desired result. Auxiliary reflecting surfaces in the form of perforated screens can be appropriately positioned in the case to produce the required reflected wave trains. Reflectors of this kind are seen in the BBC microphone depicted in Plate 8.1. Auxiliary reflecting screens of this type were used in an identical way in America by H. F. Olsen in 1951. Alternatively, the width of the case can be reduced in the immediate vicinity of the ribbon. Plate 8.2 shows a case which is heavily dished, so that the perforated metal sides are at the correct distance from each other for setting up the required standing-wave system without the need for separate auxiliary screens. This design, although unusual in appearance, is robust and inexpensive.

Ribbon microphones are susceptible to draughts and the case, while offering minimum obstruction to sound waves, must be sufficiently resistant to air flow to exclude draughts, together with dust and any small magnetic particles which would be drawn into the pole-piece gap and so impede the movement of the ribbon.

8.2.9. THE RIBBON ELEMENT

Aluminium foil is an ideal material for the ribbon, since it combines lightness with low resistivity. If maximum efficiency is the criterion, the mass of the ribbon should be equal to or comparable with the acoustic reactance of the air load on it, but if this leads to an irregular response because of high order vibrational modes, efficiency may have to be sacrificed. The smoothest response is obtained with a lightly tensioned ribbon having a thickness of about $2{\cdot}5 \times 10^{-5}$ in. The mass of this foil is slightly less than that required for maximum efficiency but the resultant loss in sensitivity is negligible in a well-designed microphone.

The width of the ribbon is important and is determined by a number of opposing factors. If the ribbon is very narrow, the gap between the pole-pieces can be reduced in width, resulting in an increased flux in the gap and an increase in the open-circuit voltage of the microphone. Unfortunately, reducing the width of the ribbon increases its resistance and in turn increases the internal impedance of the microphone considered as an alternator. When working into a load, the voltage drop which occurs as a result of the increased ribbon resistance can adversely affect the bass response of the microphone. Taking into consideration all the conflicting factors, a ribbon width of 0·25 to 0·23 in. is a good compromise.

8.2.10. GENERAL

The open-circuit voltage of a ribbon microphone is small, understandably less than that of a moving-coil microphone, but its internal impedance is very low so that a transformer with a turns ratio of about 1/22 can be used to match the microphone to a 300Ω load. With a transformer of this ratio the sensitivity compares quite favourably with that of other microphones. The open-circuit sensitivity referred to the secondary of the ribbon-to-line transformer is of the order of − 75 dB with reference to 1 V/dyn/cm^2. In connecting the ribbon to the primary of the transformer, precautions are necessary to minimise induction pick-up from stray alternating fields. If the first amplifier is at some distance from the microphone, a balanced line and carefully screened transformers will be required.

As has already been explained, the response of a pressure-gradient microphone to sounds of low frequency is much greater if the wave-front is spherical than if it is plane. Most of the sound sources in a studio are of sufficiently small acoustical dimensions to be virtually point sources of sound, producing waves which are spherical in the immediate vicinity of the source. It is therefore necessary, if excessive bass response is to be avoided, to place the microphone at some distance from artists or solo instruments. For normal programme purposes the distance between the artist and a pressure-gradient microphone should not be less than 2 ft.

The directional response characteristic in the horizontal plane of a well-designed pressure-gradient microphone is a figure-of-eight

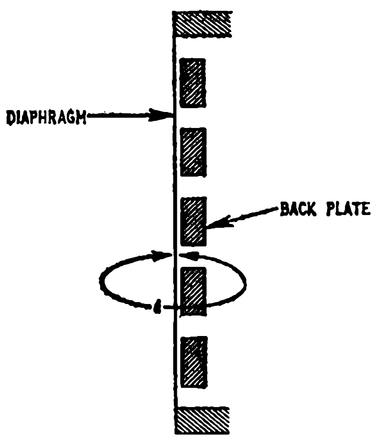


Fig. 8.10.—Diagrammatic cross-section of a pressure-gradient operated electrostatic microphone

loop, almost identical with that of Fig. 6.3. At the highest frequencies, however, some slight distortion in the shape of the loop may be apparent. In the vertical plane the characteristic is also a figure-of-eight, but since the length of the ribbon may be comparable with the wavelength of the higher audio frequencies, the directional response characteristic is not entirely independent of frequency.

8.3. Bi-directional Electrostatic Microphones

While pressure-gradient microphones of the electrodynamic type were in common use in Britain and America, Continental countries, Germany in particular, favoured pressure-gradient microphones based on the electrostatic principle. Pressure-operated electrostatic microphones had been a feature of the German broadcasting service for many years so the development was a logical one.

In the electrostatic type of microphone the back plate has holes drilled through it (Fig. 8.10) which permit acoustic pressures to reach the back of the diaphragm by way of the apertures in the back plate. The diaphragm is then pressure-gradient operated and moves as a result of the difference in phase of the pressures at front and rear. A uniform response with respect to frequency is obtained when the displacement of the diaphragm is independent of the frequency of the driving force (Section 7.2.1).

8.3.1. CONTROL CRITERION

Assuming the pressure p_2 at the back of the diaphragm is equal in amplitude to the pressure p_1 at the front but differs from it in phase due to the shift introduced by the path length d, then the force F_m is given by

$$F_m = 2Ap_1\pi\frac{f}{c}d$$

and the displacement x is given by

$$x = \frac{force}{\omega.Z_m}$$

$$= \frac{2Ap_1\,\pi fd}{\omega c\left\{R_{m_d} + \mathrm{j}\left(\omega m_d - \dfrac{1}{\omega C_{m_d}}\right)\right\}}$$

where m_d = mass of the diaphragm,
C_{m_d} = compliance of the diaphragm,
R_{m_d} = mechanical damping of the system.

The displacement amplitude x is independent of frequency if the mass and compliance reactance of the diaphragm are so small, compared to R_{m_d}, that they can be neglected in the calculation. Then

$$x = \frac{2Ap_1\,\pi df}{2\pi fcR_{m_d}}$$

$$= \frac{Ap_1d}{cR_{m_d}}$$

and the amplitude is independent of frequency.

8.3.2. THE RESISTANCE-CONTROLLED DIAPHRAGM SYSTEM

The condition for uniform sensitivity is satisfied if the diaphragm system is resistance-controlled. It is essential therefore that the diaphragm should be light, so that the mass, and hence the mass reactance are at a minimum; and, if possible, a large mechanical hysteresis loss should be associated with any flexing of the diaphragm, thus ensuring a large R_{md}. The diaphragm tension should be of the lightest, in order to reduce the stiffness of the system and keep the compliance reactance at a low value. This is in contradistinction to the highly tensioned diaphragms of the pressure-operated electrostatic microphones.

Nickel, plastics and amylacetate have been used successfully as diaphragm materials, but unless care is taken in the choice of the material and in the tensioning arrangements, troubles can arise in

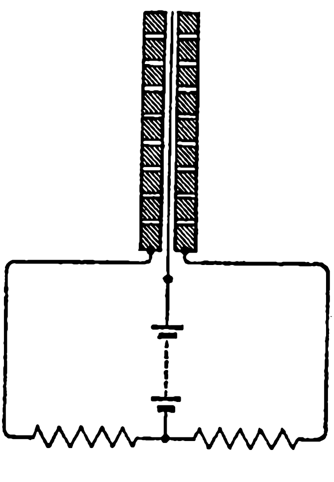


Fig. 8.11.—Diagrammatic representation of a symmetrical push-pull bi-directional electrostatic microphone

microphones that are subjected to the high temperatures produced by television lighting.

In the manufacture of amylacetate diaphragms, the acetate (Vinidier) is poured on to a glass plate and allowed to dry at room temperature. When hard, the film, which is about 0·001 in. thick, is stripped from the plate, held in a fixture under tension and sputtered with a very thin layer of gold which serves as the electrode.

The inherent mechanical resistance of the diaphragm is augmented by additional resistance obtained from the alternating movement of the air in narrow slots and tubes in the back plate. Although it is difficult to achieve over a wide frequency band, the diaphragm system can be substantially resistance-controlled throughout the major part of the operating range of the microphone. If for any reason the system ceases to be so controlled, then the sensitivity and response of the microphone are affected. Suppose that at the lower audio frequencies the compliance reactance of the diaphragm predominates: the displacement x is

$$x = \frac{2Ap_1\ \pi fd\ \omega C_{m_d}}{\omega c}$$
$$= \frac{2Ap_1\ \pi fd\ C_{m_d}}{c}$$

The displacement is thus proportional to frequency and as the frequency decreases, the output of the microphone falls.

If at the high audio frequencies the mass reactance of the diaphragm predominates, the displacement x is given by

$$x = \frac{2Ap_1\ \pi fd}{\omega^2\ cm_d}$$
$$= \frac{Ap_1 d}{2\pi fcm_d}$$

In this region the displacement is inversely proportional to f and as the frequency increases, the displacement decreases and the output of the microphone decreases in proportion.

The foregoing discussion is based on the assumption that the force is proportional to frequency. This assumption is not necessarily valid throughout the working range of a particular microphone.

8.3.3. DIRECTIONAL CHARACTERISTIC

Pressure-gradient electrostatic microphones employ circular diaphragms, and if the apertures which conduct the sound pressures from front to back of the diaphragm are symmetrically disposed, the directional characteristic in the horizontal plane is virtually identical with that in the vertical plane. At the higher audio frequencies, where the diameter of the diaphragm is comparable with the wavelength of the sounds, the figure-of-eight diagram will not be independent of frequency; furthermore, if phase shift is introduced by the apertures in the back plate, the two loops will be of unequal size. If the phase shift is appreciable, the directional characteristic tends towards that shown in Fig. 6.5 (a).

8.3.4. SYMMETRICAL BI-DIRECTIONAL ELECTROSTATIC MICROPHONE

The microphone in Fig. 8.10 is asymmetric in general construction and because of this, the directional characteristic may be affected. By adding another similarly shaped and similarly perforated electrode to the front or unoccupied face of the diaphragm, a more symmetrical arrangement is possible. The additional electrode will provide the required acoustical correction and need not be connected to the polarising supply, but if the push-pull arrangement of Fig. 8.11 is adopted, the sensitivity of the microphone is increased and the electrostatic attraction which formerly existed between the diaphragm and the single electrode or back plate can now be balanced out.

8.3.5. GENERAL

The sensitivity of pressure-gradient electrostatic microphones is relatively high in comparison with that of the pressure-gradient electrodynamic types and is of the order of $-$ 60 dB with reference to 1 V/dyn/cm^2. The frequency response characteristic sometimes exhibits irregularities which are due to acoustical and mechanical resonances. We have already seen that the diaphragm is prone to certain vibrational modes and these together with cavity resonances may affect the response.

As in all electrostatic microphones, the internal impedance is high and careful shielding is necessary if electrostatic pick-up of hum is to be reduced to a minimum. Dampness and high humidity can affect the performance of the microphone and the same precautions as those which apply to pressure-operated electrostatic microphones (Section 7.2) apply also to the pressure-gradient electrostatic types.

REFERENCE

1. Shorter, D. E. L. and Harwood, H. D., "The Design of a Ribbon-type Pressure-gradient Microphone for Broadcast Transmission", *BBC Engineering Monograph No. 4* (Dec. 1955).

9

Uni-directional Microphones

9.1. Introduction

IN THE THEATRE and in the concert hall it is often desirable to suppress or reduce unwanted noise coming from the audience or from sources unconnected with the programme. A uni-directional microphone, that is, one which is live on one face only, possesses properties which are particularly useful in these conditions, for not only can it discriminate against unwanted sources of noise, but because of its cardioid directional characteristic, it can accept sound on the live face over a large angle. For example, if suitably placed, the microphone can cover a wide stage in a studio or concert hall. Since it is insensitive to sound arriving at the back and comparatively insensitive to sound arriving from the sides, much of the direct audience noise and general reverberation can be suppressed. The ability to discriminate against sounds from the sides and rear makes a uni-directional microphone particularly suitable for distant sound pick-up in television studios or in sound reinforcing systems.

One of the first uni-directional microphones was produced in 1933 by J. Weinberger, H. F. Olson and P. Massa; they used a series arrangement of pressure and pressure-gradient elements in the form of a light ribbon suspended in a magnetic field, as in a conventional pressure-gradient ribbon microphone, except that the ribbon element was divided into two portions, one pressure-operated and the other pressure-gradient operated.

The construction of the pressure-gradient section and its action was similar to that of a normal bi-directional ribbon microphone, but the pressure-operated section of the ribbon possessed some novel features. To prevent sound pressures from reaching the back of the ribbon this section was enclosed, but to enclose it without special precautions would have been unsatisfactory, for the cavity so formed might have affected the response and directional characteristic of the microphone. Unlike the pressure-gradient section, the pressure section had to be resistance- rather than mass-controlled. It was therefore arranged to operate into a pipe or tube which effectively prevented the sound pressures reaching the back of the ribbon (Fig. 9.1). The tube was only a few feet in length but it was damped and appropriately terminated, so that it behaved as if it were infinitely

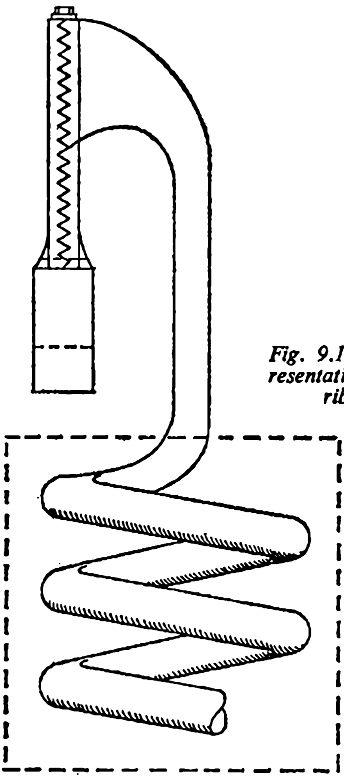


Fig. 9.1.—Diagrammatic representation of a uni-directional ribbon microphone

long. The impedance it presented was mainly resistive and of the value required to ensure that the pressure section of the ribbon was resistance-controlled.

The acoustic resistance or r_a of an infinitely long tube is $r_a = \rho c/A_t$ where ρ = density of air, c = velocity of sound in air, A_t = cross-sectional area of the pipe.

If the density ρ is in g/cm^3, the velocity c is in cm/sec and the cross-sectional area A_t is in cm^2, the acoustic resistance is $r_a \simeq 42/A_t$.

A long tube would have resulted in a cumbersome microphone, and for practical reasons the length was reduced to about 6 ft. The tube was in the form of a labyrinth, as indicated in Fig. 9.1.

If the losses in the tube had been small, reflection would have occurred at the end farthest from the ribbon, and a standing wave would have been set up in the tube. The designers' problem was to introduce sufficient damping to reduce reflections to a minimum and at the same time ensure that the impedance of the tube was mainly resistive and of a value equal to or approximating to $42/A_t$.

Tufts of felt appropriately packed and suitably spaced were inserted in the tube, producing an acoustic circuit whose constants could be adjusted by varying the quantity and position of the felt, so that over the audio-frequency range the tube presented the desired acoustic resistance. Since the reflection from the distant end was negligible, the tube, although of finite length, behaved as if it were infinitely long.

In spite of some diffraction introduced by the labyrinth, the directional characteristic of the microphone in the horizontal plane was satisfactory and the series combination of the pressure and pressure-gradient sections resulted in a microphone whose characteristic was substantially uni-directional over a wide frequency range.

9.2. Three-element Uni-directional Microphones

Another very successful microphone of the uni-directional type produced in the late 1930s employed three diaphragm-actuated

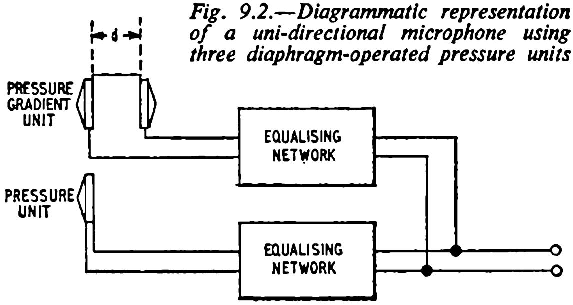


Fig. 9.2.—Diagrammatic representation of a uni-directional microphone using three diaphragm-operated pressure units

pressure units with Rochelle salt crystals as the transducing element. Two of the units were differentially connected, so that their output was proportional to the difference in pressure at two points in space separated by a small distance d (Fig. 9.2). This combination had a bi-directional characteristic and acted as the pressure-gradient element of the microphone. The third unit acted as the pressure-operated element and its output was combined with that of the pressure-gradient section. Simple resistor-capacitor networks equalised the response and provided the required phase shift, so that the characteristic, which was substantially uni-directional, was maintained over the operating range. Another novel feature of the microphone was its variable directional characteristic. A switch incorporated in the microphone allowed a choice of an omni-directional, bi-directional or uni-directional characteristic.

9.3. Two-element Uni-directional Microphones

Modern two-element uni-directional microphones are usually of the electrodynamic type and are developments of the original design by R. N. Marshall and W. R. Harry,[1] who used a bi-directional ribbon as the pressure-gradient element and a moving coil as the pressure element. Plate 9.1 shows the constructional details of a typical high-quality modern uni-directional microphone. The acoustic design of the two units closely follows standard practice, the moving-coil unit being resistance-controlled and the pressure-gradient unit mass-controlled.

It has been shown in Chapter 6 that a uni-directional characteristic can be obtained by combining omni-directional and bi-directional characteristics, and if certain conditions are satisfied, the general shape of the combined characteristics is a cardioid.

In a pressure-gradient microphone the generated e.m.f. e_{pg} varies with the angle of incidence and at any angle θ is $e_{pg} \cos \theta$. The e.m.f. e_p produced by a pressure-operated microphone is independent of the angle of incidence, and if the two microphones are so connected that the e.m.f. reinforce each other at $\theta = 0$, then $e_\theta = e_p + e_{pg} \cos \theta$.

If $e_p = e_{pg}$, then

$$e_\theta = e_p (1 + \cos \theta)$$

The equation indicates that the directional characteristic is a cardioid.

It is comparatively simple in a practical design to ensure that $e_p = e_{pg}$ at a single frequency, but it is difficult to match the outputs of the two units in both magnitude and phase over the entire audio-frequency range.

To satisfy the phase requirements both units should occupy the same point in space, but because of their finite size, they must in a practical microphone be separated by a finite distance. If the distance separating the units is small compared with the wavelength, they are effectively at the same point in space. In order to reduce the spacing between them to a minimum, both the pressure and pressure-gradient units should be small and compact. To this end, the permanent magnets associated with the bi-directional unit are usually placed behind the ribbon (Plate 9.1). Their positions at the extremities of the pole-pieces ensure that the acoustic pressures have free access to the back as well as to the front of the ribbon. The pressure unit is also small and is inclined at an angle of about 30° to the vertical, enabling the spacing between the units to be reduced still further.

9.3.1. THE MAGNITUDE AND PHASE OF THE INDUCED VOLTAGES

It is shown in Chapter 8 that the front and rear surfaces of the ribbon in a pressure-gradient unit are separated acoustically, not by the mere physical thickness of the ribbon, but by a distance *d*, called the path length, which is of the order of an inch or so in a practical microphone.

In this analysis it is assumed that the pressure unit is situated acoustically at a point midway between the front and rear surfaces of the ribbon, that is, at $d/2$ from each. The magnitude and phase of the pressures acting on the ribbon and on the diaphragm of the moving

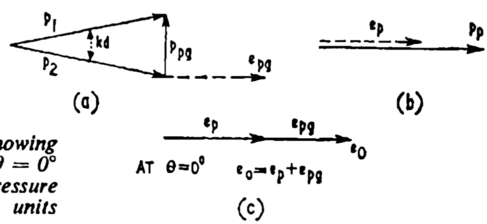


Fig. 9.3.—Vector diagrams showing the phase relationship at $\theta = 0°$ of the e.m.f. from the pressure gradient and pressure units

coil are shown in Fig. 9.3, together with the voltages induced in the ribbon and in the coil by virtue of their movement in their respective magnetic fields.

In Fig. 9.3 (a) p_1 and p_2 are the pressures on the front and rear surfaces of the ribbon, the resultant pressure being p_{pg}. This pressure causes the ribbon to move, and because the mechanical system is mass-controlled and analogous to an inductive reactance, the velocity lags 90° on the pressure. Since the induced voltage e_{pg} is in phase with the velocity, it too must lag 90° on p_{pg}.

In Fig. 9.3 (b) the pressure on the diaphragm of the moving-coil unit is represented by p_p, and because this unit is situated acoustically midway between the front and rear surfaces of the ribbon, the pressure p_p lags on p_1 by $kd/2$ radians and leads p_2 by the same amount. The diaphragm system is resistance-controlled and the voltage induced in the coil e_p by virtue of its movement will be in phase with p_p. Fig. 9.4 (a) and (b) shows that e_p and e_{pg} are in phase and for $\theta = 0°$ the combined voltage $e = e_p + e_{pg}$. If a cardioid characteristic is desired, e_p and e_{pg} must be equal in magnitude and similar in phase at $\theta = 0°$.

The conditions which obtain when sound impinges on the back of the microphone $\theta = 180°$ are shown in Fig. 9.4. The 180° change

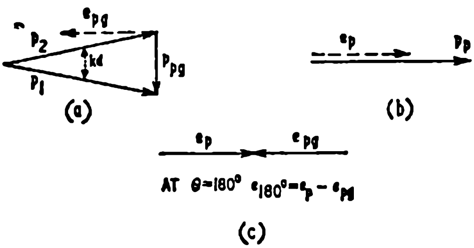


Fig. 9.4.—Vector diagrams showing the phase relationship at $\theta = 180°$ of the e.m.f. from the pressure gradient and pressure units

in direction of the sound wave causes the acoustic pressures to act on the rear surface of the ribbon before affecting the front surface, and consequently p_2 now leads p_1 by kd radians. As a result p_{pg} is reversed in phase and the induced voltage which lags on p_{pg} by 90° is now 180° out of phase with e_p. Thus, for $\theta = 180°$,

$$e = e_p - e_{pg}$$

If e_p and e_{pg} are of equal amplitude, there will be no output from the microphone at this angle. This is the desired condition for a cardioid characteristic.

9.3.2. EFFECT OF MISALIGNMENT BETWEEN THE UNITS

So far in the discussion it has been assumed that the magnitude and phase of the pressures acting on the diaphragm of the moving-coil unit are independent of the angle of incidence of the sound. This is an invalid assumption in a practical microphone, especially at high frequency, when the housing or permanent-magnet structure of the moving-coil unit acts as an obstacle to sounds arriving at the back, effectively increasing the acoustic path which the wave must traverse to reach the diaphragm. The effect is analogous to a change in the position of the pressure unit relative to the pressure-gradient unit.

In order to show how a change in the position of the pressure unit affects the output of the microphone, vector diagrams have been drawn for the same angles of incidence as before (Fig. 9.5) for $\theta = 0°$ and (Fig. 9.6) for $\theta = 180°$. But it is now assumed that the pressure unit is effectively displaced from the acoustic centre of the pressure gradient by a distance d' so that a sound wave arriving along the axis at $\theta = 0°$ acts on the diaphragm of the pressure unit before encountering the front surface of the ribbon. Under these circumstances the e.m.f. produced by the two units will no longer be in phase and e_p will lead e_{pg} by kd' radians where d' is the acoustic

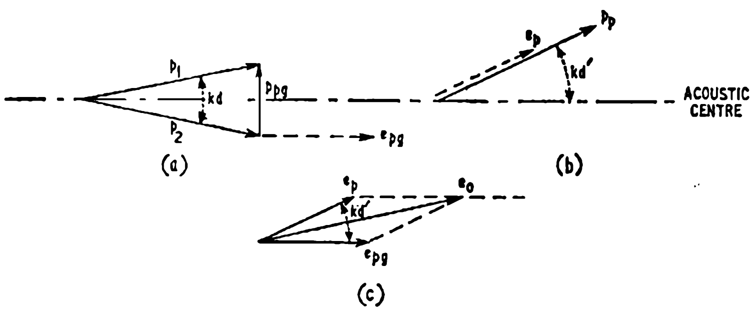


Fig. 9.5.—Vector diagrams showing the effect of misalignment on the magnitude and phase of the output voltage at $\theta = 0°$

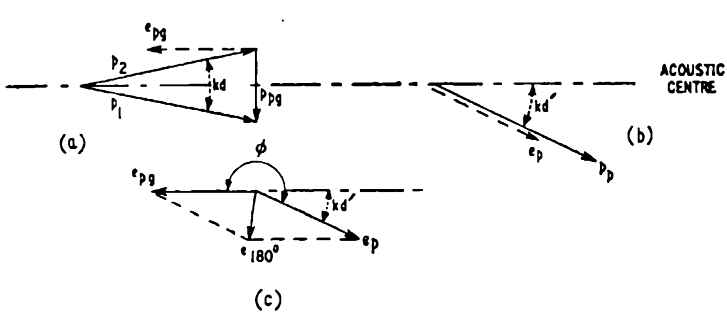


Fig. 9.6.—Vector diagrams showing the effect of misalignment on the magnitude and phase of the output voltages at $\theta = 180°$

distance separating the units on the $\theta = 0°$ axis. Although the voltages e_p and e_{pg} may be equal in magnitude, their combined voltage e is now dependent on the phase angle between them. From Fig. 9.5 (c):

$$e_o = (e_p^2 + e_{pg}^2 + 2e_pe_{pg} \cos kd')^{\frac{1}{2}}$$

In a practical microphone d' would be small compared with the wavelengths of sounds at low frequency, and the phase angle between the e.m.f. would be so small that the output of the microphone would be unaffected. As the frequency increased the phase angle would increase, since $kd' = 2\pi fd'/c$, and the output of the microphone would fall, affecting both the response and the directional characteristic.

Sounds arriving at the back of the microphone at $\theta = 180°$ act successively on the rear surface of the ribbon, the front surface of the ribbon and the diaphragm of the pressure element, in that order. Hence p_1 lags on p_2 and p_p lags on p_1, as shown in Fig. 9.6 (a) and (b).

The voltage e_p induced in the moving coil is no longer 180° out of phase with e_{pg} but lags on it by an angle ϕ greater than π; i.e., $\phi = \pi + kd'$ radians, as shown in Fig. 9.6 (c).

Although the two elements may contribute voltages of equal magnitude, their combination at $\theta = 180°$ would not result in complete cancellation, for the output of the combination at $\theta = 180°$ is

$$e_{180°} = (e_p^2 + e_{pg}^2 - 2e_pe_{pg} \cos kd')^{\frac{1}{2}}$$

The magnitude of the resultant e.m.f. $e_{180°}$ is dependent on the increase in the phase angle beyond 180°. At low frequencies the discrepancy might be small, but even a small change will affect the directional characteristic adversely. A change of only 10° (i.e. from 180° to 190°) might reduce the front-to-back discrimination of the microphone from a theoretical infinity to about 20 dB.

In the discussion we have assumed a misalignment d' between the units and also assumed that the misalignment is independent of the angle of incidence. This would certainly not be true of a practical design, for the acoustic alignment of the units would be dependent on both frequency and angle of incidence.

The pressure and pressure-gradient unit may be in perfect alignment on the $\theta = 0°$ axis, but on the $\theta = 180°$ axis the increased path which the wavefront must follow around the housing of the pressure unit before reaching the diaphragm would seriously upset the alignment (Fig. 9.7).

To reduce diffraction and misalignment effects to a minimum, the dimension of the transducing elements must be made small compared with the wavelength, a condition which is generally impossible to satisfy at high frequency.

Furthermore, the directional characteristic of both elements departs from the normal at high frequency, the effect being most

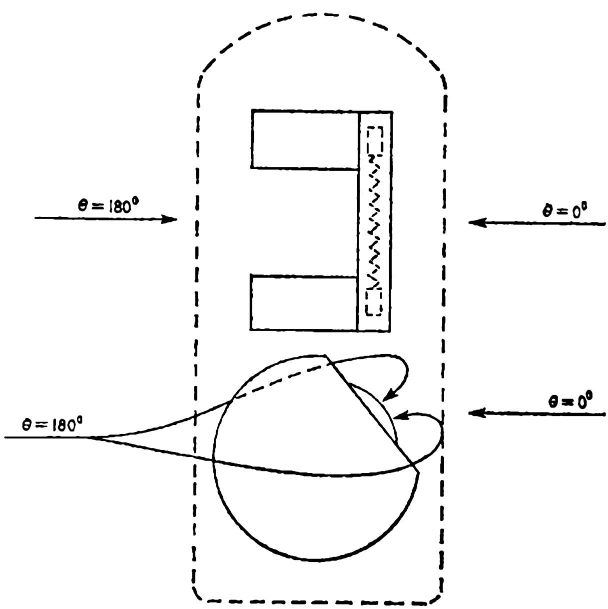


Fig. 9.7.—This shows the increase in the length of the path traversed by the wave around the pressure unit at $\theta = 180°$

marked in the pressure unit, which becomes very directional as shown in Fig. 5.3.

In order to preserve the uni-directional characteristic of the combination and achieve a sharp minimum on the 180° axis, R. N. Marshall and W. R. Harry progressively attenuated the output of the ribbon section above 2,000 cycles, and at frequencies beyond 6,000 cycles relied entirely on the directional characteristic of the moving-coil section. This artifice is still employed in modern two-element microphones where the directivity of the pressure unit is accentuated by a close-fitting baffle (Plate 9.1) and by inclining the unit at about 30° to the vertical.

The effect of tilting the pressure unit is most marked for a sound arriving at the back of the microphone on the 180° axis when its angle of incidence relative to the diaphragm of the pressure unit is 150° rather than 180°. A comparison of the curves (Fig. 5.1) for $\theta = 0°$ and $\theta = 180°$ with those for $\theta = 30°$ and $\theta = 150°$ gives some indication of the advantages gained from the tilting of the pressure unit.

9.3.3. ELECTRICAL CORRECTION

From what has been said it is obvious that it is difficult to obtain voltages of the required amplitude and phase from the two transducing elements over the entire audio-frequency range. To ensure that the directional characteristic will not vary too markedly with frequency, some form of electrical correction is commonly employed. It is unlikely that the two elements will have equal sensitivity, and in order to match the voltages a transformer is inserted in the output of one of the elements (usually the pressure-gradient unit). Matching the voltages by adjusting the turns ratio of the transformer may result in a balance at one frequency only. A perfect match over the entire frequency range is only possible if the response of the two units is identical.

It may also be necessary to provide some form of phase correction, especially if a short stiff ribbon has been employed in the pressure-gradient unit. A ribbon of this type is less sensitive to mechanical shock and to wind noise and possesses advantages if the microphone is to be used on a sound boom in a television or film studio; but the increased stiffness, which is desirable for stability, raises the natural frequency of the ribbon, bringing the resonance into the lower end of the audio-frequency range. It will be appreciated that when operating at or near resonance the ribbon is resistance- rather than mass-controlled, and the assumption that the velocity, and hence the induced e.m.f., lags 90° on the driving force is no longer valid.

Furthermore, the large output at the fundamental resonance of the ribbon introduces a pronounced peak in the frequency response of the pressure-gradient unit. A series combination of resistance and inductance connected across the output of the pressure-gradient element will reduce the peak in the response and have the added advantage of bringing the voltage e_{pg} more nearly into phase with voltage of the pressure unit e_p.

Additional phase correction may have to be provided, to compensate for phase differences between the output voltages of the elements resulting from acoustic misalignment. Although the phase difference will vary with the angle of incidence, it is likely to be a maximum at $\theta = 180°$, and it is here that a small difference in phase may have the most serious effect on the microphone's discrimination. The phase-correcting network is usually arranged to compensate for phase differences associated with the 180° axis and so to retain the maximum front-to-back discrimination. A simple L-section low-pass filter with resistance in the shunt arm is usually employed. There are obvious advantages if the network can be housed within the case of the microphone, but if it is connected directly in the low-impedance circuit associated with the pressure-gradient unit, some of the components might be difficult to house because of their size. It is usual to provide the output transformer with a high-impedance

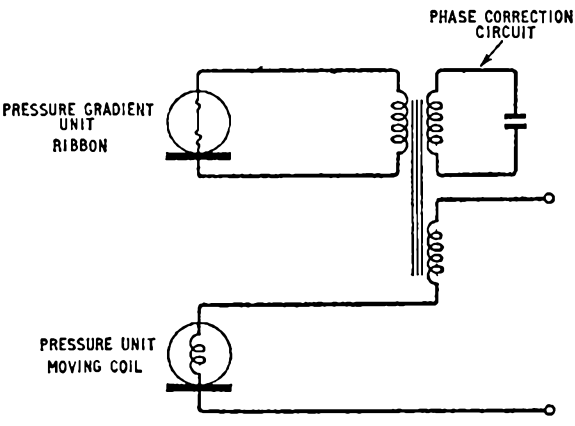


Fig. 9.8.—Combining transformer showing tertiary winding and phase-correcting capacitor

tertiary winding and the phase correction is obtained by connecting a small capacitor across this winding (Fig. 9.8). At high frequency the equivalent circuit of the tertiary winding, and its capacitor load when referred to the secondary circuit of the transformer, is shown in Fig. 9.9. The series inductances L_1 and L_2 are obtained by adjusting the leakage reactance of the windings to the required value, the resistance R in series with the capacitor C being incorporated in the winding itself. Although the actual value of the capacitor in Fig.

9.9 may be only of the order of 0·1–0·2 μF, its effective value when referred to the low-impedance secondary winding may be equivalent to several microfarads.

Apart from the phase correction for which the network is intended, it introduces a gradually increasing loss in the output of the pressure-gradient unit as the frequency is raised. The increasing attenuation

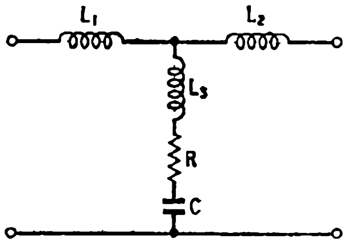


Fig. 9.9.—Equivalent circuit of phase-correcting network

characteristic is used to reduce the contribution from the pressure-gradient unit so that the directional properties of the pressure unit can be exploited at high frequency.

9.3.4. SPACING OF THE ELEMENTS IN THE VERTICAL PLANE

We have discussed the effect of spacing and misalignment of the elements in the horizontal plane but so far we have not examined the effect of the spacing between the units in the vertical direction. In the example shown in Plate 9.1 the separation between the pressure and pressure-gradient units in the microphone is of the order of an inch or so, and it might be thought that the directional characteristic in the vertical plane would be seriously affected when the spacing was comparable with the wavelength. This would certainly be so if at high frequencies the directional characteristic were obtained by combining the outputs of the two elements; but the gradual reduction of the output of the pressure-gradient element and the progressive reliance on the pressure element alone to provide directivity minimises the effect of the vertical spacing.

A three-position switch is usually a feature of two-element cardioid microphones enabling the units to be selected singly or in combination, to give a choice of an omni-directional, a bi-directional or a uni-directional characteristic. It should be remembered when positioning these microphones in studios that the response of the pressure-gradient unit will be affected by the shape of the wavefront.

Although the pressure-gradient element contributes only a part of the output of the microphone, the frequency response and directivity pattern of the combination will alter in a spherical-wave field.

9.4. Single-element Uni-directional Microphones

In the section on two-element uni-directional microphones we outlined some of the difficulties encountered in matching the response of the pressure and pressure-gradient elements and we showed how the finite separation between the units caused the sound, at certain angles, to reach one element before the other and so produce unwanted phase differences. Many of the difficulties associated with two-element uni-directional microphones can be overcome if a single element is used and so arranged that it is subject to a force derived

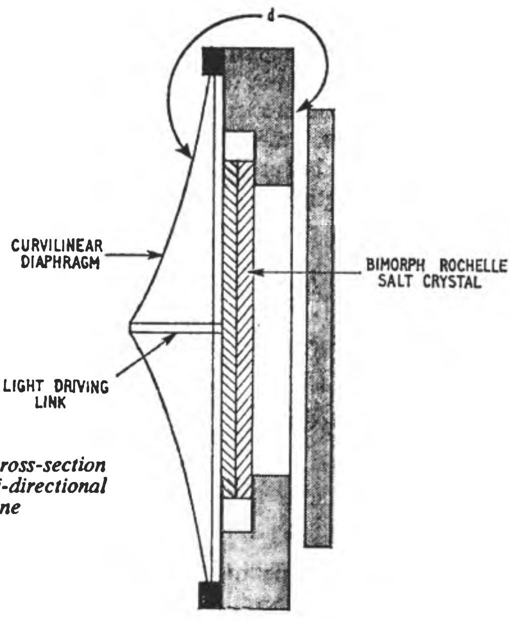


Fig. 9.10.—Schematic cross-section of diaphragm-operated uni-directional crystal microphone

in part from the pressure gradient and in part from the pressure in the acoustic wave.

In discussing the phase-shift principle (Chapter 3) we showed that the uni-directional property of these microphones was dependent on a phase-shifting network incorporated within the microphone itself. For this reason microphones of this type are often referred to in early literature as " uniphase " microphones, the term being derived from the words " uni-directional " and " phase ", denoting the system from which the microphone obtains its single-sided property.

9.4.1. UNI-DIRECTIONAL CRYSTAL MICROPHONE

One of the early single-element uni-directional microphones with a conventional diaphragm system employed a Rochelle salt crystal as the transducing element.[2] Fig. 9.10 is a schematic drawing of the microphone, showing the crystal in the form of a bimorph connected by a light mechanical link to a cone-shaped diaphragm. As has already been pointed out, the mechanical impedance of a crystal transducer is large and the curvilinear shape of the diaphragm was chosen to increase its rigidity at the driving point, thus enabling it to transmit pressures to the mechanically stiff crystal.

The circular case containing the crystal has a cover plate at the back spaced from the main body of the microphone so as to form an annular outlet to the air. Because of this aperture, the diaphragm is subject to acoustic pressures on both front and rear surfaces. The pressure at the back may be comparable in amplitude with the pressure at the front but is delayed in phase. The delay is dependent, first, on the length of the external path d_e from the front of the

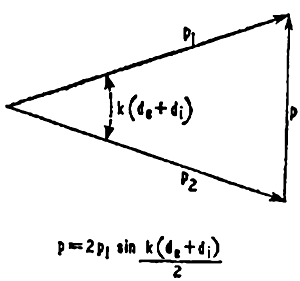


Fig. 9.11.—Vector diagram showing the magnitude and phase of the resultant pressure p

diaphragm to the opening of the annular aperture and secondly, on the phase shift introduced when the wave passes through the annular slit, reaching the back of the diaphragm by way of the circular enclosure containing the crystal.

9.4.2. FREQUENCY RESPONSE

If we assume that the wave is unattenuated in its passage through the body of the microphone and that the internal path traversed is equivalent to an acoustical path of length d_i, then the pressures on front and rear surfaces of the diaphragm are as represented in Fig. 9.11. p_1 is the pressure on the front surface of the diaphragm and p_2 the pressure on the back: p_2 lags on p_1 by $k(d_e + d_i)$ radians and the resultant pressure is

$$p = 2p_1 \sin \frac{k\,(d_e + d_i)}{2}$$

If the combined path length $(d_e + d_i)$ is small compared with the wavelength,

$$p \simeq p_1 k\,(d_e + d_i)$$

$$= 2p_1 \frac{\pi f}{c}(d_e + d_i)$$

If A is the effective area of the diaphragm and Z_m its mechanical impedance, the velocity is

$$u = \frac{2Ap_1\pi f\,(d_e + d_i)}{cZ_m}$$

The effective displacement is

$$x = \frac{2Ap_1\pi f\,(d_e + d_i)}{\omega c Z_m}$$

$$= \frac{Ap_1\,(d_e + d_i)}{cZ_m}$$

The microphone described by Bauer was stiffness-controlled and the mechanical impedance Z_m was of the form $1/\omega C_m$. Hence the effective displacement was

$$x = \frac{Ap_1\,(d_e + d_i)\,\omega\,C_m}{c}$$

From the above, the displacement is

$$x \propto f$$

The piezo-electric voltage produced by a crystal is proportional to the effective amplitude of deformation x: hence the output of the microphone, when it is stiffness-controlled, is proportional to frequency, i.e., if the frequency is doubled, the output voltage is doubled and the response rises at the rate of 6 dB per octave. This is illustrated in Fig. 9.12 (a). The response continues to rise for as long as (d_e + d_i) remains small compared with the wavelength, but when the wavelength is comparable with or larger than the dimensions of the instrument, the microphone becomes pressure-operated and its output is then independent of frequency.

To obtain a uniform response Bauer employed electrical corrections: the network took the form of a parallel combination of

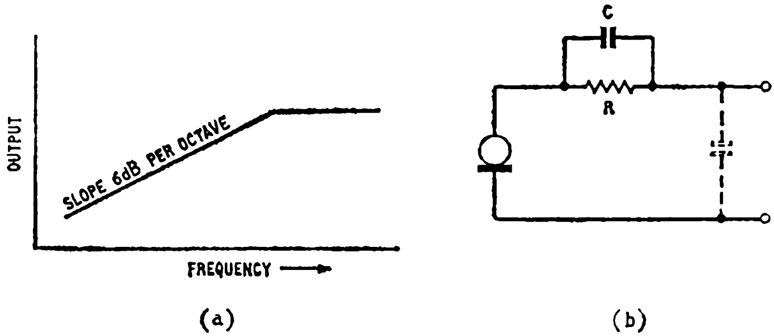


Fig. 9.12.—(a) Idealised response of a uni-directional stiffness-controlled crystal microphone; (b) parallel combination of R and C in series with the self-capacitance of the cable to correct the rising frequency response

capacitor and resistor in series with the self-capacitance of the cable between the microphone and the first stage of the amplifier (Fig. 9.13 (b)).

9.4.3. DIRECTIONAL CHARACTERISTIC

On the $\theta = 0°$ axis the effective path length ($d_e + d_i$) is a maximum and the resultant pressure on the diaphragm is

$$p = \frac{2p_1 \pi f\,(d_e + d_i)}{c}$$

This equation shows that the resultant pressure is proportional to the path length, as well as to frequency. Any variation in the effective length of the path with the angle of incidence will affect the output of the microphone. The external path d_e is not independent of the direction from which the sound arrives, but varies with the angle of incidence in a manner similar to the path of a pressure-gradient microphone.

The effective length of d_e at angle θ is

$$d_{e\theta} = d_e \cos \theta$$

so that the effective length of the combined path d at angle θ is

$$d_\theta = (d_e \cos \theta + d_i)$$

and the resultant pressure p at angle θ is

$$p_\theta = \frac{2p_1 \pi f (d_e \cos \theta + d_i)}{c}$$

If the phase shift introduced by the internal path d_i is equal to phase shift introduced by the external path d_e, then

$$p_\theta = \frac{2p_1 \pi f d_e (1 + \cos \theta)}{c}$$

At a single frequency this is of the form

$$p_\theta = a\,(1 + \cos \theta)$$

where a is a constant.

Since the output of the microphone is proportional to the pressures on the diaphragm, plotting a polar graph of the pressures for various values of θ will give an indication of the shape of the directional characteristic of the microphone.

This has been done for certain values of θ and the result shown in Fig. 9.13. The polar plot indicates that the directional characteristic is a cardioid but it is not obvious from the equation for p_θ how this characteristic is achieved. It might be advantageous to examine the variations in the pressure p with angle of incidence, using vector diagrams.

Fig. 9.14 (a), (b) and (c) shows the variations in the amplitude of p for three values of the angle of incidence, i.e. $\theta = 0°$, $\theta = 90°$ and $\theta = 180°$. In Fig. 9.14 (a) at $\theta = 0°$ the pressure at the front of the diaphragm is represented in magnitude and phase by p_1 and the pressure at the rear aperture by p_b. The phase angle between the

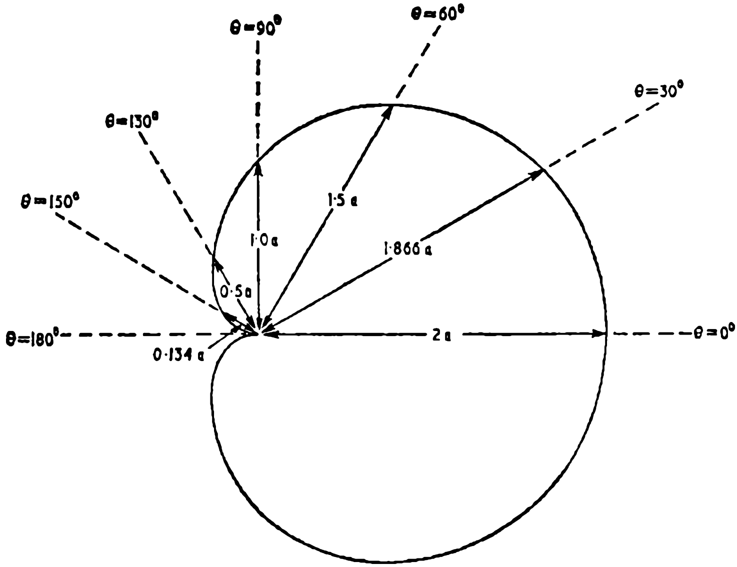


Fig. 9.13.—Polar plot of a $(1 + \cos\theta)$

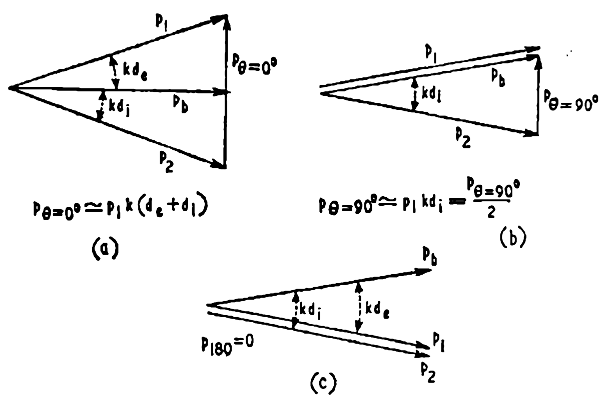


Fig. 9.14.—Vector diagrams showing the magnitude of the resultant pressure p *for* $\theta = 0°$, $\theta = 90°$ *and* $\theta = 180°$ *when* $d_e = d_i$

pressures is kd_e. The pressure at the back of the diaphragm is represented by p_2 which lags on p_b by kd_i radians.

If $p_1 = p_2$ the resultant pressure

$$p = 2p_1 \sin \frac{k(d_e + d_i)}{2}$$
$$\simeq p_1 k (d_e + d_i)$$

At $\theta = 90°$ the acoustic pressure arrives at the same instant in time at the front of the diaphragm and at the rear aperture. Thus p_1 and p_b are coincident, as shown in Fig. 9.14 (b). To reach the back of the diaphragm, the acoustic pressure must then traverse the internal path d_i, hence p_2 lags on p_1 and p_b by kd_i radians.

The resultant pressure

$$p = 2p_1 \sin k \frac{d_i}{2}$$
$$\simeq p_1 k d_i$$

If d_e and d_i are equal, the pressure on the diaphragm at $\theta = 90°$ is half the pressure at $\theta = 0°$. This is a requirement for the cardioid directional characteristic.

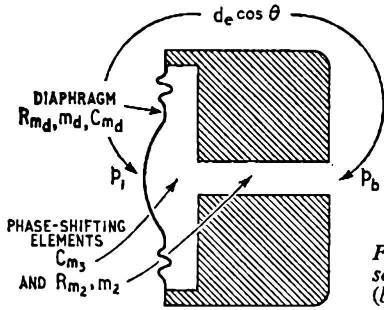
On the $\theta = 180°$ axis, the acoustic wave acts first on the rear aperture of the microphone and is represented by p_b in Fig. 9.14 (c). The acoustic pressures then reach the front and rear surfaces of the diaphragm by traversing the paths d_e and d_i; since these are acoustically of identical length, the pressures on either side of the diaphragm are equal in both amplitude and phase, and there is then no resultant pressure to cause the diaphragm to move, and hence no output from the microphone on the $\theta = 180°$ axis.

At high frequencies, diffraction effects reduce the amplitude of the pressure at the annular aperture and hence the pressure p_2 at the back of the diaphragm is also reduced.

When the dimensions of the microphone are comparable with or greater than the wavelength, p_2 is of negligible amplitude; the microphone is then pressure-operated and depends on diffraction effects for its uni-directional characteristic.

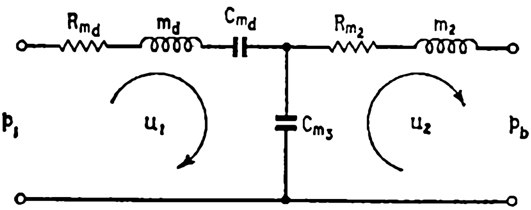
9.5. The Phase-shift Network

We have seen that the diaphragm of a phase-shift microphone, like the ribbon of a pressure-gradient microphone, is open to acoustic pressures at both front and rear faces. The essential feature which distinguishes the phase-shift type of microphone from the pressure-gradient type and gives it its uni-directional property, is the provision within the body of the microphone of a phase-shifting network coupling the back of the diaphragm to the external acoustic pressures. The vector analysis illustrated in Fig. 9.14 (a), (b) and (c)



(a)

Fig. 9.15.—(a) Diagrammatic representation of a phase-shift microphone. (b) The equivalent circuit of diaphragm and phase-shifting networks



(b)

shows that if a cardioid characteristic is required, the phase shift introduced by this network must equal the phase shift provided by the external path length.

The phase shift associated with external path d_e is proportional to frequency, for

$$\begin{aligned} kd_e &= 2\pi f \frac{d_e}{c} \\ &= af \end{aligned}$$

where a is a constant.

It is therefore a requirement of the internal phase-shifting arrangement that the phase shift provided is proportional to frequency and equal in magnitude to that introduced by the external path.

In a practical design the phase-shifting network consists of cavities and narrow passages which convey the acoustic pressures to the back of the diaphragm. A simplified arrangement illustrating the principle is shown in Fig. 9.15 (a). The acoustic pressures reach the back of the diaphragm via the narrow passage and air cavity immediately behind the diaphragm. The impedance of the air in the passage is $R_{m2} + j\omega m_2$ and that of the cavity behind the diaphragm $\dfrac{1}{j\omega C_{m3}}$. The equivalent circuit of the phase-shifting network and diaphragm is shown in Fig. 9.15 (b).

If p_1 is the pressure on the front of the diaphragm and p_b the pressure at the entrance to the aperture or passage, the velocity u_1 of the diaphragm is (Appendix 11)

$$u_1 = \frac{p_1\left[Z_2 + Z_3\left(j\frac{\omega}{c}\, d_e \cos\theta\right)\right]}{Z_1Z_2 + Z_1Z_3 + Z_2Z_3}$$

where Z_1 = impedance of the diaphragm,
Z_2 = impedance of the passage,
Z_3 = impedance of the cavity behind the diaphragm,
d_e = external path length,
θ = angle of incidence.

Arranging the equation for a better understanding of the factors affecting the choice of the impedances,

$$u_1 = \frac{p_1}{Z_1Z_2 + Z_1Z_3 + Z_2Z_3}\left[Z_2 + Z_3\left(j\frac{\omega}{c}\, d_e \cos\theta\right)\right]$$

Z_1, the impedance of the diaphragm, will determine the frequency response of the microphone and its choice will be dependent on the type of transducer employed, i.e., crystal, electrostatic or electrodynamic. Z_2, Z_3 and d_e affect the sensitivity of the microphone but Z_2 and Z_3 also determine the shape of the directional characteristic. Considering only that part of the equation which affects the directional characteristic we see that

$$\left[Z_2 + Z_3\left(j\omega\frac{d_e}{c}\cos\theta\right)\right]$$

is of the form $a + b\cos\theta$.

Equations of the form $R_\theta = a + b \cos \theta$ are called " limacons " and their polar graph takes many shapes depending on the values of a and b.

If $a = 0$ then $R_\theta = b \cos \theta$ and the polar graph is the familiar bi-directional pattern associated with pressure-gradient microphones. If $b = 0$ then $R_\theta = a$; this gives an omni-directional characteristic. Other values of a and b result in polar graphs which are in effect combinations of the omni-directional and bi-directional patterns, depending on the magnitude of a and b. If $a = b$ then $R_\theta = a(1 + \cos \theta)$ and the characteristic is a cardioid.

If a cardioid characteristic is required, Z_2 and Z_3 must be so arranged that

$$Z_3 j\omega \frac{d_e}{c} = Z_2$$

or

$$\frac{Z_3 j\omega\, d_e}{cZ_2} = 1$$

If in the design the impedance backing the diaphragm is a compliance of value C_{m3}, substituting for Z_3 the reactance $\frac{1}{j\omega C_{m3}}$, we have

$$\frac{j\omega d_e}{cZ_2 j\omega C_{m3}} = 1$$

$$Z_2 = \frac{d_e}{cC_{m3}}$$

This indicates that Z_2 is a resistance, and writing R_2 for Z_2, its value is given by

$$R_2 = \frac{d_e}{cC_{m3}}$$

If however Z_2 is a mass m_2, then Z_3 is a resistance R_3 of value

$$R_3 = \frac{Cm_2}{d_e}$$

Electrically it is difficult, under certain conditions, to produce a non-inductive resistor; it is also difficult acoustically to arrange the dimensions of the passage or slots so that they possess resistance and little inertance. By making the thickness of the slot normal to the direction of air flow small, the resistance-to-mass ratio is improved (Appendix 6).

9.6. Electrodynamic Mass-controlled Phase-shift Microphones

Electrodynamic phase-shift microphones first appeared in the 1930s and since that time have been continuously developed. Modern magnetic materials have enabled designers to reduce the size of the instruments, enabling them to compete successfully with other types of uni-directional microphones used in television and sound broadcasting studios.

9.6.1. OPERATION

A diaphragm-operated moving-coil unit is usually employed as the transducing element, and Fig. 9.16 shows diagrammatically the construction of a microphone of this type. Side-openings in the case permit the acoustic pressures to reach the air cavity behind the diaphragm via the space between the moving coil and the pole-pieces. The pole-piece slot and the air cavity comprise the main phase-shifting network, the slot providing the resistance and the cavity the

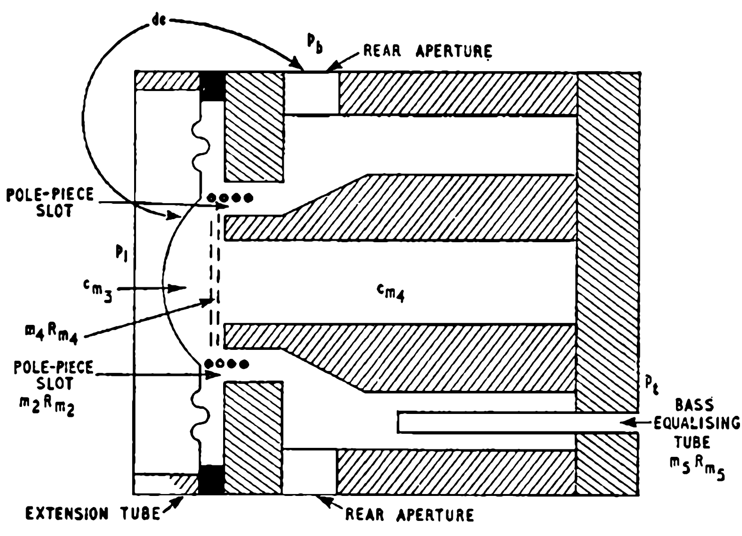


Fig. 9.16.—Cross-sectional representation of a uni-directional moving-coil microphone

required compliance. The resistance of the slot is obtained from the viscous loss introduced when the air particles move in a restricted space. Because a mass of air has to be moved before resistance is apparent, the impedance of the slot has a reactive as well as a resistive component and is of the form $R_m + j\omega m$.

The slot impedance (*see* Appendix 6) is

$$\frac{12\mu l}{t^3 w} + j\frac{\omega^6 \rho l}{5\,tw}$$

where l = length of the slot in the direction of flow,
t = thickness of the slot normal to the direction of flow,
w = width of the slot normal to the direction of flow,
μ = viscosity coefficient for air,
ρ = density of air.

The resistance to mass ratio is

$$\frac{12\mu l 5dw}{t^3 w\omega^6 \rho l} = \frac{60\mu}{t^2\omega^6\rho}$$
$$\propto \frac{1}{t^2}$$

The ratio is large if t is small.

In the analogous circuit of Fig. 9.17 the impedance of the slot is represented by the series branch of the phase shifting network

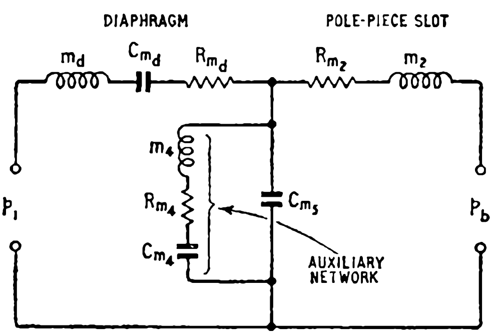


Fig. 9.17.—Equivalent circuit of a phase-shift microphone

R_{m2}, m_2 and the compliance of air behind the diaphragm by C_{m3}. The dimensions of the slot cannot be decided solely by the need to provide the necessary acoustic resistance; other factors have to be considered. The criterion is generally the safe clearance between the coil and the pole-pieces and this usually results in a slot whose resistance to mass ratio is too low.

9.6.2. FREQUENCY RESPONSE

The frequency response of the microphone is largely dependent on the mechanical impedance of the diaphragm system. Assuming the network introduces a phase shift proportional to frequency, the velocity of the diaphragm (*see* Section 9.5) is

$$u = \frac{2Ap_1\pi f(d_e + d_i)}{cZ_m}$$

$$\propto \frac{f}{Z_m}$$

The velocity, and hence the output of the microphone, will be independent of frequency if the diaphragm is mass-controlled; that is, its natural frequency must be low and preferably outside the audio-frequency range.

With a ribbon-type microphone, this presents few difficulties, but mass control of a moving-coil diaphragm system is difficult to achieve, especially at low audio frequencies. If in order to reduce the natural frequency of the system the diaphragm is provided with a very flexible suspension, it may be difficult to ensure that the coil is always correctly positioned in the magnet air gap. Furthermore, a compliant suspension increases the microphone's susceptibility to mechanical shock and tends to limit its use in operation.

In practice it is difficult to design a diaphragm and moving-coil system with a natural frequency of resonance less than 200 c/s: but the natural frequency of the system can be reduced indirectly. The auxiliary network m_4, C_{m4}, R_{m4} (Figs. 9.16 and 9.17) is in the form of a damped Helmholtz resonator and is used to increase the air load on the diaphragm, thus reducing the natural frequency of the system by about an octave and so improving the response of the microphone at low frequencies.

Mass control is easy to achieve at high frequencies, and so long as the driving force continues to increase with frequency, the conditions for uniform sensitivity hold, but in the region of 4,000 c/s the dimensions of most practical microphones are comparable with the wavelength and the microphone tends to be pressure- rather than phase-shift operated. Under these conditions, the driving force is constant and the output of the microphone with a mass-controlled diaphragm system falls at the rate of 6 dB per octave, since the velocity $u \propto 1/2\pi fm$. The high-frequency response can be improved by placing an extension tube in front of the diaphragm. The increased pressure on the diaphragm which results from cavity resonance counteracts the increasing mass reactance, and extends the frequency range of the instrument to about 10,000 c/s. Although the microphone may be virtually pressure-operated at the high frequencies, the uni-directional characteristic is maintained by diffraction effects from the microphone case or housing.

If the natural frequency of the diaphragm system is within the audio-frequency range, the diaphragm velocity falls steeply at frequencies below that of the natural frequency of the system. The mechanical impedance of the diaphragm is no longer dependent on its mass reactance, but tends to be compliance-controlled at low frequencies. Then

$$u \propto f \Big/ \frac{1}{\omega C_{md}}$$

$$\propto f^2$$

and as the frequency is reduced the output falls at the rate of 12 dB per octave.

The response at low frequencies can be improved by the use of a bass equalising tube. Its function is to provide a phase-shifting network, operating at low frequency only, and so arranged that there is a partial resonance between the mass reactance of the air in the tube and the compliance reactance of the volume of air in the case. The dimensions of the tube are chosen so that the acoustic resistance is low and an appreciable pressure is set up behind the diaphragm, the phase being such that it augments the resultant pressure on it and so overcomes the large compliance reactance of the system.

The bass equalising tube, together with the diaphragm cavity resonator, extends the frequency range of the microphone into regions where the phase-shift principle acting alone would produce unsatisfactory results.

At medium and high frequencies the bass equalising tube is inoperative, because of the high impedance it presents to the acoustic pressures. The increase in the impedance is due to the increase with frequency of the large mass reactive component of the tube impedance. The sensitivity and frequency response of a typical uni-directional dynamic microphone are shown in Fig. 9.18. The

Plate 7.1. — Moving-coil microphone showing the concentric corrugations in the diaphragm (courtesy of S. T. & C.)

Plate 7.2.—Moving-coil microphone, with a spherical case and disc-shaped screen whose diffraction effects can be predicted mathematically

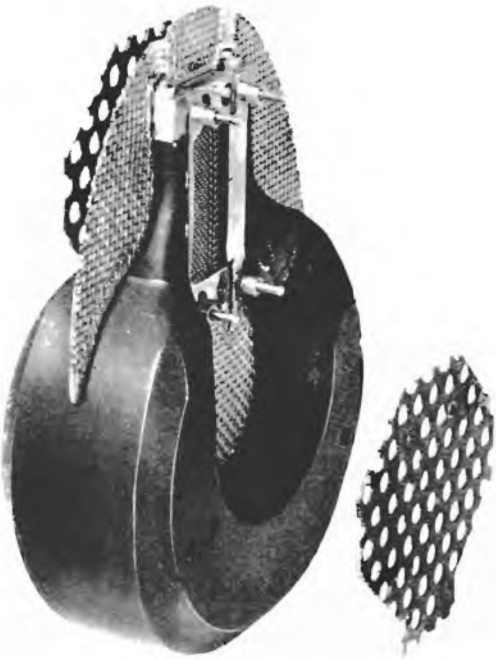


Plate 8.1. — Close-fitting fabric baffle used to increase the low-frequency sensitivity

Plate 8.2.—Ribbon microphone with the width of the case reduced to improve high-frequency response

Plate 9.1.—A high quality two-element uni-directional microphone (courtesy of S. T. & C.)

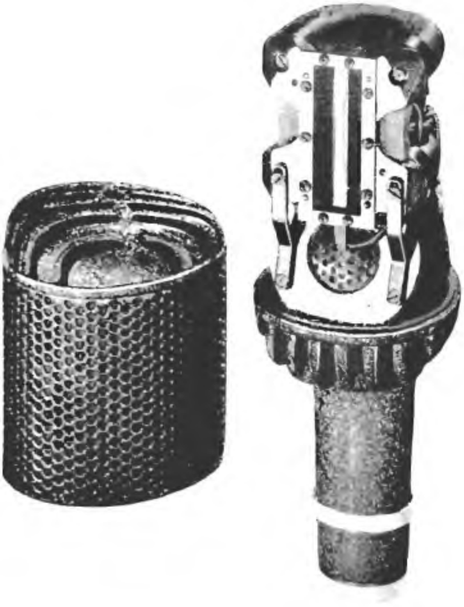


Plate 9.2.—A uni-directional dynamic microphone employing a slotted multi-path tube

Plate 10.1.—BBC commentator's microphone with carrying case and variable equalizer

Plate 10.2.—Exploded view of commentator's microphone showing lip-guard microphone inset and breath shields

(*a*)

(*b*)

Plate 11.1.—(a) Tubular line directional microphone manufactured by Sennheiser Electronic Company

(b) The pressure unit and part of the tubular line pick-up system

Plate 11.2.—Paraboloid reflector microphone used for outside broadcasts

Plate 12.1. — Silentbloc rubber diaphragm

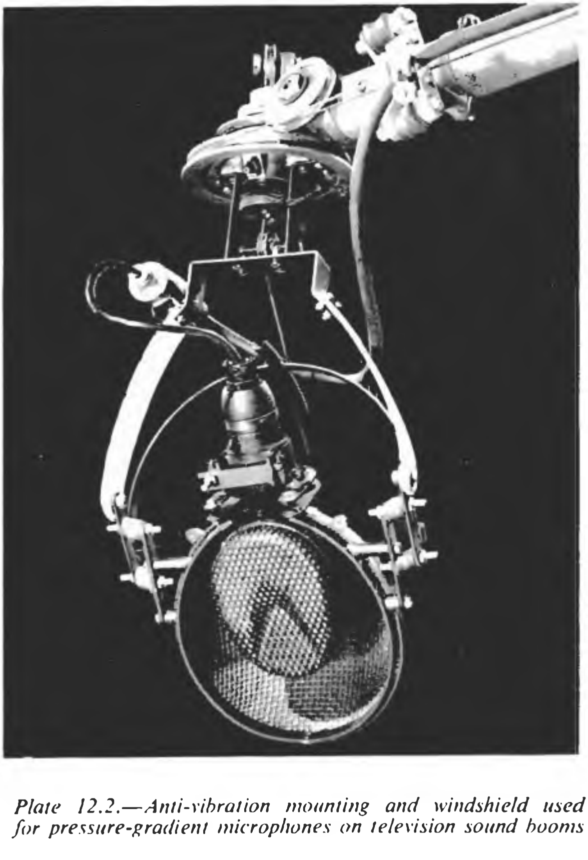


Plate 12.2.—Anti-vibration mounting and windshield used for pressure-gradient microphones on television sound booms

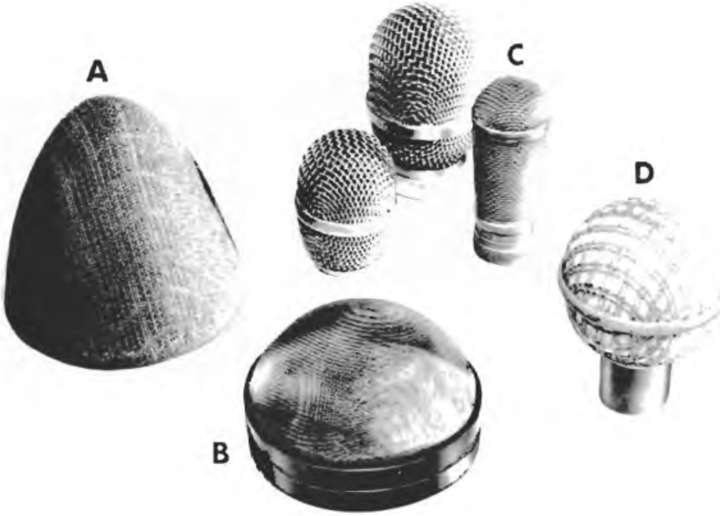


Plate 12.3.—Windshields representative of modern practice (B.C.D), compared with an obsolete type A

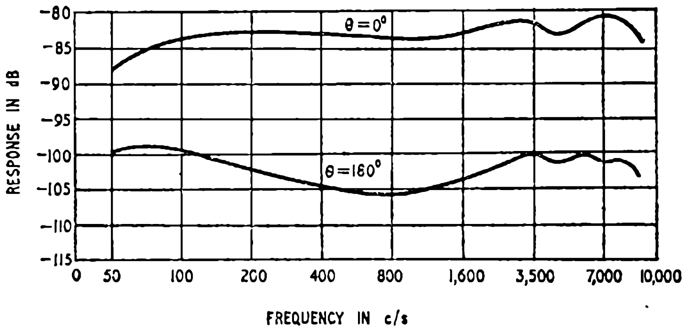


Fig. 9.18.—Typical free field response of a uni-directional microphone

curves show that the front-to-back ratio is a maximum at medium frequency, but is somewhat smaller at the low frequency when directional characteristics tend towards an omni-directional rather than a uni-directional pattern.

9.7. Electrodynamic Resistance-controlled Phase-shift Microphones

We have already seen that the force available to operate the diaphragm of a phase-shift microphone is proportional to frequency, and if an electrodynamic transducer is employed, the diaphragm system must be mass-controlled. Unfortunately, a mass-controlled system is susceptible to mechanical shock and the voltages which are produced when the microphone is suddenly accelerated or decelerated give rise to noises of low pitch and of random frequency. Because of their desirable directivity characteristic, uni-directional microphones are often employed in television studios where their use on sound booms unavoidably subjects them to shocks of this nature. A uni-directional microphone having a resistance-controlled diaphragm system would be preferable, because the signal-to-shock ratio would be greater than that of a comparable mass-controlled instrument.

Resistance-controlled dynamic systems are associated with pressure-operated microphones where the force available is independent of frequency. An examination of the equation for the pressure on the diaphragm of a phase-shift microphone (page 145) shows that the pressure is proportional to the frequency and to the path length:

$$p \propto f(d_e + d_i)$$

If a microphone could be produced whose path length varied inversely with frequency, the driving force would be constant and a resistance-controlled system would then be required.

9.8. Multi-path Microphones[3]

In 1954 A. M. Wiggins described a microphone with three sound entrances in the body of the instrument. Each entrance admitted sound pressures to the back of the diaphragm by separate paths which were frequency-dependent.

Fig. 9.19 shows diagrammatically the construction of the microphone, with the three sound ports G, H and J. The longest external path d'_e is that associated with the low-frequency sounds which enter by the rear port G, while the shortest path d'''_e is traversed by the high frequencies which reach the back of the diaphragm via the port J. The design of the phase-shift networks associated with G and J restricts the medium frequencies to the path d''_e whose entrance is at H.

9.8.1. ACTION OF THE PORTS

The selective nature of the paths can best be explained by reference to the equivalent circuit Fig. 9.20 in conjunction with Fig. 9.19. The impedance of the diaphragm assembly is represented by m_d, C_{m_d} and R_{m_d} and the compliance of the air space immediately behind the diaphragm by C_{m_1}. The shunt circuit m_6, R_{m_6}, C_{m_6} is used to improve the frequency response of the microphone and is not important in the discussion of the multi-path principle.

Low-frequency sound pressures enter the air space C_{m_2} via the port G and pass through the acoustic impedance m_2, R_{m_2} to the air cavity C_{m_3}. They then reach the air space C_{m_1} at the back of the diaphragm via the acoustic impedance m_5, R_{m_5}. The equivalent circuit shows that C_{m_2}, m_2, R_{m_2}, C_{m_3}, m_5, R_{m_5} and C_{m_1} are arranged in the form of a low-pass filter which permits low-frequency sounds a free passage to the air space at the back of the diaphragm. Sounds of medium and high frequency entering at the port G are heavily attenuated and contribute little to the resultant pressure at the back of the diaphragm.

The port H is fitted with a light flexible membrane which is arranged to be transparent to sounds of medium frequency. The membrane and the mass reactance and resistance of the air in the passage behind it together constitute the series circuit m_3, R_{m_3}, C'_{m_3} in Fig. 9.20. The natural frequency of the circuit is of the order of 500 c/s. At low frequencies its impedance is high because of the large compliance reactance of the port, and low-frequency sounds are virtually excluded. High-frequency sounds are also prevented from

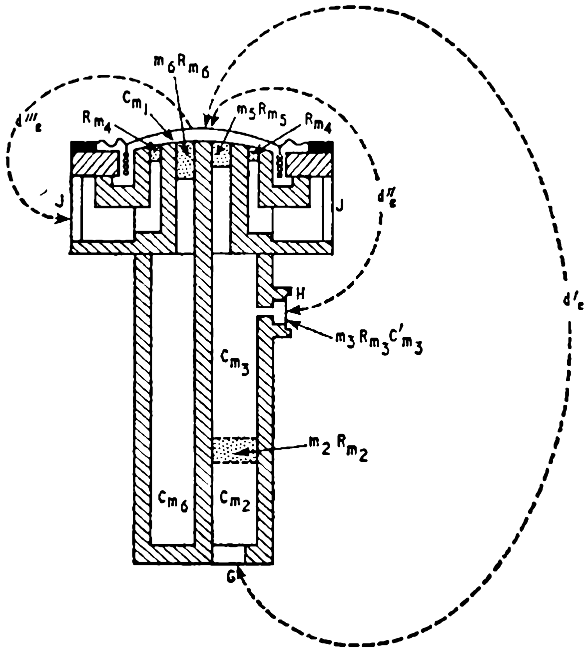


Fig. 9.19.—Cross-sectional representation of a resistance-controlled phase-shift dynamic microphone

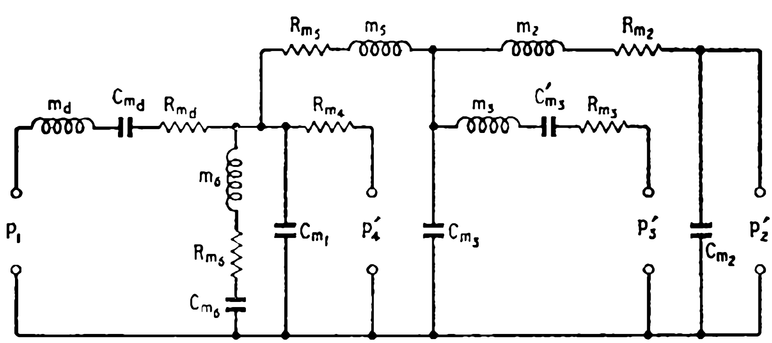


Fig. 9.20.—Equivalent circuit of a resistance-controlled phase-shift dynamic microphone

entering because of the large mass reactance of the network at high frequencies. Thus entry at the port H is restricted to sounds of medium frequency—that is, to sounds at frequencies at or near the natural frequency of the series network.

9.8.2. VECTOR DIAGRAM OF MULTI-PATH MICROPHONE

The vector diagrams (a) and (b) of Fig. 9.21 show the pressures on the diaphragm of the microphone for low and medium frequency sounds. Diagram (a) represents the conditions when sound enters by the low-frequency port G: p_1 is the pressure on the front and p_2 on the back of the diaphram. The wavelength constant k is

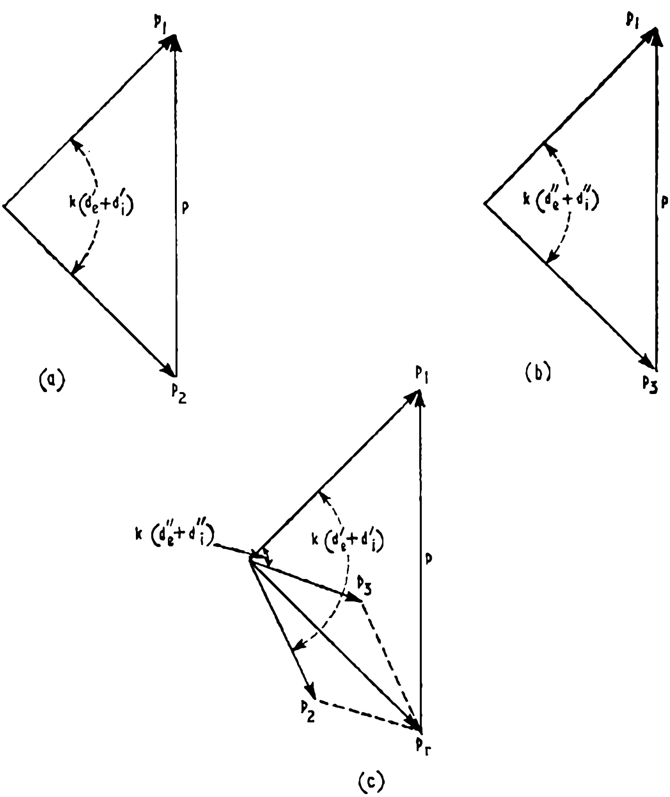


Fig. 9.21.—Vector diagrams showing the pressures on the diaphragm of a multi-path phase-shift microphone: (a) at low frequency, (b) at medium frequency and (c) at a frequency intermediate between medium and low

small at low frequency, but the external path d_e' is long and the phase shift associated with the internal path d_i' is large: hence the phase angle $k(d_e' + d_i')$ between the pressure vectors is also large and an appreciable force is available to actuate the diaphragm, even at low frequency.

Fig. 9.21 (b) shows the conditions at medium frequency when the sound port H is operative. As before, p_1 is the pressure at the front of the diaphragm: p_3 is the pressure at the back produced by sounds which enter by the port H. The phase angle between p_1 and p_3 is $k(d_e'' + d_i'')$. Although k has increased with increase in frequency, the phase angle remains substantially constant because the external path d_e'' and the internal path d_i'' have shortened, the decrease in the length of the combined path being proportional to the increase in frequency. Hence the phase angle between the vectors p_1 and p_3 in Fig. 9.21 (b) is equal to the phase angle between p_1 and p_2 of Fig. 9.21 (a) and the force available to operate the diaphragm is independent of the change in frequency.

At a frequency intermediate between medium and low frequency, sounds enter by both ports, G and H. In Fig. 9.21 (c) p_2 represents the pressure contribution of the port G in magnitude and phase. The amplitude of p_2 in diagram (c) is less than the amplitude of p_2 in diagram (a) because the frequency is such that the acoustic pressures passing through the low-pass filter suffer some attenuation. Similarly, p_3, the pressure contribution of port H, is attenuated in its passage through the series network.

The two pressures p_2 and p_3 combine to give a resultant pressure p_r at the back of the diaphragm which is equivalent to the sound pressure which would have been produced had there been a sound port somewhere along the axis between the ports G and H.

At frequencies of the order of 1,000 c/s, sounds enter by the ports H and J. This again produces a pressure at the back of the diaphragm equivalent to that from a sound aperture situated somewhere between the medium-frequency port H and the high-frequency port J.

Above 2,000 c/s sounds enter only by the high-frequency port J and although the path length $(d_e''' + d_i''')$ is small, k is large, and as before, the phase angle between the pressures at front and back of the diaphragm is virtually independent of frequency. Thus the force available to operate the diaphragm is constant and a resistance-controlled system can be employed with the added advantage of freedom from mechanical shock.

If the directional characteristic of the microphone is to be cardioid, then the conditions outlined in Section 6.3 must be observed, i.e., the length of the external path at any angle of incidence θ must be proportional to cos θ and for axial sounds when $\theta = 0°$, the lengths of the external and internal path must be equal.

9.9. Variable Path-length Uni-directional Microphones

Other methods of obtaining an acoustic path whose length varies with frequency have been developed by Continental manufacturers and incorporated in their multi-path uni-directional microphones. One of these microphones is illustrated in Plate 9.2. The multi-path section is accommodated in the slotted tube, which is visible in the diagram behind the transducer section of the microphone proper. The slots, of which there are several per inch of tube length, allow sounds to enter the microphone body at a large number of different points. The tube is filled with a porous material and so arranged that it acts as a " transmission line " whose impedance is proportional to its length and whose reactance is proportional to frequency.

9.9.1. OPERATION

The general construction of a microphone of this type is shown in the simplified schematic section of Fig. 9.22, and its equivalent circuit

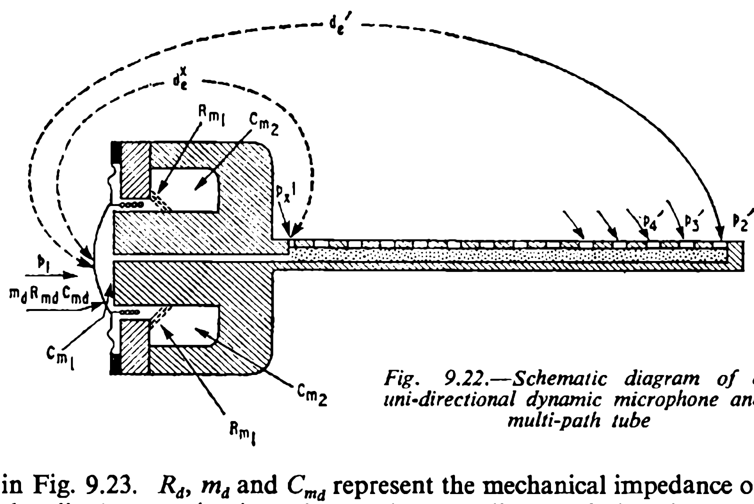


Fig. 9.22.—Schematic diagram of a uni-directional dynamic microphone and multi-path tube

in Fig. 9.23. R_d, m_d and C_{md} represent the mechanical impedance of the diaphragm circuit and C_{m_1} the compliance of the air space immediately behind it. The shunt circuit R_{m_1}, C_{m_2} provides the necessary acoustic damping to ensure that the diaphragm and coil assembly are resistance-controlled.

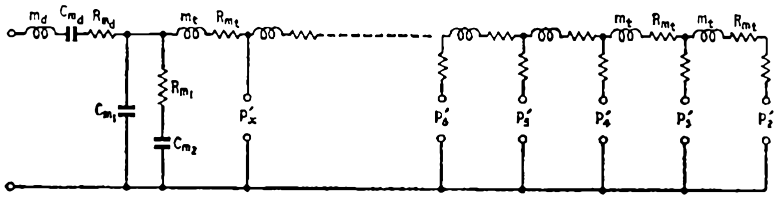


Fig. 9.23.—Equivalent circuit of a uni-directional microphone with multi-path tube

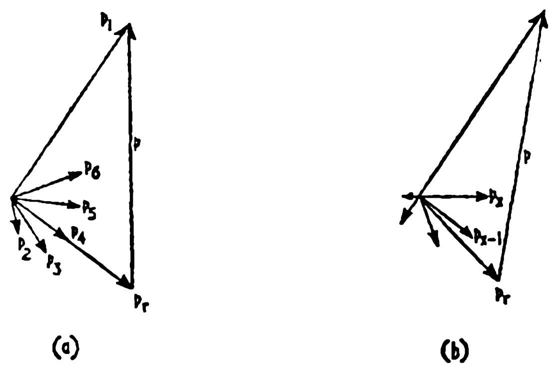


Fig. 9.24.—Vector diagram showing the pressures on the diaphragm of a microphone employing a multi-path tube (a) at low frequency, (b) at high frequency

An element of the tube transmission line is represented in the equivalent circuit by an impedance of the form $R_m + j\omega m$ so that the impedance of the tube as a whole is proportional to its length and approximately proportional to frequency.

At low frequencies, sound pressures enter at all the slots along the whole length of the tube. The impedance of the tube is low at low frequency and the pressure contributions from the slots are therefore of approximately equal amplitude and differ only in phase because of the different lengths of the sound paths involved.

The vector diagram of Fig. 9.24 (a) shows the condition at low frequencies for sound arriving on the axis of the microphone: p_1 is the pressure at the front of the diaphragm. The pressure p_r at the back of the diaphragm is the resultant of the pressure contributions from all the slots. The vector difference of p_1 and p_r is the pressure available to operate the diaphragm and is represented by p.

At high frequency, when the length of the tube is comparable with the wavelength of the sound, the pressures at the distant end p_2', p_3', etc., would be out of phase with the pressure at the diaphragm end and cancellation might occur. But at high frequency the impedance of a long length of the tube is very large, and little sound arrives from the distant end. The tube behaves as though distant slots were closed and only the slots at the diaphragm end contribute pressures which need be considered.

Because the impedance of even a short length of tube is significant, the pressure contributions from the operative slots will differ in magnitude as well as phase and will depend on the length of the external and internal path traversed. Although the paths are short at high frequencies, the phase angle $k(d_e + d_i)$ is large, since k is proportional to frequency.

The vector diagram of Fig. 9.24 (b) shows the conditions at high frequency: as before, p_1 represents the pressure at the front of the diaphragm, and p_r, the resultant pressure at the back, is the sum of the contributions from the active slots. These differ in amplitude as well as phase, the pressures from the distant slots having maximum phase angle but minimum amplitude. The pressure available to actuate the diaphragm is p and a comparison of Fig. 9.23 (a) and (b) shows that although there has been a slight change in the phase of p, its amplitude remains substantially independent of frequency. Thus a resistance-controlled diaphragm system could be employed in conjunction with an electrodynamic transducer.

9.10. Uni-directional Electrostatic Microphones

One of the most successful of modern uni-directional microphones is the twin-diaphragm electrostatic instrument. Microphones of this type are developments of the original design by Braunmühl and Weber who, as early as 1935, produced the first electrostatic cardioid microphone.

The essential features of the microphone are shown in simplified form in Fig. 9.25. Two light diaphragms D_a and D_b are positioned in close proximity to, and on either side of a circular central electrode which forms the body of the microphone. If a uni-directional characteristic is desired, only one of the diaphragms (D_a) is connected to the polarising supply. The other diaphragm (D_b) is inoperative electrically, but forms part of the phase-shifting network on which the instrument depends for its directional properties.

A number of small holes, about 20 in all, convey the acoustic pressures from the inoperative diaphragm D_b to the back of the live diaphragm D_a which moves as a result of the pressure differences on its front and rear faces.

The output of the microphone is dependent on the variations in capacitance between the diaphragm D_a and the central electrode and is proportional to the displacement of the diaphragm about its

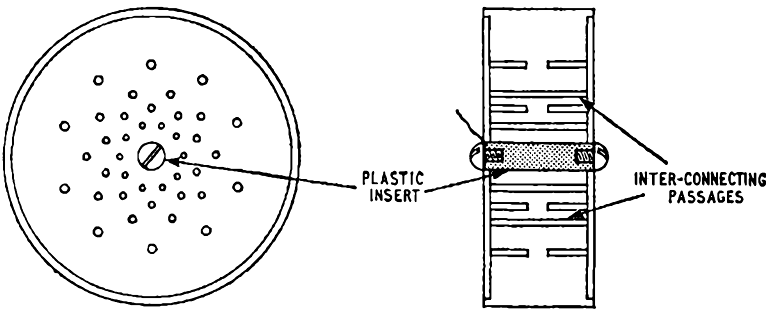


Fig. 9.25.—A diagrammatic representation of a twin-diaphragm uni-directional electrostatic microphone

mean position. Since the force on the diaphragm is proportional to frequency, the displacement is:

$$x \propto \frac{f}{\omega Z_{md}}$$
$$\propto \frac{1}{Z_{md}}$$

where Z_{md} is the impedance of the diaphragm system.

If uniform response is required, the displacement x must be independent of frequency: hence the impedance of the diaphragm must also be independent of frequency, that is, the resistive component of the impedance must be large in comparison to the mass and stiffness.

This is difficult to achieve directly, but by making use of the air space immediately behind the diaphragm, the resistance of the system as a whole can be materially increased. The impedance of the air space behind the diaphragm is of the form $R + \frac{1}{j\omega C_m}$ and is effectively in series with the diaphragm impedance. Reducing the depth of the air space will increase the resistive component by increasing the viscous losses, but unfortunately this also produces an increase in stiffness associated with the small volume of air in the cavity. The stiffness of the system can be reduced by drilling a number of deep holes of small diameter in the central electrode. Unlike the holes previously referred to, these holes do not pass through the electrode. Their purpose is to increase the volume of air behind the diaphragm, thus reducing the compliance reactance of the system as a whole. By varying the spacing between the diaphragm and the central electrode, the resistance and natural frequency of the diaphragm can be controlled. The spacing is critical and a variation of 3 to 5 μ (1 μ (micron) = 0·001 mm) in a spacing of 40 μ can produce noticeable changes in sensitivity and frequency response.

9.10.1. DIAPHRAGMS

The diaphragms of uni-directional electrostatic microphones produced immediately after the war were made of polyvinyl chloride with a thickness of only 8 μ and were coated with gold by a sputtering process. To-day nickel diaphragms have replaced many of the early plastics. When plastics are employed, the material is selected with due regard to the high temperature which may be experienced in the vicinity of television lighting units or footlights.

9.10.2. OPERATION

As has already been explained, if a uni-directional characteristic is desired, only one diaphragm D_a is connected to the polarising

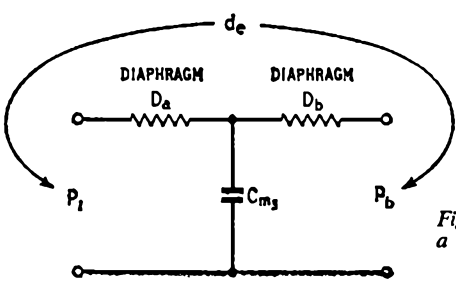


Fig. 9.26.—Equivalent circuit of a uni-directional twin-diaphragm electrostatic microphone

supply (Fig. 9.26). The electrically inoperative diaphragm D_b acts as the series element of the phase-shifting network, the shunt element being supplied by the compliance reactance of the air in the interconnecting passages. If the network introduces attenuation and a phase shift similar to that experienced by the sound wave following the external path d_e from front to back of the instrument, the polar response R_θ of the microphone is

$$R_\theta = a\,(1 + \cos\theta)$$

where a is a constant, that is, the directional characteristic is a cardioid.

The diaphragms are symmetrically arranged relative to the central electrode and either diaphragm can be connected to the supply

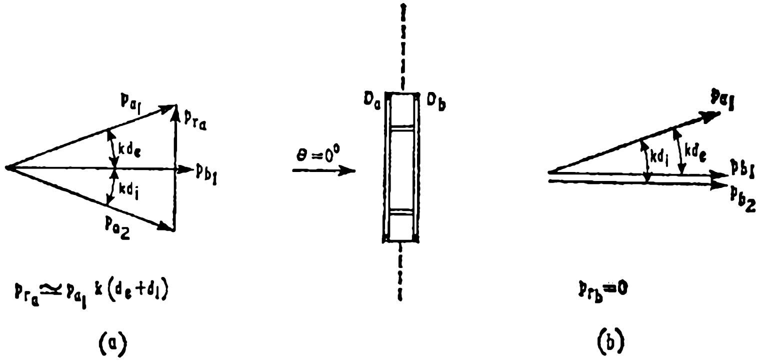


Fig. 9.27.—Vector diagram showing the pressures on the twin diaphragms of an electrostatic microphone at $\theta = 0°$: (a) pressures on the front diaphragms D_a*; (b) pressures on the rear diaphragm* D_b

without affecting the general shape of the directivity pattern. Although in this example diaphragm D_a is assumed to be connected, and only its variation about its mean position constitute the output of the microphone, it is nevertheless instructive to examine the forces on both diaphragms for sound incident at $\theta = 0°$. They are shown in Fig. 9.27 (a) and (b).

The pressure on the outer surface of diaphragm D_a is represented in Fig. 9.27 (a) by p_{a_1}. The pressure p_{b_1} on the outer surface of diaphragm D_b lags on p_{a_1} by angle kd_e where d_e is the external path length. The pressure on the inner surface of diaphragm D_a is p_{a_2} lagging on p_{b_1} by angle kd_i where d_i is the internal path length. If $(d_e + d_i)$ is small compared with the wavelength, the resultant pressure p_{r_a} on D_a is

$$p_{r_a} = p_{a_1} k\,(d_e + d_i)$$

At $\theta = 0°$ the resultant pressure on diaphragm D_b is zero (Fig. 9.27 (b)), because the pressure p_{b_1} on the outer surface of D_b lags on p_{a_1}

by angle kd_e and the pressure on the inner surface p_{b2} lags on p_{a1} by angle kd_i. Since the path lengths d_e and d_i are equal, the pressures on either side of D_b are equal in both amplitude and phase and consequently no resultant pressure is exerted on diaphragm D_b.

If, however, the sound wave is incident on the microphone at $\theta = 180°$, no resultant force is exerted on diaphragm D_a for the same reason and there is no output from the microphone. This is, of course, a requirement for the cardioid characteristic.

9.10.3. VARIABLE DIRECTIONAL CHARACTERISTICS

If both diaphragms are connected to the supply and appropriately polarised, they are equivalent to two identical cardioid microphones with their axes of maximum sensitivity at 180° to each other.

The polar response $R_{a\theta}$ of diaphragm D_a is

$$R_{a\theta} = a\,(1 + \cos\theta)$$

where a is a constant and the polar response $R_{b\theta}$ of diaphragm D_b is, from Fig. 9.28:

$$\begin{aligned} R_{b\theta} &= a\{1 + \cos(180° - \theta)\} \\ &= a\,(1 - \cos\theta) \end{aligned}$$

By combining the outputs from the diaphragm in various ways, a microphone with a variable directional characteristic is obtained. The combination of the outputs and the degree to which each diaphragm contributes is determined by the polarising potentials.

Fig. 9.29 shows the circuit arrangements for varying the polarising supply. Diaphragm D_a is at a fixed potential with reference to the central electrode, but the potential on D_b varies with the setting of the potentiometer slider. When the slider is in the position shown in Fig. 9.29, diaphragm D_b is of the same potential as the central electrode and is therefore inoperative. The polar response R_c of the combination is determined by D_a only; hence

$$R_c = a\,(1 + \cos\theta)$$

As already explained, this gives a cardioid directional pattern. When the slider is moved to position O the polarising voltage on D_b is such that the outputs of the two diaphragms are added. The polar response of the combination is now

$$\begin{aligned} R_c &= a\,(1 + \cos\theta) + a\,(1 - \cos\theta) \\ &= 2a \end{aligned}$$

The output is independent of the angle of incidence and the response is omni-directional. As the slider is moved gradually from the central position towards O, the characteristic changes from the cardioid to an omni-directional pattern. If the slider is moved towards position B, the potential on diaphragm D_b is reversed

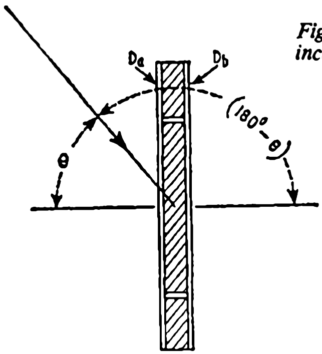


Fig. 9.28.—Diagram showing the angle of incidence relative to the front and rear diaphragm D_a *and* D_b

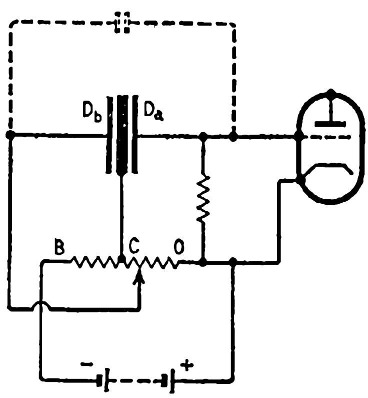


Fig. 9.29.—Circuit arrangements for varying the polarising supply to the rear diaphragm D_b

and the outputs of the two diaphragms are in opposition. The polar response R_c of the combination is

$$R_c = a\,(1 + \cos\theta) - a\,(1 - \cos\theta)$$
$$= 2a\cos\theta$$

The characteristic is a bi-directional figure-of-eight. Intermediate positions of the slider between C and B give a characteristic intermediate between the cardioid and the figure-of-eight.

If the directional characteristic is controlled by a switch rather than by a potentiometer, the time constant of the polarising circuits must be sufficiently long, usually several seconds, to avoid switching clicks.

9.10.4. SENSITIVITY AND RESPONSE

The sensitivity of an electrostatic microphone is generally about 20 dB above that of a comparable electromagnetic type, but the relatively small capacitance between diaphragm and central electrode —less than 80pF—requires that it be installed close to the first stage of the amplifier. Miniature valves and small components have enabled designers to produce pre-amplifiers of small physical size which do not materially affect the directional characteristic of the instrument. The frequency response of certain types of electrostatic

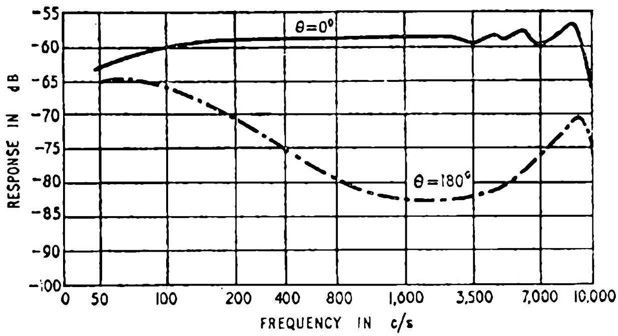


Fig. 9.30.—Frequency response curves of a uni-directional electrostatic microphone

microphones is remarkably smooth and a front-to-back ratio of about 15 dB to 20 dB can be obtained over a band of frequencies greater than three octaves, usually in the region of 400 to 1,600 c/s.

The general shape of the response curves at $\theta = 0°$ and $\theta = 180°$ is shown in Fig. 9.30. While the front to back ratio is satisfactory at medium and at high frequency, the curves indicate that the microphone tends to become omni-directional at the low frequencies.

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2. Bauer, B. B., " Uniphase uni-directional microphones ", *J. Acoust. Soc. Am.*, *13* (1941).
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10

Anti-noise Microphones

THIS CHAPTER is concerned with microphones designed for a specific purpose and used only in special circumstances, or in noisy acoustic conditions. Their method of operation will not necessarily involve new principles, but will either combine types of microphones which have already been discussed, or will exploit a property associated with a particular type of microphone for a particular purpose.

10.1. History of Anti-noise Microphones

In the early days of broadcasting it was the practice when commentaries had to be transmitted from outside events, inseparable from a background of crowd noise, to house the commentator and his microphone in a sound-proof booth. The booth was small because it had to be portable and was lined with a material, usually building board, having relatively poor sound-absorbing properties, especially at the low frequencies. The dimensions of the booth were such that the frequencies of the low-order eigentones were within the male speech band, and as a consequence the speech quality was poor. In 1937 the BBC Research Department developed a close-talking microphone with noise-reducing properties and the special sound-proof booths for commentaries were no longer required.

The new microphone was of the pressure-gradient type and while the quality of the speech output was somewhat inferior to that of the pressure-gradient microphones employed in studios, it was as good as that obtained from many of the pressure-operated moving-coil microphones of the period, if not better. It was not the first noise-reducing microphone, for the problem had been encountered before in a more acute form when speech had to be transmitted from fighting ships and fighting vehicles under warfare conditions. Anti-noise microphones were in use in the First World War, but the BBC *lip ribbon* microphone, as it was originally called, is believed to have been the first anti-noise microphone which produced speech of a quality acceptable for broadcasting.

10.2. Principle of Operation

The noise-reducing effect depends on the ability of the microphone to discriminate between the wanted speech and the unwanted noise; this noise is usually produced by a large source of sound, such as a crowd, or originates at some distance from the microphone, so that the wavefront associated with the noise tends to be plane rather than spherical. On the other hand, the acoustic pressures produced in the act of speaking appear to emanate from a point source located at a distance of just over half an inch behind the lips of the speaker and as a consequence their wavefront is spherical.

In Chapter 3 and Appendix 10 it is shown that the pressure gradient in a spherical wave is greater than the pressure gradient in a plane wave by a factor

$$\left\{1+\left(\frac{1}{rk}\right)^2\right\}^{\frac{1}{2}} = \left\{1+\left(\frac{c}{2\pi fr}\right)^2\right\}^{\frac{1}{2}}$$

and it is on this that the design of the close-speech anti-noise microphone depends. The factor indicates that if a pressure-gradient microphone is placed close to the mouth of a speaker, the bass tones in the speaker's voice will be over-emphasised, and the emphasis will increase as the microphone is brought closer to the speaker's lips.

10.2.1. DISCRIMINATION AGAINST DISTANT SOUNDS

If the microphone has been designed so that its axial response to a plane-wave source is independent of frequency, then the response when operating close to a source of spherical sound waves (the commentator's mouth) is greater at low frequencies than at high frequencies and the response/frequency curve could be represented by AB in Fig. 10.1. In Outside Broadcast programmes, noise may be present at the microphone but it is usually weaker than the commentator's speech and the line CD is assumed to represent the level and the response of the microphone to a distant source of plane waves, i.e., audience or crowd noise. The position of the microphone close to the commentator's mouth produces a speech quality with excessive emphasis on the low-frequency tones. The quality would be unsuitable for broadcasting, but can be corrected by passing the output of the microphone through an equaliser so designed that the response for close speech is independent of frequency. If the effective

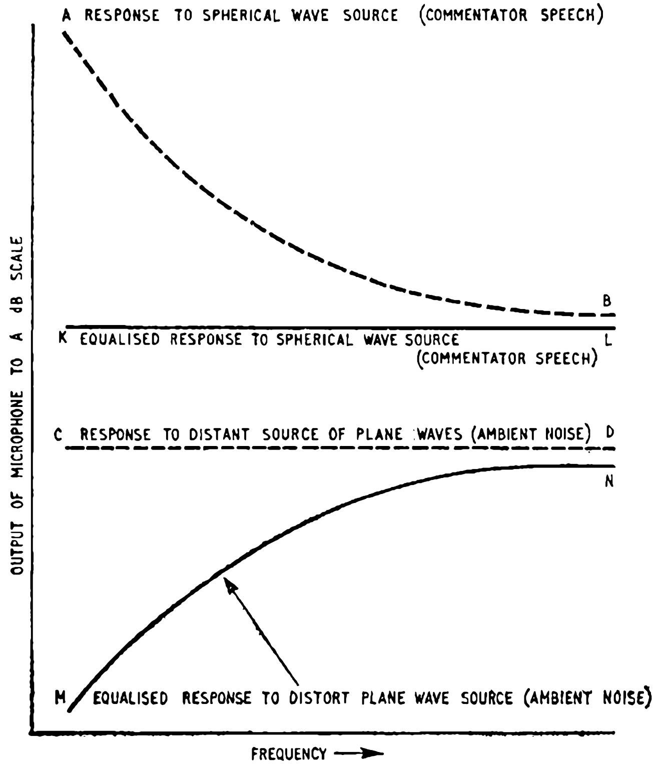


Fig. 10.1.—Diagrammatic representation showing the response and signal-to-ambient-noise separation before and after equalisation

distance of the microphone from the commentator's mouth is 2½ in., then in order to remove the excessive bass the equaliser would require to introduce a loss of about 18 dB at 100 c/s, 11 dB at 250 c/s, 5½ dB at 500 c/s and 2½ dB at 1,000 c/s. Since the background noise picked up by the microphone also passes through the equaliser, it is reduced by the same amount at the corresponding frequencies, resulting in considerable improvement in the speech-to-noise ratio. The response and level of the speech at the output of the equaliser are represented by the straight line in KL (Fig. 10.1) and the response and level of the background noise, by the curve MN.

The curves KL and MN show that the suppression of the background noise varies with frequency and is only fully effective over the lower half of the frequency range. At high frequencies the speech-to-noise ratio is less satisfactory, because the microphone and equaliser in combination are unable to discriminate against the high-frequency components of the distant noise. If however, the noise is random in direction, which is approximately what happens in practice, a further improvement of about 5 dB can be achieved throughout the frequency range because of the bi-directional characteristic of the microphone.

So far we have not taken into account the shielding provided by the head and body of the commentator. It is obvious that this will have its maximum effect at high frequency, especially if the noise source is situated behind the commentator. If however the noise is of random incidence then practical tests indicate[1] that the shielding effect is small and the improvement in speech-to-noise ratio is not likely to exceed 3 dB and that it is effective only at frequencies above 2,000 kc/s.

10.2.2. DISTANCE OF THE MICROPHONE FROM THE MOUTH

The closer the microphone is to the mouth, the greater is the discrimination against ambient sound, but speech quality and breathing noises set a limit to the minimum working distance. As the microphone is brought close to the mouth, breathing noises become apparent and objectionable. The noises produced result partly from the stream of air acting directly on the ribbon and partly from eddies and vortices set up by the air stream impinging on the edges of the microphone case or similar obstruction. The sounds produced by the air stream acting directly on the ribbon or moving element are usually of low pitch, but those from the eddies or vortices have a wide frequency spectrum and are capable also of causing air cavities within the microphone case to resonate at their natural frequencies.

Apart from the over-emphasis of the bass, speech quality can be affected in other ways. If a microphone having a bi-directional characteristic is brought very close to the mouth, the nose of the speaker may lie in or near the plane of minimum sensitivity. The nose plays an important part in the production of certain sounds: for instance, " m " and " n " sounds are made with the mouth closed either by closing the lips (m) or by closing the mouth by pressing the tongue against the top teeth (n). If the microphone is too close to the mouth, speech may still be intelligible but it will lose much of its natural quality because the nasal sounds will be heavily attenuated.

Furthermore, the position of the apparent source of sound behind the speaker's lips is not fixed, but varies with frequency and with certain vowel sounds. If the variations in the position of the apparent point source are comparable with the distance of the microphone from the lips, the response of the microphone is affected and the speech quality deteriorates. Taking into account the conflicting factors, it is usual to fix the minimum position of the microphone at a distance of about 2 to 2½ in. in front of the speaker's lips. In a practical design the minimum position is accurately controlled by means of spacing bars fitted to the mouth guard of the microphone.

10.3. Practical Commentator's Microphone

The BBC-S.T. & C. commentator's microphone was specifically designed for close speech and embodies the anti-noise principles already discussed. Its general appearance and main features are shown in Plates 10.1 and 10.2. The microphone assembly is enclosed in a perforated brass case lined with fine mesh phosphor-bronze gauze, the transformer being housed in the handle. The permanent magnet is U-shaped, with its curved portion towards the commentator's mouth. This arrangement permits the ribbon to be located at the required distance (2⅛ in.) from the lips of the speaker and yet results in a compact design. Moreover, the magnet in this position shields the ribbon from the direct air stream from the mouth but unfortunately it also acts as a partial obstruction to the high-frequency speech components and to a certain extent modifies the response of the microphone at the higher audio frequencies. The obstruction effect is reduced by making the magnet with openings in the limbs close to the pole-pieces but this inevitably reduces the cross-section of the limbs and affects the sensitivity by limiting the flux density in the gap. In spite of the flux limitations, the sensitivity of the microphone ($-$ 91 dB with reference to 1 V/dyn/cm^2) is adequate for even a whispered commentary.

The ribbon, which is made of aluminium foil, weighs 0·2 mg/cm^2 and has an average thickness of 0·6 μ.* Its fundamental resonance lies between 40 and 60 c/s and is adjusted by varying the tension. The factors which affect the dimension and the choice of the ribbon material have already been discussed in Chapter 8.

The microphone assembly is located in the case in a shock-free mounting, consisting of two rubber-covered pillars at the back and a pad of sponge rubber at the front.

To suppress or reduce the breath noise, shields are provided which extend over the top and the front of the microphone. For hygienic reasons they are made of plastic or of stainless-steel woven mesh, thus avoiding the use of materials which might corrode or become

* μ = 1 micron = 0·001 mm.

sodden with prolonged use. The shields also help to reduce condensation within the microphone case.

Commentators' microphones have often to be used in conjunction with television monitors or other electrical equipment and are sometimes subject to stray magnetic fields generated by the power supplies or by the scanning circuits. These fields affect the microphone transformer and may also induce hum voltages in the low-impedance leads. To reduce the induction pick-up to a minimum, a balanced wiring system is used to connect the ribbon to the microphone transformer which in turn is shielded from the stray fields by a Mumetal screening case $\frac{1}{32}$ in. thick (Section 12.6).

10.3.1. CONTROL CRITERION

Pressure-gradient microphones normally operate in plane-wave conditions and have mass-controlled moving systems, but since the BBC lip ribbon microphone is intended solely for close speech, its design is such that its response when close to a point source of sound tends to be independent of frequency. This is achieved by making the ribbon resistance-controlled rather than mass-controlled. The necessary resistance is provided by silk screens placed on either side of the ribbon and in close proximity to it. The screens serve a dual purpose, for they also act as windshields and permit the microphone to be used out-of-doors in stormy conditions without risk of damage to the ribbon.

10.3.2. EQUALISATION

Although the ribbon is substantially resistance-controlled, this of itself is not sufficient to produce a completely uniform response over the working range in close-speech conditions. Additional attenuation at low frequency is necessary for good speech quality, and an electrical correcting network in the form of a constant-impedance equaliser connected across the output of the microphone transformer provides the necessary attenuation. It has been found advantageous to be able to change the characteristics of the equaliser to deal with the different speech conditions which in practice may vary from a whisper to a shout. Quiet speech when reproduced at normal listening level sounds bass-heavy. On the other hand, loud or excited speech when reproduced at a lower level appears thin and lacking in bass. The characteristic of the equaliser is varied by means of a three-position switch. In the position " maximum bass " the response rises at low frequency (Fig. 10.2) and this setting of the equaliser is used for loud or excited speech. For normal speech the " medium bass " setting

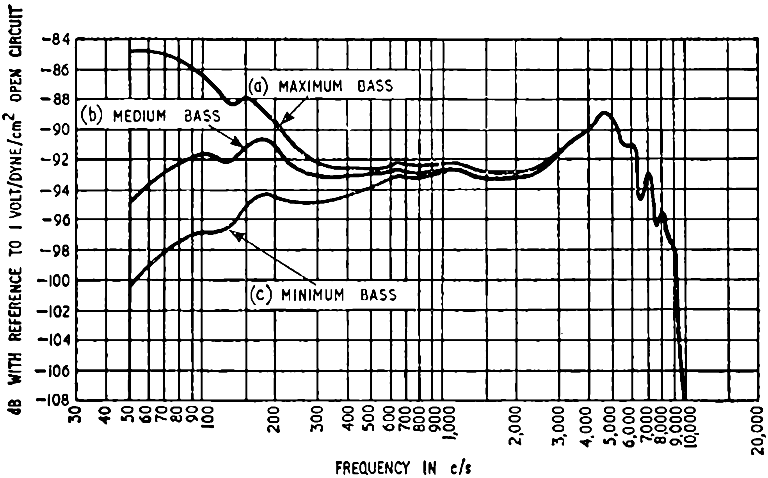


Fig. 10.2.—Commentator's microphone: effect of equaliser setting on frequency response

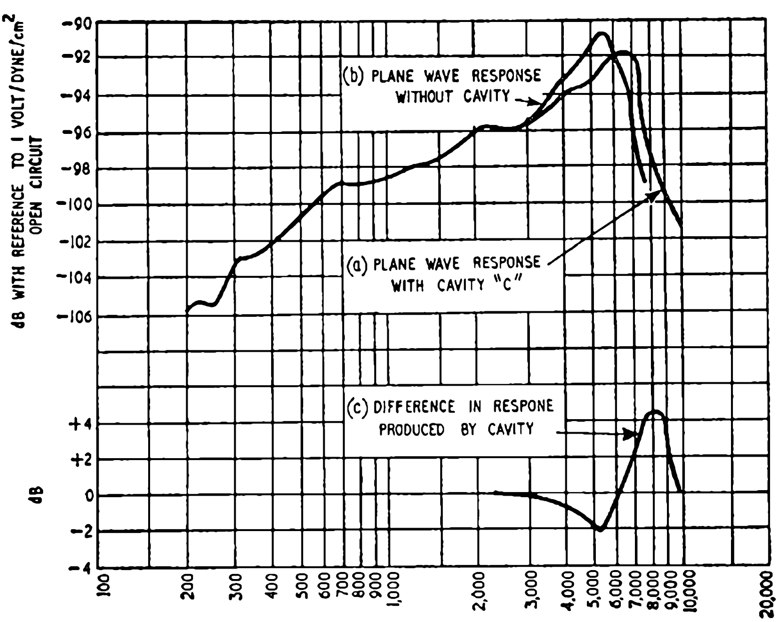


Fig. 10.3.—Effect of the compensating cavity on the frequency response with the equaliser set on medium bass

is appropriate and the " minimum bass " setting is more suitable for quiet or whispered speech.

10.3.3. COMPENSATING CAVITY

The response curves in Fig. 10.2 show a peak in the region of 5,000 c/s. The presence of the magnet in front of the ribbon is responsible for the peak, because the air enclosed in the cavity formed by the magnet and ribbon resonates at about 5,000 c/s and both the amplitude and the phase of the pressures on the front surface of the ribbon are affected. In order to compensate to some extent for the amplitude and phase changes on the front surface of the ribbon, the back of the ribbon is partially enclosed in a box-like compensating cavity, which can be seen in Plate 10.2.

In the absence of cavity, the axial plane-wave response of the microphone rises to a maximum at 5,000 c/s and falls steeply beyond 6,000 c/s. The compensating cavity restores to some extent the normal phase relationships on either side of the ribbon and extends the frequency response of the microphone by about 2 kc/s (Fig. 10.3). A slight rise in the spherical-wave response at 5 kc/s is still apparent but this is not a disadvantage, for it helps to compensate for certain of the characteristics associated with close-range speech.

10.4. Second-order Pressure-gradient Microphones

The discrimination provided by first-order pressure-gradient microphones is adequate for all normal noise conditions encountered in broadcasting, but there are occasions when a greater degree of discrimination would be advantageous, especially in noise fields of high intensity. Second-order pressure-gradient microphones have been produced which can discriminate against very high levels of ambient noise, but in their present state of development they are unlikely to be used in a broadcasting service, because of the poor quality associated with their rather restricted frequency range. Nevertheless it might be of interest to consider briefly some of their features and characteristics.

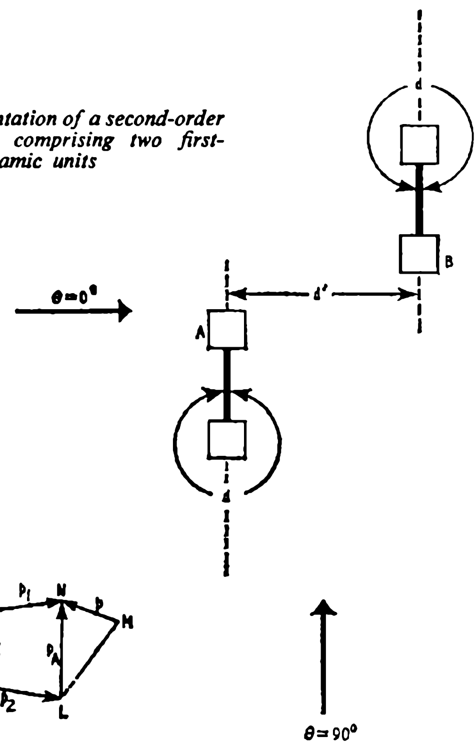
10.4.1. GENERAL

A second-order pressure-gradient microphone can be constructed by connecting in opposition two first-order pressure-gradient microphones. The microphones are arranged one behind the other, if possible on a common axis, so that the distance between them is small compared with the wavelength. The response of the combination will then be proportional to the difference in the pressure gradients at two closely spaced points in the acoustic field. Two bi-directional microphones could be used, or two phase-shift microphones with cardioid characteristics, and while either combination would be a true second-order pressure-gradient microphone, the directional characteristics of the two combinations would differ.

10.4.2. RESISTANCE-CONTROLLED BI-DIRECTIONAL COMBINATION

If a second-order pressure-gradient microphone is constructed by connecting two bi-directional ribbon microphones in opposition, the performance of the combination and some of its properties can be

Fig. 10.4.—Schematic representation of a second-order pressure-gradient microphone comprising two first-order electrodynamic units



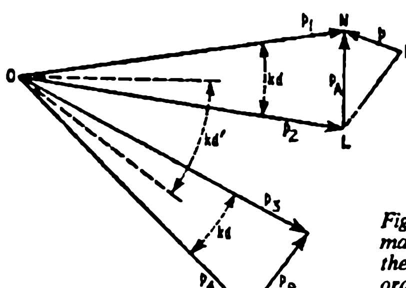


Fig. 10.5.—Vector diagram showing the magnitude and phase of the pressures on the moving elements in a twin-unit second-order pressure-gradient microphone

deduced from the vector relationships of the forces acting on the surface of the two ribbons. The simplest case to consider is that of two resistance-controlled ribbon microphones having identical path lengths d and situated so that the distance d' between their acoustic centres is small compared with the wavelength (Fig. 10.4). It will also be assumed that the source of sound is at an infinite distance from the combination, so that the microphones are actuated by plane waves.

In Fig. 10.5 p_1 and p_2 represent the pressures acting on the front and rear surfaces of ribbon A in both magnitude and phase. The resultant pressure on this ribbon is given by

$$p_A = 2p_1 \sin \frac{kd}{2}$$
$$\simeq p_1 kd$$
$$\text{if } d \ll \lambda$$

If the microphone is resistance-controlled, then p_A to a suitable scale will also represent the output V_A of microphone A in both magnitude and phase.

$$V_A = ap_A$$
$$= ap_1kd$$

where a is a constant of proportionality. Similarly, p_3 and p_4 are the pressures on the front and rear surfaces of ribbon B, the resultant pressure being represented by p_B.

Since the microphone is resistance-controlled, its electrical output V_B is proportional to and in phase with p_B: then

$$V_B = ap_B = ap_3kd$$

and since units A and B have identical path lengths d and operate in a plane-wave field,

$$V_A = V_B = ap_1kd \quad \text{for } p_1 = p_2 = p_3 = p_4$$

The voltages V_A and V_B are equal in amplitude but differ in phase by angle kd' where d' is the acoustic spacing between the units.

Since the units are connected in opposition, their combined voltage V is the vector difference between V_A and V_B. V can be evaluated by drawing LM equal and parallel to p_B. Since the angle MLN $= kd'$, MN represents to a suitable scale the output voltage V in both amplitude and phase.

$$\begin{aligned} MN &= 2p_A \sin \frac{kd'}{2} \\ &= p_A kd' \end{aligned}$$
if $d' \ll \lambda$

The combined voltage

$$V = a' p_A kd'$$

where a' is a constant

$$\begin{aligned} \therefore V &= a' (p_1 kd)\, kd' \quad \text{for } p_A \propto p_1 kd \\ &= a' p_1 k^2 dd' \\ &= 4a'\pi^2 p_1 \frac{f^2}{c^2} dd' \end{aligned}$$

The expression is accurate only at the low frequencies when the source of sound is located on the axis of the combination.

If the sound arrives at an angle to the axis, the effective path length d of the units and the effective spacing d' between them is altered.

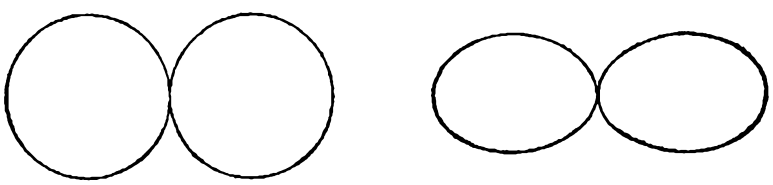


Fig. 10.6.—Directional characteristics of a first- and second-order pressure-gradient microphone

For an angle of incidence θ the effective path length is $d \cos \theta$ and the effective spacing is $d' \cos \theta$.

The voltage V_θ at angle θ is then

$$\begin{aligned} V_\theta &= 4a'\pi^2 p_1 \frac{f^2}{c^2} d \cos \theta \, d' \cos \theta \\ &= 4a'\pi^2 p_1 \frac{f^2}{c^2} d\, d' \cos^2 \theta \end{aligned}$$

The directional characteristic is a $\cos^2$ function and while the general shape is the basic figure-of-eight that is associated with pressure-gradient microphones, a comparison of the cosine and cosine^2 curves of Fig. 10.6 indicates that a second-order microphone is more directional than a first-order microphone.

10.4.3. UPPER LIMIT OF FREQUENCY

The expression for V_θ shows that the output and hence sensitivity is proportional to dd'. If the path length d of the individual units is fixed by design considerations, the voltage V is proportional to the spacing d' between the units. There is a limit to the size of d' which is determined by the frequency range the microphone is required to cover. At frequencies at which $d' \simeq \lambda$, the output is very small. When $d' = \lambda$, the units forming the combination are spaced a wavelength apart, in which case the pressure gradients at the individual units are equal and the output of the combination is zero.

At this frequency $d' = \lambda = c/f$, hence the upper frequency limit of the combination is:

$$f = \frac{c}{d'}$$

It would be difficult to design a microphone with an upper frequency limit of 10,000 c/s using two pressure-gradient ribbon microphones of normal construction, for the distance between the units would be relatively small, d' being less than $1\frac{1}{2}$ in. Furthermore, the output of the microphone is proportional to d' and with such close spacing, the sensitivity would be low. If the spacing is increased the sensitivity improves, but the design is cumbersome and the frequency coverage limited. It might be argued that the disadvantages of a limited frequency coverage and a bulky design are too high a price to pay for the cos^2 directional characteristic. This might be true of microphones used in normal working conditions, but second-order pressure-gradient microphones have the additional property of being able to discriminate against ambient noise to an even greater extent than first-order microphones. Practical microphones have been produced which will permit close speech to be transmitted in conditions where the ambient noise is on the threshold of pain.

10.4.4. ANTI-NOISE CHARACTERISTICS

The anti-noise properties of second-order microphones, like those of first-order types, are a maximum at low frequency and the ability to discriminate against ambient noise diminishes as the frequency increases. The vector diagrams in Fig. 10.7 (a) and (b) show that at low frequency the sensitivity is greater in a spherical-wave field than in a

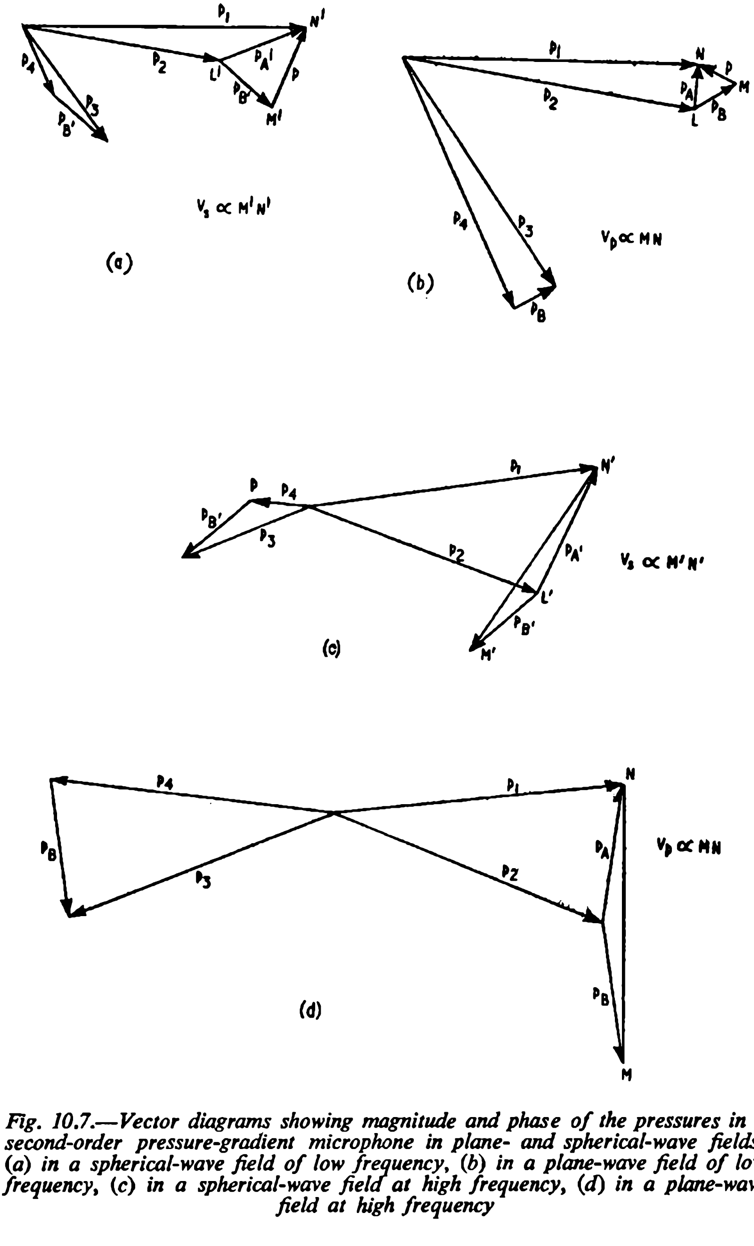


Fig. 10.7.—Vector diagrams showing magnitude and phase of the pressures in a second-order pressure-gradient microphone in plane- and spherical-wave fields: (a) in a spherical-wave field of low frequency, (b) in a plane-wave field of low frequency, (c) in a spherical-wave field at high frequency, (d) in a plane-wave field at high frequency

plane-wave field. In drawing the diagrams it has been assumed that the spherical-wave and plane-wave fields will produce identical pressures on the front surface of the first ribbon.

At low frequencies, as shown in Fig. 10.7 (a) and (b), the output of the microphone in the spherical-wave field is proportional to M′N′ and is greater than the output MN in a plane-wave field. As the frequency is raised, the difference between the outputs in the spherical-wave and plane-wave fields becomes smaller and at high frequencies, as shown in Fig. 10.7 (c) and (d), the output in the plane-wave field is greater than the output in the spherical-wave field, that is MN > M′N′. The discrimination is then negative and the anti-noise properties are lost.

10.4.5. RATIO OF R.M.S. VALUES OF OUTPUTS IN SPHERICAL-WAVE AND PLANE-WAVE FIELDS

The ratio of the r.m.s. value of the output in the spherical-wave field to that in the plane-wave field is a measure of the anti-noise properties of the microphone. The r.m.s. value of the output in the spherical-wave field is proportional to

$$\frac{p_{max}}{r^3\sqrt{2}}(4 + k^4r^4)^{\frac{1}{2}} \text{ (\textit{see} Appendix 10)}$$

and the r.m.s. value of the output in the plane-wave field is proportional to

$$\frac{p_{max}}{r\sqrt{2}}k^2$$

The axial discrimination is

$$\frac{\dfrac{p_{max}}{r^3\sqrt{2}}(4 + k^4r^4)^{\frac{1}{2}}}{\dfrac{p_{max}\,k^2}{r\sqrt{2}}} = \frac{1}{k^2r^2}(4 + k^4r^4)^{\frac{1}{2}}$$

$$= \left(1 + \frac{4}{k^4r^4}\right)^{\frac{1}{2}}$$

If the noise is random in direction, the $\cos^2$ directional characteristic of the microphone increases the discrimination still further and the average random discrimination is then

$$\sqrt{5}\left(1 + \frac{4}{k^4r^4}\right)^{\frac{1}{2}}$$

Comparing the axial discrimination of a first-order p.g. microphone $\left(1 + \frac{1}{k^2r^2}\right)^{\frac{1}{2}}$ with that of a second-order microphone $\left(1 + \frac{4}{k^4r^4}\right)^{\frac{1}{2}}$, it will be seen that both are functions of kr and it is possible to make a direct comparison of the anti-noise properties by plotting the axial and random discrimination of both types against kr. The results are

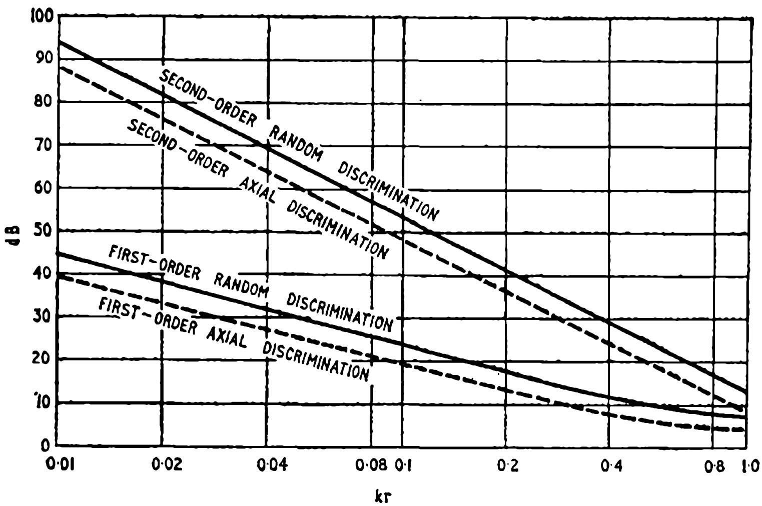


Fig. 10.8.—Theoretical discrimination of first- and second-order pressure-gradient microphone

shown graphically in Fig. 10.8. The discrimination of a first-order microphone $\propto 1/k$, that is, $\propto 1/f$ so the graph has a negative slope at low frequencies of 6 dB per octave, while the graph of the second-order type, whose discrimination $\propto 1/k^2$, has a negative slope of 12 dB per octave.

10.5. Higher-order Pressure-gradient Microphones

It is theoretically possible to construct higher-order gradient microphones and as the order of gradient increases, the discrimination increases and the negative slope of the curve increases in proportion. If the order of gradient is n, then the slope of the discrimination curve plotted against kr or against frequency is $6n$ dB per octave. Moreover as the order of the gradient becomes larger, the microphone becomes more and more directional, resulting in an improved discrimination against noise of random incidence. We have already seen that for random noise the discrimination for a first-order microphone is increased by a factor $\sqrt{3}$ and by $\sqrt{5}$ for one of the second-order.

If the order of gradient is n, then the increase in the directivity due to the improvement in the directional characteristic is $\sqrt{(2n + 1)}$. As n becomes large, the solid angle within which the microphone receives sound decreases and for large values of n the microphone is sensitive only on its axis.

Higher-order microphones can be obtained by connecting in opposition microphones or microphone combinations of a lower order. We have already seen that if the outputs of two suitably spaced first-order microphones are connected in opposition, a second-order microphone is produced. Two second-order combinations when connected differentially form a third-order microphone, but it is obvious that as the order of gradient increases, the microphone becomes awkward and bulky. There are other practical difficulties in combining microphones in this way, for if the discrimination is not to be adversely affected each of the first-order microphones forming the combination must have identical sensitivity and response and this applies not only to the magnitude of their outputs but also to the phase.

10.6. Single-diaphragm Second-order Pressure-gradient Microphones

Second-order or higher-order microphones, which employ two or more first-order units in combination, must of necessity be large and difficult to produce on a commercial scale, because of the need to select units which are correctly balanced. In this section, a second-order pressure-gradient microphone will be described which employs only a single diaphragm and a single transducer.[2] The advantages of this design are that it is compact and cheap and also eliminates the difficulties associated with the selection of twin units of equal sensitivity.

A second-order p.g. microphone could be defined as one whose output is dependent on the variations in pressure at four points in space, and in the twin-unit microphones just described four surfaces are required on which the pressures can act. The surfaces are the front and rear surfaces of ribbon A (Fig. 10.4) and the front and rear surfaces of ribbon B. It is possible to replace the four surfaces by four sound entrances, A, B, S and R, which admit acoustic pressures to different sides of a single diaphragm (Fig. 10.9). The entrances are so spaced that the diaphragm experiences a force proportional to the second order of the pressure gradient.

The two entrances A and B, which are opposite each other, allow the acoustic pressures to enter the upper chamber and so exert a force on the upper surface of the diaphragm while the entrances R and S, also opposite each other, admit the pressures on the lower chamber and so to the lower surface of the diaphragm.

The action can best be understood by imagining that only the sound entrance A is open and all others closed. The diaphragm is then

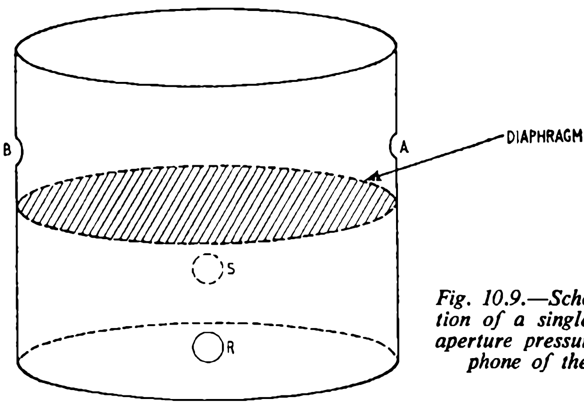


Fig. 10.9.—Schematic representation of a single-diaphragm four-aperture pressure-gradient microphone of the second order

subject to pressures on its upper surface only, and as a result is pressure-operated. If now aperture R is opened, acoustic pressures reach the lower surface of the diaphragm, and since apertures A and R are separated by a finite distance, the pressures exerted on either side of the diaphragm differ in phase. The diaphragm experiences a force proportional to the pressure gradient. Thus the apertures A and R acting together form one pressure-gradient unit, while the apertures S and B in combination form the other. By arranging for opposite entrances to admit pressures to the same surface of the diaphragm, the resultant force exerted is the difference of the forces obtained from the two pressure-gradient combinations; hence the effective force on the diaphragm is proportional to the second order of the pressure gradient.

10.6.1. THE RESULTANT FORCE ON THE DIAPHRAGM

An expression for the resultant force on the diaphragm can be obtained from the diagrams showing the vector relationship of the pressures acting on the diaphragm. It is assumed that the sound entrances are all in one plane and are symmetrically disposed on two axes at right angles, the distance between two apertures on a common axis being d, as in Fig. 10.10 (a). The force on the diaphragm is dependent on the angle of incidence and the simplest case is that of a plane wave whose direction of propagation is along one or other of the axes containing the sound entrances A and B, or R and S. If the direction of propagation is along the axis containing A and B, the sound apertures R and S are effectively at a distance $d/2$ from both A and B. If p_1 represents the pressure at A in magnitude and phase, the pressures p_2 and p_3 at R and S lag on p_1 by angle $kd/_2$. Aperture B is at an effective distance d from aperture A, hence the pressure p_4 at B lags on p_1 by kd radians (Fig. 10.10 (b)).

The resultant pressure on the diaphragm is the difference of the pressures acting on its two faces, that is,

$$\text{vector sum of } p_2 \text{ and } p_3 - \text{vector sum of } p_1 \text{ and } p_4.$$

The vector sum of p_2 and p_3 is represented by OM in Fig. 10.10 (c),

$$\therefore OM = p_2 + p_3 = 2p_1$$

since p_2 and p_3 are in phase and since the magnitudes of p_1, p_2, p_3, p_4 are equal. The vector sum of p_1 and p_4 is represented by ON in Fig. 10.10 (c);

$$\therefore ON = 2p_1 \cos \frac{kd}{2}$$

$$\text{The resultant pressure } p = 2p_1 - 2p_1 \cos \frac{kd}{2}$$

$$= 2p_1 \left(1 - \cos \frac{kd}{2}\right)$$

If the area of the diaphragm is A, the available force F is

$$F = 2Ap_1 \left(1 - \cos \frac{kd}{2}\right)$$

$$= 2Ap_1 \left(1 - \cos \frac{\pi d}{\lambda}\right)$$

Since the expression is a function of $1/\lambda$, the force varies with frequency but unlike that in other pressure-gradient microphones, the

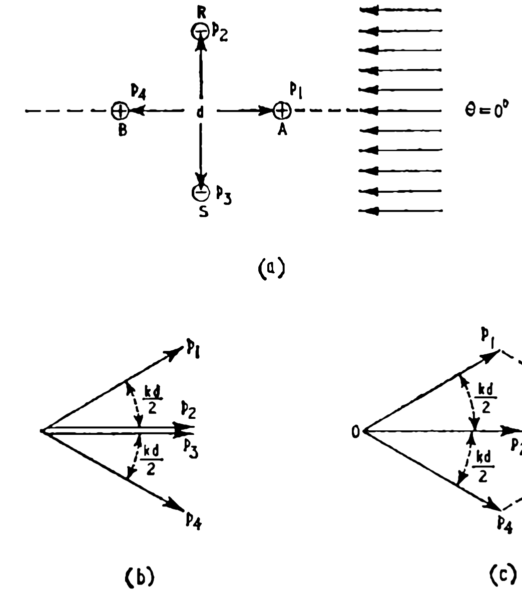


Fig. 10.10.—(a) Spacing and orientation of the sound entrances. (b) Vector diagrams showing the magnitude and phase of the resultant pressure on the diaphragm

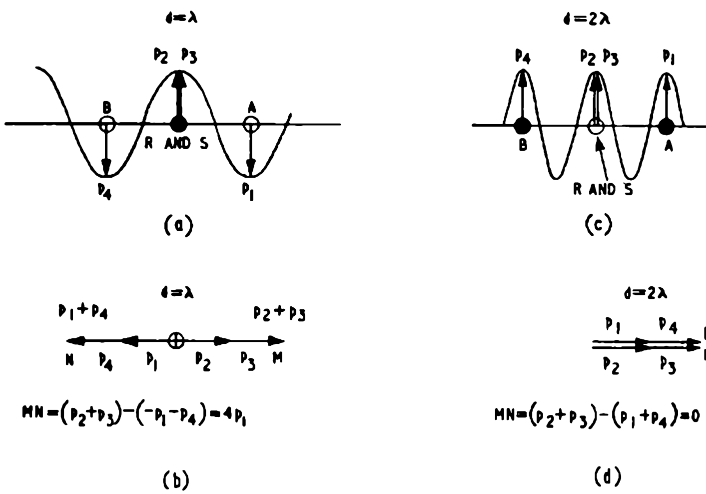


Fig. 10.11.—Wave and vector diagrams showing magnitude and phase of the pressures at d $= \lambda$ *and at* d $= 2\lambda$

force is not a maximum at the half-wavelength condition but reaches its maximum at the full-wavelength condition when $d = \lambda$. At this frequency,

$$F = 2Ap_1 (1 - \cos \pi)$$

Since $\cos \pi = -1$, $F = 4Ap_1$.

If the frequency is increased beyond the full-wave condition, the force diminishes and at $d = 2\lambda$ is zero.

$$F = 2Ap_1 (1 - \cos 2\pi)$$

Since $\cos 2\pi = 1$, $F = 0$.

The two conditions, when $d = \lambda$ and when $d = 2\lambda$, are illustrated in the wave and vector diagrams of Fig. 10.11. When $d = \lambda$, the pressures p_1 and p_4 at apertures A and B are in phase (Fig. 10.11 (a)) but are negative with respect to the pressures p_2 and p_3 at apertures R and S. The resultant pressure p is,

$$\begin{aligned} p &= p_2 + p_3 - (-p_1 - p_4) \\ &= 4p_1 \end{aligned}$$

When $d = 2\lambda$, the wave diagram in Fig. 10.11 (c) shows that the pressures p_1, p_2, p_3 and p_4 are all in phase and have the same magnitude. The resultant pressure p (Fig. 10.11 (d)) is,

$$\begin{aligned} p &= (p_2 + p_3) - (p_1 + p_4) \\ &= 0 \end{aligned}$$

10.6.2. RESPONSE

The expression for the available force is plotted as a function of $kd/2$ in Fig. 10.12 and as the wave and vector diagrams of Fig. 10.11 indicate, the force is a maximum when $kd/2 = \pi$ and is zero when $kd/2 = 2\pi$. Other maxima and minima are obtained as the value of $kd/2$ increases, the maxima occurring at 3π, 5π, 7π, etc., and the minima at 4π, 6π, 8π, etc. The shape of the curve beyond the first maximum is of academic interest only, for it is unlikely that a practical microphone would produce a useful output much beyond this maximum.

If the diaphragm is coupled to a resistance-controlled electrodynamic transducer, the shape of the response curve is similar to that of Fig. 10.12.

At low frequency the output increases at the rate of 12 dB per octave until a frequency is reached which makes $kd/2 \simeq 1$, or

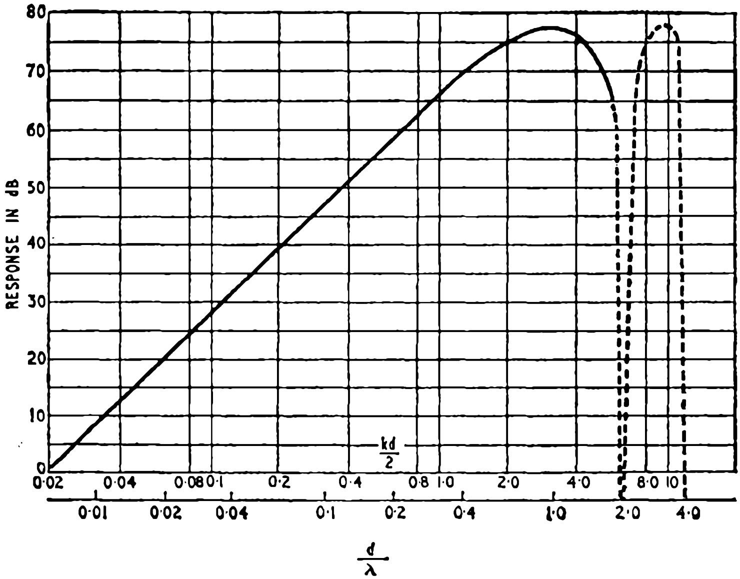


Fig. 10.12.—Theoretical response of a single-diaphragm four-aperture pressure-gradient microphone of the second order

$d/\lambda \simeq 1/\pi$; above this frequency the slope of the curve and hence the output gradually diminishes and the response reaches a maximum when $d/\lambda = 1$. Beyond this the output falls steeply.

10.6.3. DIRECTIONAL CHARACTERISTIC

If the sound arrives at an angle to the axis AB or to the axis RS, the output of the microphone falls because the resultant pressure on the diaphragm diminishes as a result of the change in the effective spacing between the sound entrances. When a sound is incident at an angle θ to the axis AB, the effective spacing between the entrances A and B is reduced from d to $d \cos \theta$ (Fig. 10.13). Hence the phase angle between the pressures p_1 and p_2 acting on the upper surface of the diaphragm is reduced to $kd \cos \theta$. In contrast, the effective spacing between the apertures R and S is increased from zero to $d \sin \theta$ and consequently p_2 and p_3 are now no longer in phase but differ by an angle $kd \sin \theta$.

As before, the resultant pressure on the diaphragm is the vector sum of the pressures acting on its two surfaces.

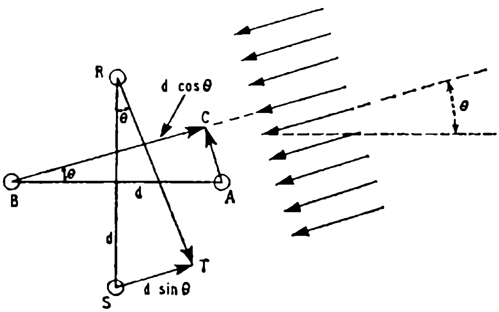
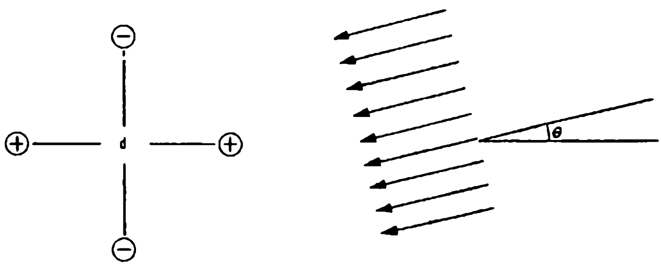
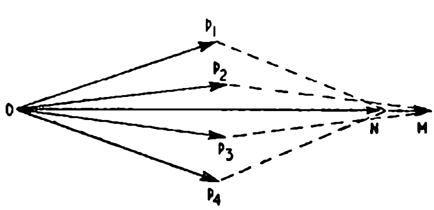


Fig. 10.13.—Showing how the effective spacing between the sound ports varies with the angle of incidence



(a)



(b)

Fig. 10.14.—(a) Spacing and orientation of the sound entrances for a wave at angle of incidence θ. (b) Vector diagrams showing the magnitude and phase of the pressures for an angle of incidence θ

The vector sum of the pressures p_2 and p_3 acting on the lower surface is represented in Fig. 10.14 (b) by OM since $p_1 \doteq p_2 = p_3 = p_4$.

$$OM = 2p_1 \cos\left(\frac{kd \sin \theta}{2}\right)$$

The vector sum of the pressures acting on the upper surface is represented by ON,

$$\therefore ON = 2p_1 \cos\left(\frac{kd \cos \theta}{2}\right)$$

The resultant pressure on the diaphragm is represented by MN,

$$\begin{aligned} \therefore MN &= OM - ON \\ &= 2p_1 \cos\left(\frac{kd \sin \theta}{2}\right) - 2p_1 \cos\left(\frac{kd \cos \theta}{2}\right) \\ &= 2p_1 \left\{\cos\left(\frac{kd \sin \theta}{2}\right) - \cos\left(\frac{kd \cos \theta}{2}\right)\right\} \end{aligned}$$

If values are assigned to $kd/2$ it will be found that at the low frequencies the pressure rises at the rate of 12 dB per octave and, as before, the rate of rise decreases as the frequency is raised.

The equation also shows that the effective pressure is dependent on the angle of incidence. By assigning values from 0° to 360° to θ in the expression, the directional characteristic is obtained. This is in the form of a double figure-of-eight or a four-leaved rose, as in Fig. 10.15. The microphone has two axes of maximum sensitivity at right angles to each other and two axes of minimum or zero sensitivity which are inclined at an angle of 45° to the axis of maximum sensitivity.

The physical rather than the mathematical explanation for the position of axes of zero sensitivity is apparent from Fig. 10.16. Since the wavefronts are at right angles to the direction of propagation, the acoustic pressures arrive simultaneously at the entrances A and R; consequently the phases and amplitudes of the pressures at these two entrances are equal and since they admit the sound pressures to opposite faces of the diaphragm, the pressures nullify each other. Similarly the amplitudes and phases of the pressures entering via

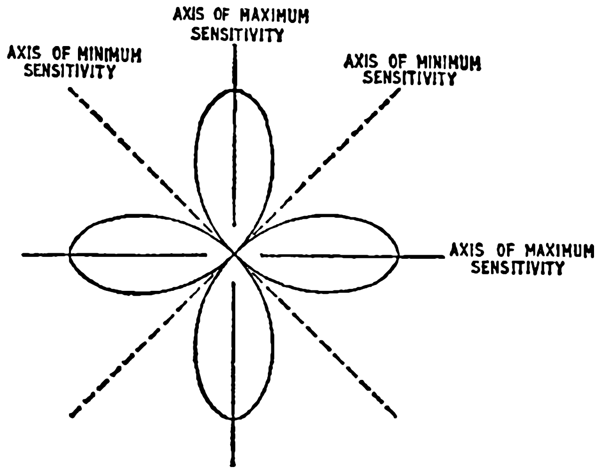


Fig. 10.15.—Directional characteristic of a single-diaphragm four-aperture second-order pressure-gradient microphone showing the axes of maximum amd minimum sensitivity

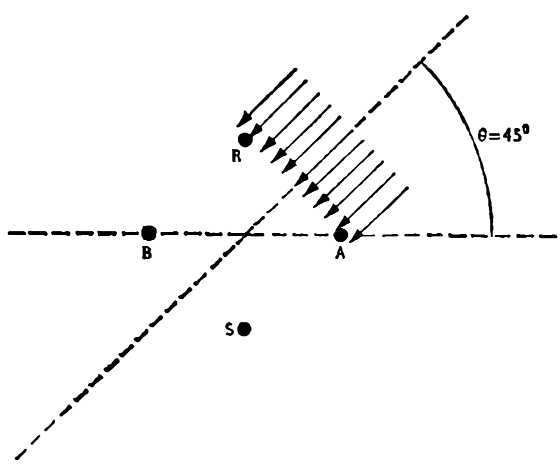


Fig. 10.16.—Position of the wavefront relative to the sound entrances A and R for angle of incidence $\theta = 45°$

B and S are equal, and since they too act on opposite sides of the diaphragm, cancellation is complete.

10.6.4. ANTI-NOISE PROPERTIES

This microphone, like other second-order pressure-gradient microphones, discriminates against ambient noise and favours close speech.

Fig. 10.17 (a) shows the conditions in a spherical-wave field when a point source of sound is located on the axis of maximum sensitivity AB, close to A. The pressure at A is p_1' and the pressure at B is p_4'. p_4' is smaller than p_1' because the pressures in the spherical-wave field are inversely proportional to the distance from the sound source. The resultant of the pressures p_1' and p_4' is represented in the diagram by ON'. At the apertures R and S the pressures are equal in amplitude and are represented by p_2' and p_3' and their resultant by OM'. As before the effective pressure on the diaphragm is,

$$ON' - OM' = M'N'$$

In Fig. 10.17 (b) the magnitude and phase of the effective pressure on the diaphragm in a plane-wave field of comparable amplitude are represented by NM. Comparison of the effective pressure NM on the diaphragm in the plane-wave field with the effective pressure $M'N'$ in the spherical-wave field shows that $M'N'$ is greater than NM for the low-frequency conditions depicted and consequently the microphone will discriminate against plane-wave ambient noise in favour of close speech.

As the frequency increases, the discrimination decreases and it is clear from Fig. 10.17 (c) and (d), representing the conditions at a frequency beyond the half-wavelength condition ($d/\lambda = 1/2$), that although the effective pressure $M'N'$ in the spherical-wave field has increased, the effective pressure NM in the plane-wave field has increased at a greater rate, and now $M'N'$ is smaller than NM. The microphone, in common with other pressure-gradient microphones, does not discriminate against high-frequency noise.

The frequency at which discrimination ceases depends on the distance of the microphone from the spherical-wave source and on the spacing d between the sound entrances. The closer the microphone is to the lips of the speaker, the greater is the discrimination. Practical considerations as well as speech quality may set a limit to the minimum distance. The smaller the spacing is between the sound entrances, the greater is the discrimination, but as d is reduced the

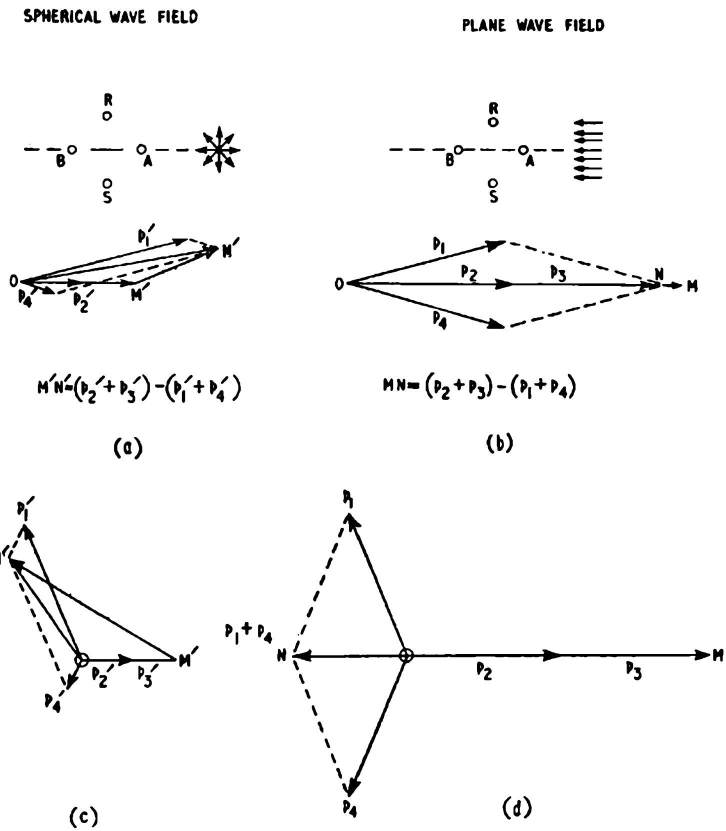


Fig. 10.17.—Vector diagrams showing the magnitude and phase of the pressure on the diaphragm of a four-aperture second-order pressure-gradient microphone (a) in a spherical-wave field of low frequency, (b) in a plane-wave field of low frequency, (c) in a spherical-wave field of high frequency, (d) in a plane-wave field of high frequency

sensitivity falls and the actual value of d in a practical design is a compromise between sensitivity and discrimination.

Beaverson and Wiggins[2] showed that at low frequencies where the microphone can be considered to act as a true differential device the axial discrimination is

$$\left(1+\frac{3}{k^2r^2}+\frac{9}{k^4r^4}\right)^{\frac{1}{2}} = \frac{1}{k^2r^2}\,(9+3k^2r^2+k^4r^4)^{\frac{1}{2}}$$

10.6.5. RANDOM DISCRIMINATION

The random discrimination of a second-order aperture microphone is superior to that of a first-order type, because the directional characteristic (Fig. 10.15) indicates that the microphone is insensitive to sound over four large solid angles, and as a consequence its random efficiency is only 18 per cent: that is, it picks up 18 per cent of sound power in a random field, in comparison to the 100 per cent picked up by an omni-directional microphone.

Random efficiency is a power measurement and is therefore proportional to the square of the pressure, while axial discrimination is a pressure ratio, i.e., the ratio of the pressure gradient in a spherical wave to the pressure gradient in a plane wave, and is therefore proportional to pressure. Random discrimination is obtained by multiplying the axial discrimination by a factor which is dependent on the random efficiency of the microphone.

The factor for a random efficiency of 18 per cent is

$$\left(\frac{100}{18}\right)^{\frac{1}{2}} = 2{\cdot}36$$

Hence the random discrimination is

$$\frac{2{\cdot}36}{k^2r^2}\,(9+3k^2r^2+k^4r^4)$$

Assigning values to kr and plotting the results shows that the theoretical random discrimination (curve A, Fig. 10.18) has the same general shape as that of other second-order microphones, that is it has a negative slope of 12 dB per octave for values of kr less than 1. Above this the slope decreases, and for values of $kr = 5$ or more, the curve is almost horizontal and the discrimination is constant at

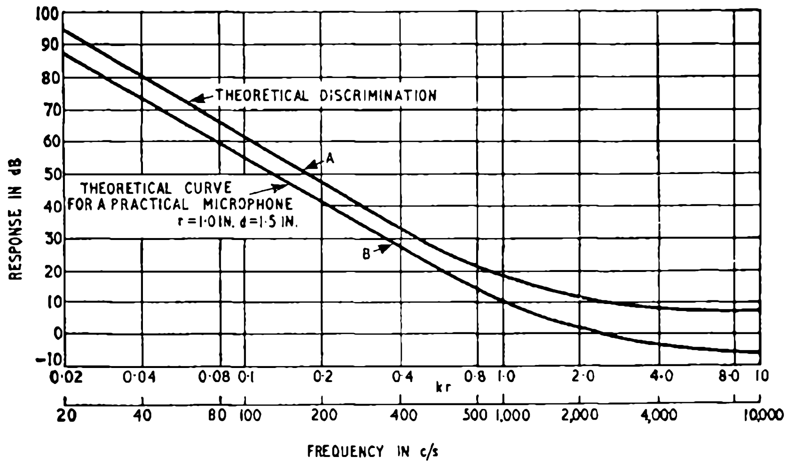


Fig. 10.18.—Theoretical and practical discrimination curves of a four-aperture second-order pressure-gradient microphone (after Beaverson and Wiggins)

7·4 dB. Thus the discrimination above $kr = 5$ is due largely to the directional characteristic of the microphone, i.e.,

$$\begin{aligned} \text{dB} &= 20 \log \left(\frac{100}{18}\right) \\ &= 7{\cdot}4 \end{aligned}$$

Curve A was obtained by a mathematical process which assumed that the spacing between the sound entrances is infinitely small. This is not a practical condition and the curve should be compared with the theoretical curve for a practical microphone, with a spacing between the sound entrances of $d = 1{\cdot}5$ in., and operating at a distance of 1 in. from a point source of sound (curve B, Fig. 10.18). It will be seen that the effect of the spacing on the slope is negligible at low frequency; furthermore, the actual value of the discrimination is within 7 dB of the theoretical maximum. An even closer approximation to the theoretical maximum could be obtained if the spacing were reduced or if the microphone were brought closer to the sound source.

Curve B shows that for the values of d and r selected, the discrimination is zero at a frequency of about 2,500 c/s. Above this frequency the discrimination is negative. This is the condition shown in the vector diagrams of Fig. 10.17 (c) and (d) where the effective force on the diaphragm in the plane-wave field (diagram (d)) is greater than in the spherical-wave field (diagram (c)).

10.6.6. EQUIVALENT CIRCUIT

The distance of the microphone from the sound source and the spacing between the sound entrances are not the only factors affecting sensitivity and discrimination. The equivalent circuit of the microphone (Fig. 10.19) will make the problem more obvious.

If the transducer is of the electromagnetic type, the diaphragm could be either resistance-controlled or mass-controlled. A mass-controlled system would necessitate a very flexible suspension for the diaphragm and coil unit in order to keep the natural frequency of the system as low as possible. With such a flexible mounting the chances of accidental contact between the coil and pole-pieces are greatly increased, and if a generous gap clearance is provided to reduce the possibility of contact, the sensitivity of the microphone is impaired because of the reduction in the gap flux. Moreover a mass-controlled system with its low natural frequency of resonance is susceptible to mechanical shocks and is far from ideal in a hand-held microphone.

Resistance control of the diaphragm system is more likely to give the best results, in which case m_d, C_{md} and R_{md} of Fig. 10.19 (a) can be replaced in Fig. 10.19 (b) by R_{md} which represents the mechanical

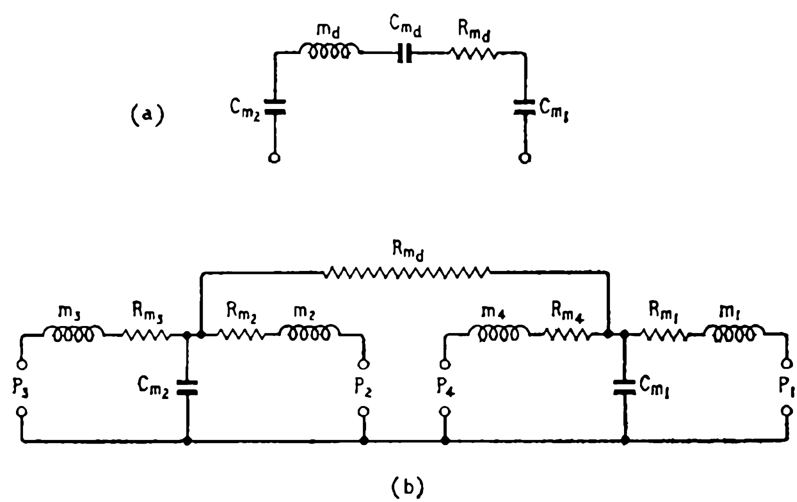


Fig. 10.19.—(a) Equivalent circuit of diaphragm and air cavities; (b) equivalent circuit of four-aperture microphone having a resistance-controlled diaphragm

resistance of the diaphragm and coil system. As before, the compliance of the upper cavity is represented by C_{m_1} and the lower by C_{m_2}. The acoustic pressures p_1, p_2, p_3 and p_4 act on the air in the cavities via the sound apertures; these themselves possess both mass and resistance, m_1, m_2, m_3 and m_4, and R_{m_1}, R_{m_2}, R_{m_3} and R_{m_4} respectively.

In developing the equations for response and sensitivity, it was assumed that the phase of the pressures on the diaphragm would be determined solely by the effective spacing between the sound entrances. The required phase relationship between the pressures is maintained if $C_{m_1} = C_{m_2}$ and if the phase shifts introduced by the aperture impedances m_1, R_{m_1}; m_2, R_{m_2}; m_3, R_{m_3}; m_4, R_{m_4} are all equal.

The sensitivity of the microphone is also affected by the magnitude of C_{m_1} and C_{m_2}, because, in conjunction with the aperture impedances, they form part of the L-section low-pass filter which determines the amplitude as well as the phase of the pressures applied to the diaphragm. The critical frequency of the filter must be high, because the amplitudes of the pressures in the air cavities fall at the rate of 12 dB per octave at frequencies beyond the critical frequency of the filter, and affect the response of the microphone accordingly. If a good high-frequency response is desired, the mass reactance of the first impedances must be small and the compliance reactance of the air cavities must be large. For good discrimination, the following balance conditions must also be satisfied:

$$R_{m_1} = R_{m_2} = R_{m_3} = R_{m_4}$$
$$m_1 = m_2 = m_3 = m_4$$
$$C_{m_1} = C_{m_2}$$

The proper balancing of the acoustic impedances is one of the major obstacles to the production of gradient microphones of a higher order than the second, and at this stage of microphone development improved discrimination is more likely to be achieved by reducing random pick-up. This would have the added advantage of improving the signal-to-ambient-noise ratio at high frequencies where discrimination is usually poor or even negative.

REFERENCES

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2. Beaverson, W. A. and Wiggins, A. M., " A second-order gradient noise-cancelling microphone using a single diaphragm ", *J. Acoust. Soc. Am., 22*, No. 5 (1950).

11

Directional Microphones

11.1. Purpose of Directional Microphones

In the previous chapter we examined and discussed the method of operation of microphones used for the special purpose of picking up speech at close range in noisy surroundings, and we showed that a high degree of discrimination against noise was possible because of the different shapes of the wavefronts associated with close speech and distant noise. This chapter is concerned with microphones that are used to pick up sound at a greater distance than normal, and to reject signals or noise from other sources. Since the desired signal is at a considerable distance from the microphone, the wavefronts of both the programme source and the unwanted noise are plane and there is no way of distinguishing between them except by means of position. The microphone must therefore be highly directional, picking up wanted sounds only on the axis of maximum sensitivity and rejecting unwanted sounds or noise coming from other directions.

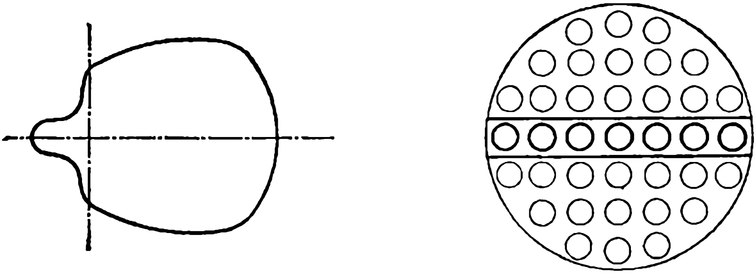
Microphones of this type are classed as directional microphones and are used on occasions in television or sound broadcasting, to make audible to the listeners distant sounds associated with the programme in order to enhance presentation. They have been employed in the theatre to pick up the rhythm of tap-dancers and they have also been used, in conjunction with other, more conventional microphones, for speech or questions from individual members of a large audience difficult to reach with a microphone on a trailing lead. But it is in the field of outside broadcasting that they are most useful, especially in the type of programme where the action is fast and where it is difficult to bring a microphone of the normal type close enough to pick up the appropriate sounds or sound effects. A typical illustration of their use in television is to pick up the impact noises in ball games, and for this purpose they are sometimes mounted on the camera following the play. The telephoto lens of the camera can bring the viewer a close-up of the scene and this usually requires that the sound picture should coincide with the vision picture if realism is to be preserved; this is achieved by using a directional microphone in conjunction with a long-focus or zoom lens.

Directional microphones are divided into two main groups. First, there are microphones whose directivity depends on their dimensions being larger than the wavelengths of the sounds in their operating range. Because of their size, these microphones are known as *dimensional* or *wave-type* microphones. Secondly, there are gradient microphones, whose directivity depends on the difference of the pressure gradients at two or more points in the sound field. In general, gradient microphones are smaller than the dimensional types.

11.2. Dimensional Microphones

The principle of operation of dimensional microphones has been discussed in Chapter 5 which showed that when the dimensions of a pressure-operated microphone are large compared with the wavelength of sound, the directional characteristic ceases to be independent of the angle of incidence and becomes more and more directional as the frequency increases.

If a large microphone could be constructed with a diaphragm diameter D of about 6 ft, the directional characteristic at 200 c/s, when $D \simeq \lambda$, would be similar in shape to Fig. 11.1 (a) and the microphone would be highly directional over the major portion of the audio-frequency range. Obviously, a microphone of this size is



(a) (b)

Fig. 11.1.—(a) Directional characteristic of pressure-operated microphone at high frequency when $D/\lambda = 1$. *(b) A series- or parallel-connected group of pressure units of equal sensitivity uniformly spaced over a large circular area has a directional characteristic similar to (a) but at a much lower frequency*

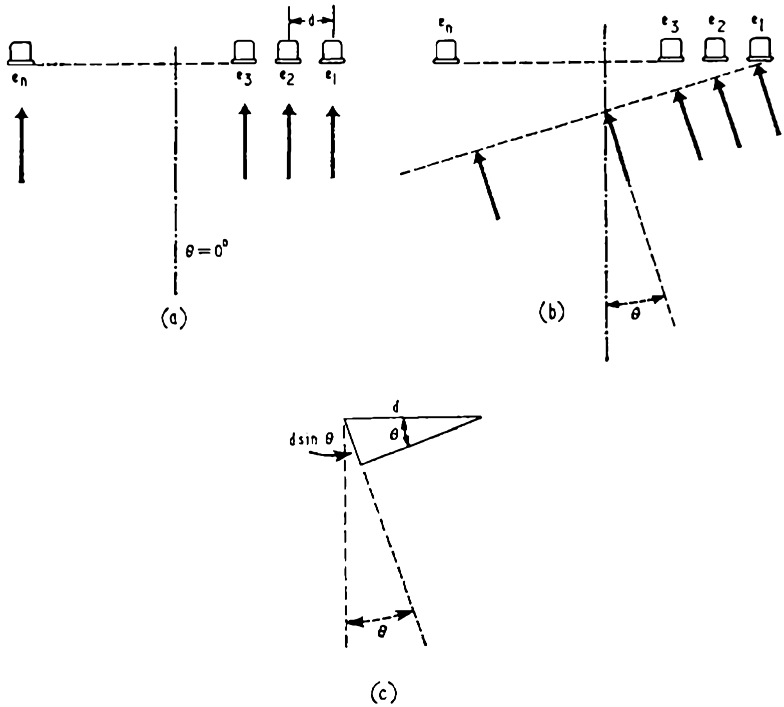


Fig. 11.2.—(a) Plane wave incident at $\theta = 0°$ on a horizontal array of microphones. (b) Plane wave incident at $\theta°$ on a horizontal array of microphones. (c) The effective spacing d *depends on the angle of incidence*

impracticable but instead a large number of pressure-sensitive microphones of conventional design could be connected in series and uniformly spaced over the area occupied by the 6-ft diaphragm, as in Fig. 11.1 (b). Provided the microphones had equal response and sensitivity, the directional characteristic of the group would be very similar to that of the large diaphragms. The group would be directional in both the horizontal and the vertical planes, since the combination is symmetrical about either axis. Most programme sources and most noises (aircraft noises excepted) originate in the horizontal plane, and directivity in the vertical sense is unnecessary. An appreciable saving in cost can be made if directivity is confined to the horizontal plane, by using a single row of microphones taken from the central portion of the diaphragm area (Fig. 11.1 (b)). This row of microphones, or the array as it will now be called, is directional in the plane containing the main dimension and if it is mounted with this dimension parallel to the ground, it is directional in the horizontal plane.

11.2.1. DISTRIBUTED ARRAY

The distributed array of Fig. 11.2 (a) consists of a row of n pressure-sensitive microphones connected in series and equally spaced over a distance. When sound is incident on the array at $\theta = 0°$, the wavefronts arrive at the same instant at all the units and if they have identical response and sensitivity, the voltage contributions e from the individual units will be in phase.

Hence the output of the array E at $\theta = 0°$ is

$$E = ne$$

If the wave arrives at an angle other than $\theta = 0°$, the wavefronts reach different units at different times (Fig. 11.2 (b)), and voltage contributions from the individual units vary in phase progressively throughout the array. The phase angle ϕ between any two adjacent units depends on the effective spacing, on the frequency and on the

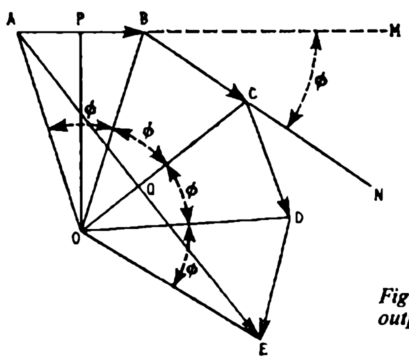


Fig. 11.3.—Vector summation of the output of the elements in an array (vector polygon)

angle of incidence. If the spacing between two units in the array is d, the effective spacing is $d \sin \theta$, as shown in Fig. 11.2 (c), and the phase angle $\phi = kd \sin \theta$.

11.2.2. OUTPUT OF THE ARRAY AT ANGLE θ

The output E_θ of the array at angle θ is the vector sum of the outputs of the individual units. In Fig. 11.3 the voltages e_1, e_2, e_3, etc. of the individual units are represented by AB, BC, CD, etc. BC lags on AB by angle ϕ and CD lags on BC by angle ϕ, etc.

$$M\hat{B}N = \phi$$

$$O\hat{B}A + O\hat{B}C + N\hat{B}M = 180°$$

$$O\hat{B}A + O\hat{A}B + A\hat{O}B = 180°$$

$$\text{Since } O\hat{A}B = O\hat{B}C$$
$$A\hat{O}B = \phi$$

OP is perpendicular to AB,

$$\therefore AB = 2OA \sin \phi/2$$

The vector sum of AB, BC, CD, DE is represented by AE, and since OQ is perpendicular to AE,

$$AE = 2OA \sin 4\phi/2$$

From the above expression it follows that the output voltage E_θ of an array of n units is

$$E_\theta = 2OA \sin n\phi/2$$

Now, $AP = OA \sin \phi/2$,

$$\therefore OA = \frac{AP}{\sin \phi/2}.$$

Substituting this value for OA in the equation for E_θ, we have

$$E_\theta = 2AP \frac{\sin n\phi/2}{\sin \phi/2}$$

But $2AP = BC = e$,

$$\therefore E_\theta = e \frac{\sin n\phi/2}{\sin \phi/2}$$

11.2.3. DIRECTIONAL CHARACTERISTIC

The phase angle ϕ is dependent on the length l of the array, the spacing d between the pressure elements, and the frequency.

$$\phi = kd \sin \theta$$

and since $k = \frac{2\pi}{\lambda}$ and $d = \frac{l}{(n-1)}$ (*see* Fig. 11.2(c)),

$$\phi = \frac{2\pi l}{\lambda(n-1)} \sin \theta$$

At low or very low frequencies λ may be large compared with l and consequently ϕ will be small: then

$$\frac{\sin n\phi/2}{\sin \phi/2} \simeq n$$

Hence the output of the array at angle θ is

$$E_\theta = ne$$

As is to be expected, the array is non-directional at low frequencies, or when l is small compared with λ.

At a higher frequency l may be comparable with, or larger than λ; in this case

$$\frac{2\pi l}{\lambda\,(n-1)}$$

s large, hence the actual value of the phase angle

$$\phi = \frac{2\pi l}{\lambda\,(n-1)} \sin \theta$$

is dependent on the value of $\sin \theta$, i.e., on the angle of incidence. The array is therefore directional when l/λ is large and the output for any particular angle of incidence can be obtained from the expression for E_θ.

11.2.4. DIRECTIONAL CHARACTERISTIC OF A 10-ELEMENT ARRAY

The directional characteristic of a 10-element array for various values of the ratio l/λ is shown in Fig. 11.4. As indicated in Fig. 11.4

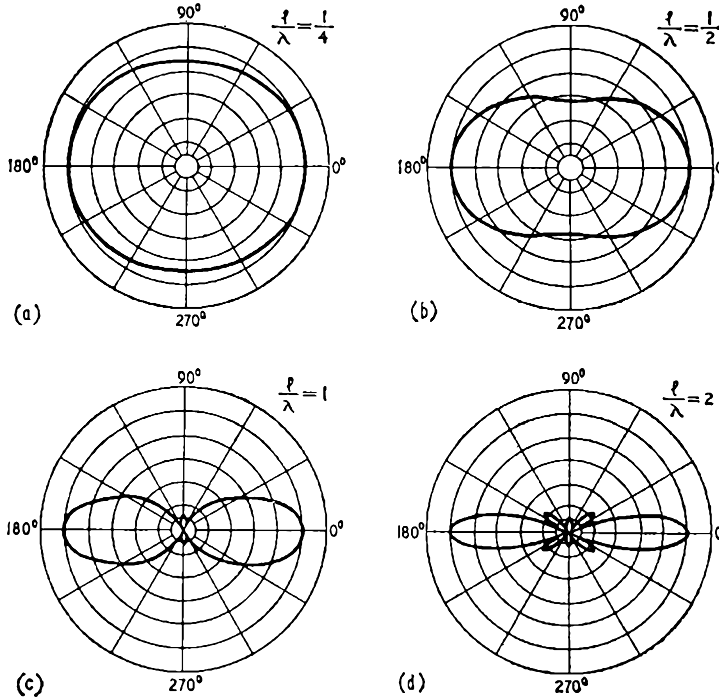


Fig. 11.4.—Polar response of the array of Fig. 11.2 (a) for various values of the ratio l/λ. (a) $l/\lambda = \frac{1}{4}$; (b) $l/\lambda = \frac{1}{2}$; (c) $l/\lambda = 1$; (d) $l/\lambda = 2$

(a), $l/\lambda = 1/4$; the array is omni-directional when the ratio l/λ is small.

As the frequency increases the array becomes directional and when $l/\lambda = 1$, it is highly directional, as shown in Fig. 11.4 (c), with maximum sensitivity at $\theta = 0°$ and $\theta = 180°$. The directional characteristic has also two axes of minimum sensitivity where the output is zero.

The angles of incidence for zero output can be obtained from the expression for E_θ.

$$E_\theta = e\frac{\sin n\phi/2}{\sin \phi/2}$$

For a 10-element array operating at $l/\lambda = 1$, zero output occurs when

$$\frac{10\phi}{2} = \pi$$

$$\text{i.e., } \phi = \frac{\pi}{5}$$

$$\text{Now } \phi = \frac{2\pi}{\lambda} \cdot \frac{l}{(n-1)} \sin\theta,$$

and since $l = \lambda$ and $(n - 1) = 9$

$$\phi = \frac{2\pi}{9} \sin\theta$$

$$= \frac{\pi}{5}$$

$$\sin\theta = \frac{9}{10} \quad \text{or} \quad \theta = 64{\cdot}2^\circ$$

The other angles of incidence giving zero output are:

$$\theta = 116^\circ,\ \theta = 244^\circ \text{ and } \theta = 296^\circ$$

As the frequency increases, the array becomes more and more directional and more side-lobes appear in the characteristic. When $l/\lambda = 2$, as in Fig. 11.4 (d), the directivity characteristic has six side lobes and four axes of zero sensitivity.

11.2.5. THE VECTOR POLYGON

The vector diagrams (Fig. 11.5) for a 10-element array when $l/\lambda = 2$, show how the shape of the vector polygon changes with angle of incidence. When θ is small, the phase angle is small and the e.m.f. vectors e_1, e_2, e_3, etc., lie along an arc of a circle whose radius is large. Hence E_θ is approximately equal to E, as in Fig. 11.5 (a) and (b).

As the angle of incidence increases, the radius of the circle containing the vector polygon decreases and the polygon tends to close (Fig. 11.5 (c)). The polygon closes when $\theta \simeq 26{\cdot}8^\circ$ and the output of the array is zero, as shown in Fig. 11.5 (d). For an angle greater than $26{\cdot}8^\circ$ the radius of the containing circle continues to decrease, and certain of the e.m.f. vectors overlap. Fig. 11.5 (e) shows the overlap and the second maximum associated with the minor lobe in the first quadrant of the directional characteristic (Fig. 11.4 (d)).

When $\theta \simeq 64{\cdot}2°$, the vector polygon closes twice and the output of the array is again zero. This condition is represented by Fig. 11.5 (f). The polygon has apparently 5 sides instead of 10. This is the result of the double closure, for each side now comprises two concurrent vectors that is, e_6 lies on top of e_1 and e_7 lies on top of e_2, etc.

For angles of incidence greater than 64·2° and smaller than 90°, the radius of the containing circle decreases as θ increases, resulting in an increased overlap of the vectors which produces the output in the directional characteristic associated with the lobe between 64·4° and 90°.

Output at $\theta = 90°$

At 90° the output of the array (when $l = \lambda$ or 2λ or 3λ, etc.) is $1/n$ of the output on the axis. This can be deduced from the ratio

$$R_\theta = \frac{\text{output of array at angle } \theta}{\text{output of the array at } \theta = 0°}$$

$$= \frac{E_\theta}{E_{\theta=0°}}$$

$$= \frac{e\dfrac{\sin n\phi/2}{\sin \phi/2}}{ne}$$

$$= \frac{1}{n}\left(\frac{\sin n\phi/2}{\sin \phi/2}\right)$$

If l/λ is an integer and $\theta = 90°$ so that $\sin\theta = 1$, then

$$\sin\frac{\phi}{2} = \sin\frac{\pi l}{\lambda(n-1)} \quad \text{(See Section 11.2.3)}$$

$$= \sin\frac{a'\pi}{(n-1)}$$

where a' is an integer.

$$\therefore \sin n\frac{\phi}{2} = \sin n\frac{a'\pi}{(n-1)}$$

$$= \sin\left(a'\pi + \frac{a'\pi}{(n-1)}\right)$$

$$\text{Hence } \frac{\sin n\phi/2}{\sin \phi/2} = \frac{\sin(a'\pi + a'\pi/(n-1))}{\sin a'\pi/(n-1)}$$

$$= \pm 1$$

depending on the value of the integer a'.

$$\text{Consequently } R_{\theta=90^\circ} = \frac{1}{n}\left\{\frac{\sin(a'\pi + a'\pi/(n-1))}{\sin a'\pi/(n-1)}\right\}$$

$$= \pm\frac{1}{n}$$

The side-lobes at $\theta = 90°$ are $1/n$ of those on the axis of maximum sensitivity. By using a large number of pressure elements, the size of the side-lobes can be reduced, but this makes for a bulky and costly instrument. Moreover, the directional characteristic of the array is equally sensitive at both front and rear and this added complication may make it difficult to find a suitable site for the

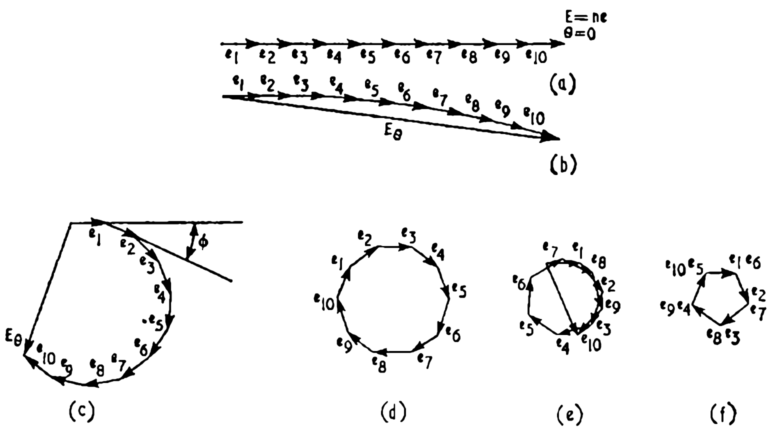


Fig. 11.5.—Vector diagram for a 10-element array at $l/\lambda = 2$, showing how the shape of the vector polygon varies with the angle of incidence

microphone in certain types of outside broadcasts. For this and other reasons, microphone arrays of this type are seldom met with in practice but have been discussed at length here because the basic theories apply with only minor modifications to the more practical types of directional microphones.

11.2.6. NON-DIRECTIONAL PROPERTY

Before leaving this subject we must stress that while an array of the type just described is highly directional in the plane containing the length dimension l, it is non-directional in the plane at right angles to l. Occasionally in sound broadcasting studios, microphone arrays consisting of three or more pressure elements mounted vertically have been employed. This arrangement has the practical property that while it is omni-directional in the horizontal plane, it is directional or highly directional in the vertical sense and picks up less reverberant sound than would a normal omni-directional unit. Thus artists or instruments can be positioned at a distance from the array without the reverberant sounds impairing or destroying diction. Alternatively, the array could be used in somewhat over-lively acoustic conditions where the programme material might make the use of bi-directional or uni-directional microphones impractical. If such an array is used in a studio it should be remembered that the directional characteristic is not independent of frequency and as a result the technical quality of the programme must be affected.

11.3. Multi-tube Directional Microphones

We have seen that microphone arrays used as directional devices are likely to be expensive because of the large number of pressure units required to produce a suitable directional characteristic and also because the bi-directional nature of the characteristic may prove something of a disadvantage in certain circumstances or on certain sites.

A highly directional microphone employing a single pressure-operated unit was developed by the Western Electric Company and was described in the Journal of the Acoustical Society of America by W. P. Mason and R. N. Marshall.[1] The instrument consisted of a group of 50 thin-walled aluminium tubes, $\frac{3}{8}$ in. in diameter, but of different length, so arranged that they could easily be coupled to a standard pressure-operated microphone of the electrodynamic type (Fig. 11.6).

The impedance of a single tube is purely resistive at resonance and it was claimed that since a large number of tubes of different lengths

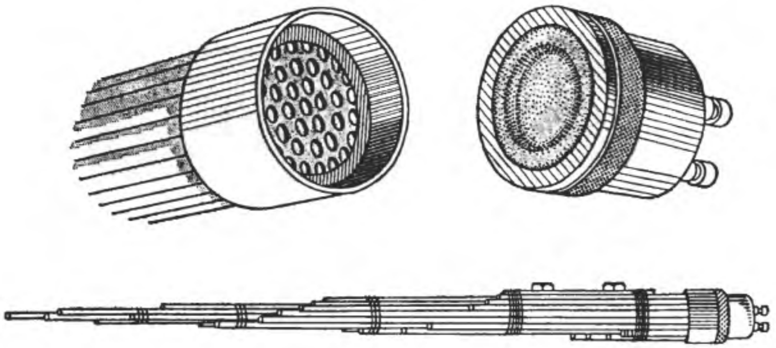


Fig. 11.6.—Multi-tube directional microphone developed by the Western Electric Company

was employed, the multiple resonances produced by the group occurred at intervals so close together that the combination acted as a constant acoustic resistance over a wide frequency range and presented to the diaphragm of the microphone the required termination for minimum distortion.

11.3.1. PRINCIPLE OF OPERATION

The principle of operation of the multi-tube microphone can best be explained by reference to the simplified sketches of Fig. 11.7 (a) and (b). Only four tubes decreasing in equal steps by d are shown, rather than the 50 tubes employed in the practical microphone.

Assuming the velocity of sound within the tubes is the same as that in free space,* then all sounds approaching from the front of the microphone on the $\theta = 0°$ axis have the same total distance to travel, irrespective of their point of entry, to reach the diaphragm. Thus on the $\theta = 0°$ axis the contributions from the various tubes all arrive at the diaphragm in the same phase.

If sound is incident at an angle θ to the axis, the pressure contributions from the various tubes differ in phase on reaching the diaphragm, because of the different length of the path traversed. For example, if a plane wave is incident at $\theta = 90°$, the acoustic pressures enter the tube openings at the same instant (provided that the width of the group of tubes is small compared with the wavelength

* Daniels[2] has shown that velocity of sound in a tube c_t will approximate to the velocity c of sound in free space if the product of the radius r of the tube and the square root of the frequency f transmitted is large enough, i.e., if $rf^{\frac{1}{2}} = 0{\cdot}6$, then $c_t = 0{\cdot}8c$ or if $rf^{\frac{1}{2}} = 2{\cdot}0$, then $c_t = 0{\cdot}93c$.

of the sound), and since adjacent tubes differ in length by d, the phase angle ϕ between the pressure contributions from adjacent tubes is:

$$\phi = kd \text{ for } \theta = 90°$$

At any angle of incidence θ the phase angle between the pressure contributions is dependent on the spacing d between the tube openings, and on the wavelength of the sound. Fig. 11.8 has been drawn to a larger scale for clarity and shows a plane wave incident at angle θ on two adjacent tubes. At the instant depicted, acoustic

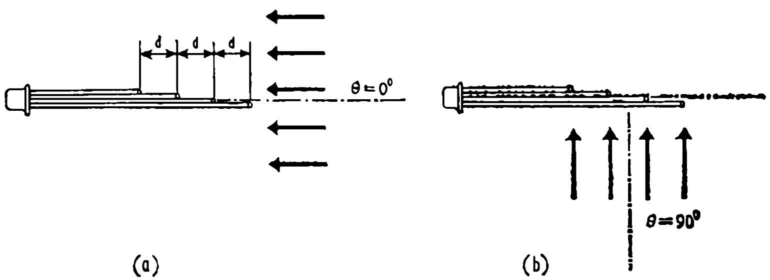


Fig. 11.7.—Plane wave incident at $\theta = 0°$ and $\theta = 90°$ on a four-tube directional microphone

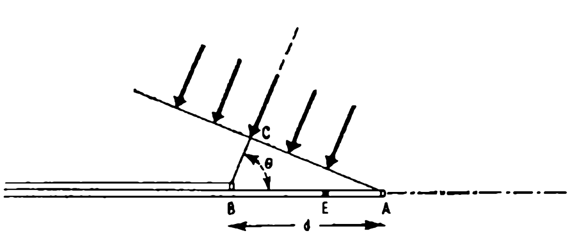


Fig. 11.8.—Diagram showing how the effective spacing between the pick-up points varies with angle of incidence

pressure is just entering the tube whose open end is at A. Before pressure can enter the tube whose open end is at B, the wavefront must move from C to B. Meanwhile, the pressure which entered at A moves to E. Since the velocity of sound in the tube is assumed to be equal to that in free space,

$$\begin{aligned} AE &= CB \\ &= d \cos \theta \end{aligned}$$

The effective path difference between the two pressures, represented by BE, is

$$\begin{aligned} BE &= AB - AE \\ &= d - d\cos\theta \\ &= d\,(1 - \cos\theta) \end{aligned}$$

If n tubes are employed in the instrument, differing in length by d, then the effective acoustic length l of the group is,

$$l = (n - 1)d$$

$$\text{or } d = \frac{l}{(n - 1)}$$

The phase angle in terms of l, n and θ is

$$\begin{aligned} \phi &= \frac{kl}{(n-1)}\,(1 - \cos\theta) \\ &= \frac{2\pi l}{\lambda\,(n-1)}(1 - \cos\theta) \end{aligned}$$

Pressure on the Diaphragm at Angle θ

Assuming no reflections within the tubes and neglecting any impedance-matching problems, the pressure P on the diaphragm is the vector sum of the pressure contributions from the n tubes. If at angle θ the pressure contributions differ successively from one another by a phase angle ϕ, then the resultant pressure on the diaphragm P_θ is (*see* Section 11.2.2),

$$P_\theta = p\,\frac{\sin n\phi/2}{\sin \phi/2}$$

where $\phi = \dfrac{2\pi l}{\lambda\,(n-1)}\,(1 - \cos\theta)$.

Since the output of the microphone E_θ is proportional to the pressure on the diaphragm,

$$E_\theta = ap\,\frac{\sin n\phi/2}{\sin \phi/2}$$

where a is a constant depending on the pressure unit employed.

11.3.2. DIRECTIONAL CHARACTERISTIC

At low frequency l/λ is small: hence the phase angle

$$\phi = \frac{2\pi l}{\lambda\,(n-1)}\,(1 - \cos\theta)$$

is small, so that

$$E_\theta = ap\,\frac{\sin n\phi/2}{\sin \phi/2} \simeq apn$$

since $\sin n\phi/2 \simeq n\phi/2$ and $\sin \phi/2 \simeq \phi/2$ when ϕ is small.

The expression is independent of θ, hence the microphone is non-directional at low frequencies or if l is small compared with λ.

As the frequency increases, the phase angle ϕ increases and its magnitude is dependent on the ratio l/λ and the value of $(1 - \cos\theta)$. Hence the output E_θ is dependent on the frequency and on the angle of incidence.

The directional characteristic can be obtained either from the equation for E_θ, or from the ratio:

$$R_\theta = \frac{\text{output at angle } \theta}{\text{output at } \theta = 0^\circ}$$

At $\theta = 0^\circ$ all the pressure contributions from the tubes are in phase, hence $E_\theta = apn$ and

$$R_\theta = \frac{ap.\dfrac{\sin n\phi/2}{\sin \phi/2}}{apn} = \frac{1}{n}\left(\frac{\sin n\phi/2}{\sin \phi/2}\right)$$

The directional characteristic for various values of l/λ for a 50-tube microphone is shown in Fig. 11.9. Comparing this characteristic with that of the array of microphones, it may be seen that the tubular directional microphone is uni-directional at frequencies where its length l is comparable with the wavelengths, and while it

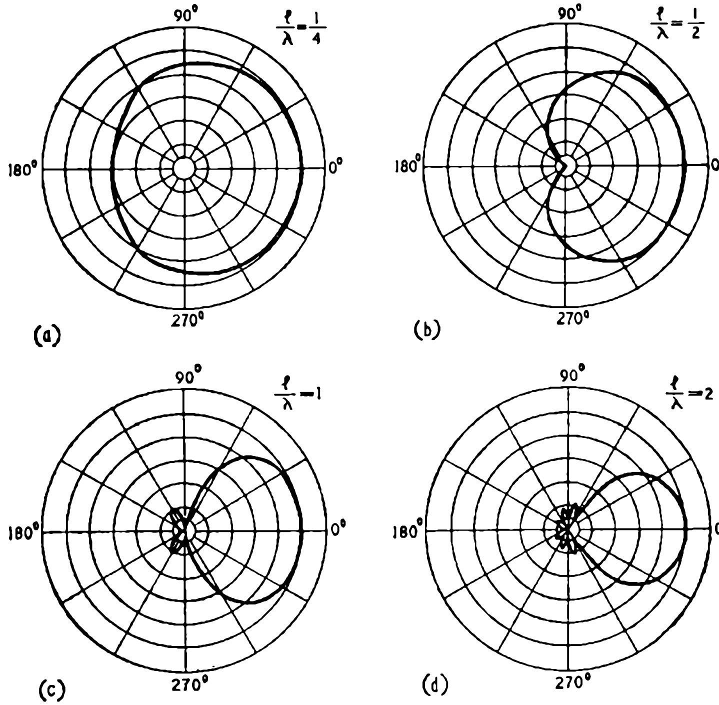


Fig. 11.9.—Polar response of a 50-tube directional microphone for various values of l/λ. (a) $l/\lambda = \frac{1}{4}$; (b) $l/\lambda = \frac{1}{2}$; (c) $l/\lambda = 1$; (d) $l/\lambda = 2$ (after W. P. Mason and R. N. Marshall)

discriminates against sounds in the third and fourth quadrant, it has less discrimination than the array in the first and second quadrants.

The Western Electric Company's microphone is about 5 ft long, this being a good compromise between directivity and ease in handling in confined spaces such as doorways and passages. With an effective length of 5 ft, the microphone is omni-directional at frequencies below about 100 c/s, as shown in Fig. 11.9 (a). At 200 c/s, $l/\lambda \simeq \frac{1}{2}$ and the directivity index and directional characteristic (Fig. 11.9 (b)) are comparable with those of a cardioid microphone. The directivity index for a number of frequencies is shown in Table 11.1 and was derived from measurements made indoors.

Outdoor tests with the microphone at a band contest showed that if the microphone was located at the centre of the field and aimed at the band, following it as it paraded to and fro, very pleasing results without equalisation were obtained; moreover, crowd noise was reduced to a minimum and some very annoying echoes audible on conventional microphones were completely eliminated. On one occasion in a large parade, three bands were all within range and all

Table 11.1. DIRECTIVITY INDEX FOR SOME FREQUENCIES

Frequency	200 c/s	500 c/s	1,000 c/s	2,000 c/s	4,000 c/s	8,000 c/s
Directivity index	$\frac{1}{3{\cdot}4}$	$\frac{1}{8}$	$\frac{1}{12}$	$\frac{1}{25}$	$\frac{1}{25}$	$\frac{1}{39}$

playing different music. With the tubular microphone, it was possible to pick up clearly any one of the three.

One effective demonstration was obtained by placing around the microphone six people in a circle, 60° apart, and asking all to talk at the same time. The microphone successfully eliminated all speech but that of the person towards whom it was pointed.

Since the directional characteristic varies very markedly with frequency, a microphone of this type would only be used indoors in exceptional circumstances. The omni-directional nature of the characteristic at low frequencies would probably require equalisation. Nevertheless, Mason and Marshall point out that in an auditorium where high-frequency reverberation is very annoying, the tubular microphone can make a tremendous improvement. In one particularly bad case, speech was clearly picked up from the rear of the auditorium by the directional microphone, but was unintelligible if an omni-directional microphone was used, because of excessive reverberation.

11.4. Multi-tube Directional Microphone with Progressive Delay

H. F. Olson, at about the same time as Mason and Marshall, produced a directional microphone of almost identical type. It too consisted of a group of tubes with their open ends as pick-up points, but a pressure-operated ribbon was used in place of the moving-coil transducer. The ribbon was coupled to the common junction of the tubes and was terminated in a long pipe suitably damped to act as an acoustic resistance.

In an improved version of this microphone, a progressive delay was inserted in each tube in the form of a crook, illustrated in Fig. 11.10 (a). The length of the crook was arranged so as to be proportional to, or to be $1/x$ of, the distance between the open end of the tube and its common junction. Thus each successive crook was longer than the previous one, the largest crook being inserted into the longest tube.

As shown in Fig. 11.10 (b), the effective length of the longest tube before insertion of the crook is $(n - 1)d$ where n is the number of

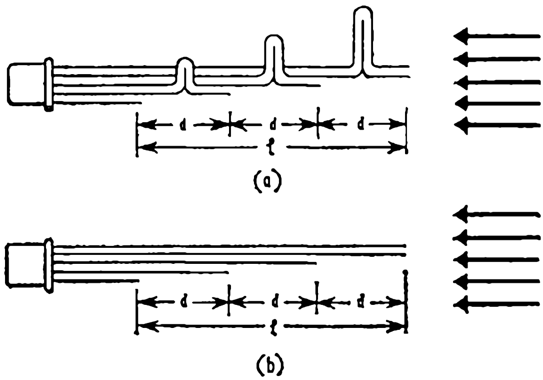


Fig. 11.10.—Four-tube directional microphone (a) with delay crooks and (b) without crooks

tubes in the group. The length of the crook inserted into this tube is $(n - 1)d/x$. The length of the second longest tube before insertion of the crook is $(n - 2)d$ and the length of its crook is $(n - 2)d/x$. The difference in length of adjacent crooks is therefore,

$$\frac{d}{x}(n-1) - \frac{d}{x}(n-2) = \frac{d}{x}$$

11.4.1. THE PHASE ANGLE AND RESULTANT PRESSURE

It has already been shown (Section 11.3) that the phase angle between the pressure contributions from adjacent tubes, which results from the spacing d between their open ends, is given by

$$\phi = k\,(d - d\cos\theta)$$

The insertion of the crooks in the tubes increases the path difference between adjacent tubes by d/x which in turn increases the phase angle by kd/x radians. Hence for a microphone with progressive delay proportional to the effective length of the tubes, ϕ is given by

$$\phi = kd\left(1 - \cos\theta + \frac{1}{x}\right)$$

If n tubes are employed, the effective length l of the group is

$$l = (n - 1)d$$

$$\text{that is, } d = \frac{l}{(n-1)}$$

$$\text{hence } \phi = \frac{2\pi l}{\lambda(n-1)}\left(1 - \cos\theta + \frac{1}{x}\right)$$

Having deduced the expression for ϕ, the resultant pressure P at the common junction of the tubes can be obtained using the vector polygon method (Section 11.2.5):

$$P = \frac{\sin n\phi/2}{\sin \phi/2}$$

$$= \frac{\sin \dfrac{n\pi l\,(1 - \cos\theta + 1/x)}{\lambda\,(n-1)}}{\sin \dfrac{\pi l\,(1 - \cos\theta + 1/x)}{\lambda\,(n-1)}}$$

From this expression the directional characteristic of the microphone can be calculated, assuming that the output of the transducer is proportional to the resultant pressure at the junction of the tubes.

11.4.2. DIRECTIONAL CHARACTERISTIC

The directional characteristic for various values of the ratio l/λ and for a delay path where $1/x = 1/4$ is shown in Fig. 11.11. A comparison of these characteristics with those of Fig. 11.9 shows that the same polar response can be obtained with a shorter length of line or that for the same value of l/λ the delay crooks increase the directivity of the instrument.

The characteristics also show that the progressive delay affects the axial response of the microphone, and that the output falls off as the frequency is increased. The reason for this is that the pressure

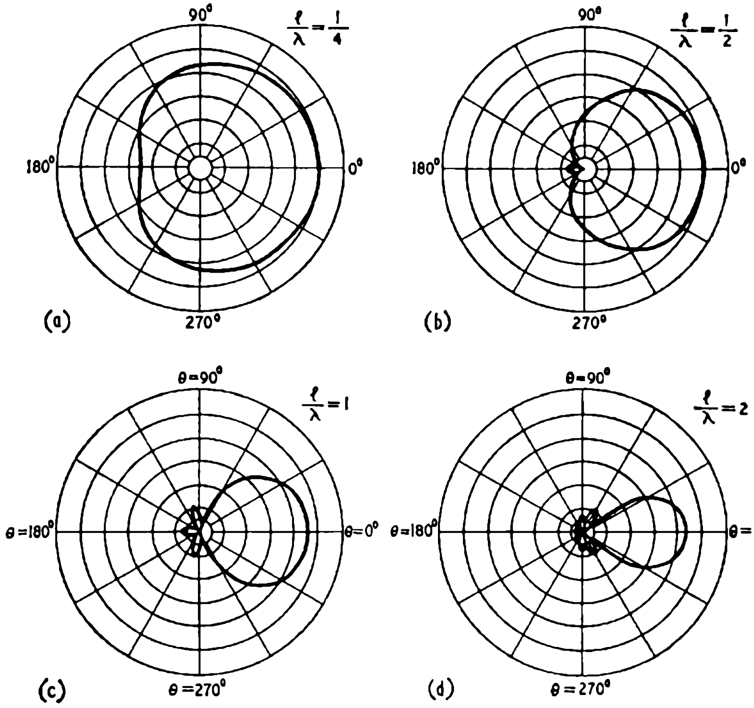


Fig. 11.11.—Polar response of directional microphone with progressive delay for various values of l/λ. (a) $l/\lambda = \frac{1}{4}$; (b) $l/\lambda = \frac{1}{2}$; (c) $l/\lambda = 1$; (d) $l/\lambda = 2$ (after H. F. Olson)

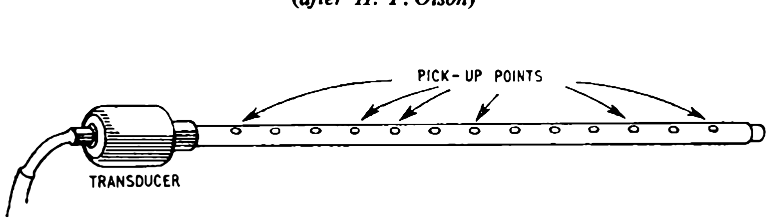


Fig. 11.12.—Schematic view of single-tube directional microphone

contributions for a plane wave at $\theta = 0°$ are no longer in phase at the common junction because of the different lengths of the delay crooks introduced in the tubes and, furthermore, the phase angle between the pressures increases with increase in frequency.

The general expression for ϕ is

$$\phi = kd\,(1 - \cos\theta + 1/x)$$

For axial incidence,

$$\theta = 0° \text{ and } \cos\theta = 1$$

hence $\phi = kd/x$.

Since $k = \dfrac{2\pi f}{c}$, the phase angle between the pressure contributions at $\theta = 0°$ increases as the frequency increases, and the axial response of the microphone falls off. The length of the crooks is therefore a compromise between directivity and a poor high-frequency response.

Tubular directional microphones have limited usefulness out-of-doors in conditions of other than still air, for winds of quite low velocity passing across the open ends of the tubes can produce edge-tones which may set the air in the tubes vibrating, resulting in excessive wind noise.

11.5. Single-tube Directional Microphones

Designers have tried from time to time to reduce the number of tubes used in the construction of a directional microphone without sacrificing its directional properties. Microphones have been produced employing only a single tube with a large number of pick-up points or sound entrances evenly distributed along the length of the tube (Fig. 11.12).

Olson showed that it was advisable to reduce the coupling between the sound entrances to a minimum, otherwise sound pressure which entered by one port could leave by one or more of the others before reaching the diaphragm of the transducer. He isolated the ports from each other by a system of cavities and passages in a manner similar to that already discussed in Chapter 9 in connection with variable path-length microphones.

11.5.1. SINGLE-TUBE DIRECTIONAL MICROPHONES FOR FREQUENCIES BELOW 1 C/S

An interesting microphone employing a single tube and operating below 1 c/s has been described by Daniels.[3] The frequencies for which it is intended are well below the range of audible sound, but a brief description of the microphone will be given because of the novel features associated with its design. It is intended to operate and be directional in the frequency range of $\frac{1}{40}$ c/s to 1 c/s and it has done useful work in connection with the study of microbaroms.

A *microbarom* is a cyclic variation in the earth's atmospheric pressure, believed to be produced by storm waves at sea. The great waves alternately compress and rarefy the atmosphere, producing micro-variations in the barometric pressure, which correspond in frequency to the ocean waves. The frequency of the barometric variations is of the order of $\frac{1}{5}$ c/s. It is therefore necessary to use a very long pick-up tube, and a pipe approximately 2,000 ft long is used to obtain directivity and so reduce wind and ambient noises.

Fig. 11.13 (a) is a diagrammatic sketch of the tube, showing the position of the sound entrances which are situated where the tube diameter changes abruptly. The sound entrances in the pipe are analogous to an alternator whose internal impedance is that of the entrance itself.

The impedance Z_{PA} of the sound entrance in pipe A is made equal to the characteristic impedance Z_{OA} of the section, that is,

$$\begin{aligned} Z_{PA} &= Z_{OA} \\ &= \frac{\rho c}{a_1} \end{aligned}$$

where a_1 is the cross-sectional area of the pipe A,
ρ is the density of the air,
c the velocity of sound in air.

The characteristic impedance Z_{OB} of the pipe B is $\dfrac{\rho c}{a_2}$, where a_2 is the cross-sectional area of pipe B. Since the diameter of B is larger than that of A, the characteristic impedance of the pipe B is less than that of A. This condition is made use of in arriving at an appropriate value for the impedance Z_{PB} of the sound entrance in pipe B.

The value of Z_{PB} is so arranged that

$$Z_{PB} \| Z_{OA} = Z_{OB}$$

$$\text{or } \frac{1}{Z_{PB}} + \frac{1}{Z_{OA}} = \frac{1}{Z_{OB}}$$

$$\text{or } \frac{1}{Z_{PB}} = \frac{1}{Z_{OB}} - \frac{1}{Z_{OA}}$$

That is, the combination of Z_{PB} in parallel with the characteristic impedance of the section of the line to the right of YY' (Fig. 11.13) is equal to the characteristic impedance of the section to the left of it. The sections are thus correctly terminated for transmission from left to right. This means that sound pressures which enter the ports and travel to the left reach the diaphragm of the transducer, but those which travel towards the right do so down a pipe which is effectively infinitely long. Thus there are no reflections from the distant end of the pipe or from the discontinuities associated with the changes in diameter to cause complications at the diaphragm.

Twelve pipe sections of different diameter are used in the construction of the microphone pick-up tube. The characteristic impedance of a section of pipe is inversely proportional to its diameter and

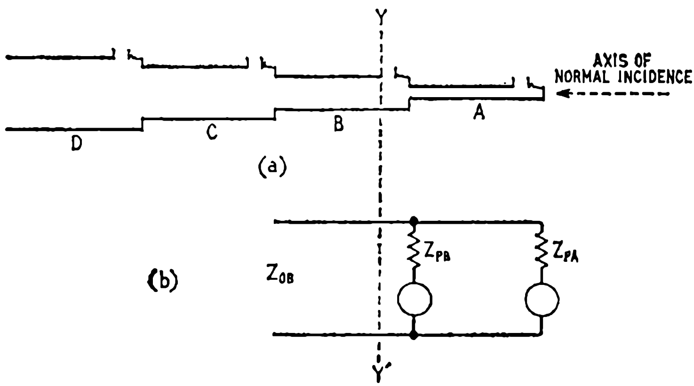


Fig. 11.13.—(a) Schematic section of directional microphone for frequencies below 1 c/s. (b) Equivalent circuit of sound entrances

because standard pipes are used, the diameter is not under the control of the designer and consequently the changes of impedance which occur at the pipe junctions are not equal. Acoustical resistances are inserted in the pipe; they have the effect of changing the impedance of the section to the desired value.

We have seen that the sound entrances must be placed in the pipe in the region where the impedance changes abruptly and if the entrances are placed only at the junctions of sections, their number is not large enough to ensure the desired polar response. Additional acoustical resistances are inserted in the pipe, about 20 ft apart, and distributed symmetrically with respect to the junctions. These have the effect of changing the characteristic impedance of the section. Using the twelve pipe junctions and the parallel-resistance technique just described, 99 equal changes in pipe impedance are contrived, making it possible to position correctly the 100 sound entrances required in the design. The acoustical resistances consist of holes of small diameter drilled through brass plugs inserted in the wall of the pipe. Constructed in this way the line microphone is non-reflecting and has virtually no loss over the frequency range for which it was designed.

11.6. Tubular-line Microphones

When discussing the principle of operation of line microphones, it was assumed that the velocity of sound in the tubes is the same as that in free space. This assumption is not valid if tubes of small diameter are used in the construction of the line, for the velocity of sound in the tubes, especially at low frequencies, is less than that in free space and as a consequence the pressure contributions from the pick-up points at $\theta = 0°$ do not arrive in phase at the diaphragm of the microphone. The axial sensitivity of the system is therefore reduced. Furthermore, any mismatch between the diaphragm impedance and the impedance of the tubular line or group of tubes will result in a further loss of general sensitivity.

Reference was made in Chapter 7 to a moving-coil microphone which was provided with a matching section so that it could be coupled, with minimum loss, to a tube to form an inconspicuous microphone. The matching section employed was analogous to a velocity transformer and provided an almost exact match between the resistance-controlled diaphragm of the microphone and the characteristic impedance of the tube, thus reducing reflection loss and eliminating tube resonance. This type of matching section has been developed and is incorporated in a directional microphone, manufactured by the Sennheiser Electronic Company of West Germany.

The Sennheiser microphone has a tube about 1 in. in diameter and 80 in. long, coupled to an electrodynamic pressure unit via a matching section, illustrated in Plate 11.1 (a) and (b). The tubular line is unique in that it can be considered to have an infinite number of pick-up points, which are arranged in the form of a narrow slot running the full length of the tube. Sound pressures enter the tube along the whole length of the slot but in order to do so must pass through a layer of felt before reaching the interior of the tube. The impedance of the felt is of the form $R + j\omega m$ and provides the necessary damping and de-coupling between adjacent elements of the slot.

11.6.1. PRINCIPLES OF OPERATION

For sounds incident at $\theta = 0°$ the pressure contributions from the slot are all in phase at the transducer end of the tube and the magnitude and phase of the resultant pressure p on the diaphragm are represented by the vector AB in Fig. 11.14 (a). AB is the resultant of a large number of very small pressure contributions, all of the same phase.

When sounds arrive at angle θ to the normal axis of incidence, the pressure contributions are no longer in phase at the diaphragm but

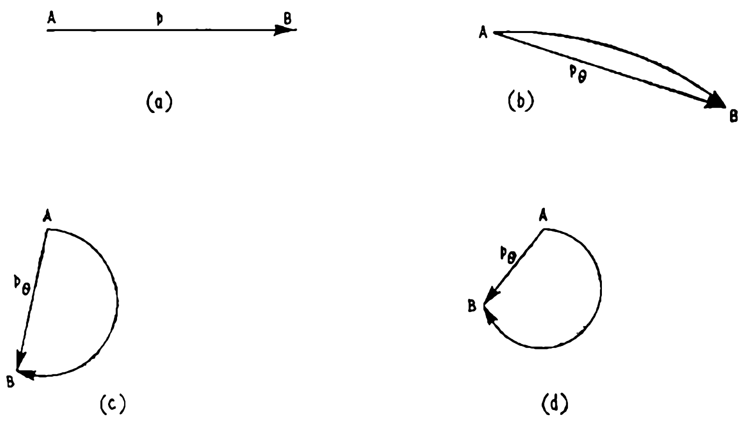


Fig. 11.14.—Vector diagrams of a tubular line pick-up system showing the resultant pressure on the diaphragm for various angles of incidence θ (Sennheiser Microphone)

differ successively by a small phase angle. The resultant pressure p_θ is less than p and in Fig. 11.14 (b) its magnitude and phase are represented by the chord AB. The arc AB in the figure is a portion of a vector polygon having a large number of infinitely small sides. Each small side is in effect part of the arc of the circle. For small angles of incidence the radius of the arc containing the sides of the polygon is large, consequently the arc and the chord are approximately equal in length, in which case $p \simeq p_\theta$, but if θ is large the radius of the circle is small and the arc tends to close, thus bringing about a change in both the magnitude and the phase of p_θ (Fig. 11.14 (c) and (d)).

It should be noted that the arcs *AB* in Fig. 11.14 (b), (c) and (d) are all of the same length and are equal to the length of the vector *AB* in Fig. 11.14 (a). Varying the angle of incidence θ does not change the length of the arc *AB* but merely changes its curvature, so altering the length of the chord *AB* and hence the magnitude and phase of p_θ.

Expression for p_θ in Terms of the Pressure for Axial Incidence p

At any angle of incidence θ pressures enter the slot along its whole length and the resultant pressure p_θ on the diaphragm is represented in Fig. 11.15 (a) by the chord *AB*. *AD* and *BD* are tangents to the arc and indicate the phase of the pressure contributions which enter the

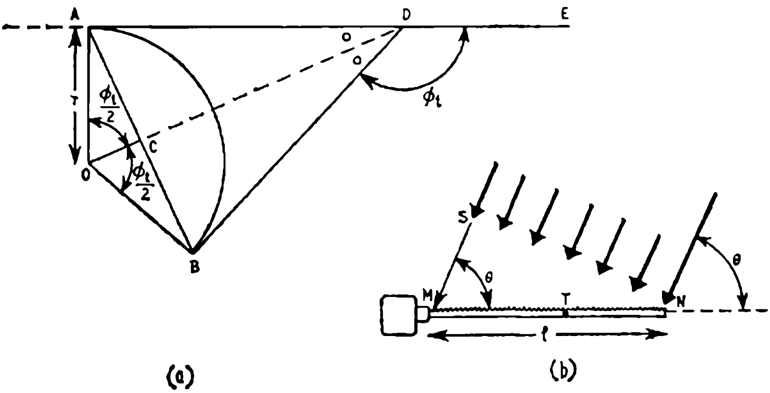


Fig. 11.15.—(a) Sennheiser microphone: vector diagram showing the resultant pressure on the diaphragm at angle of incidence θ. (b) Diagram showing how the phase angle between the pressure contributions from the near and distant end of the line varies with angle of incidence θ

tube at the extremities of the slot (*M* and *N* in Fig. 11.15 (b)). The angle $B\hat{D}E$ is the phase angle ϕ_t between the pressure contributions at *M* and *N*.

In Fig. 11.15 (a) *AO* and *BO* intersect at *O* and since they are drawn perpendicular to the tangents *AD* and *BD*, then *O* is the centre of the circular arc *AB* and *AO* and *BO* are radii.

$$A\hat{O}B = B\hat{D}E = \phi_t$$

The resultant pressure p_θ is represented by the chord AB and

$$\begin{aligned} p_\theta &= AB \\ &= 2AO \sin \phi_t/2 \\ &= 2r \sin \phi_t/2 \end{aligned}$$

since AO is the radius of the circular arc AB.

Since ϕ_t is in radians, the radius r can be expressed in terms of the angle ϕ_t and the arc AB, that is,

$$\begin{aligned} r &= \frac{arc\ AB}{\phi_t} \\ &= \frac{p}{\phi_t} \end{aligned}$$

Hence the pressure p_θ on the diaphragm for an angle of incidence θ is given by

$$p_\theta = \frac{2p}{\phi_t} \sin \frac{\phi_t}{2}$$

The Total Phase Angle ϕ_t

In Fig. 11.15 (b) the path difference between the pressure entering the tube at N and the pressure entering at M is MT (page 213) and

$$\begin{aligned} MT &= MN - NT \\ &= l - l \cos \theta \\ &= l\,(1 - \cos \theta) \end{aligned}$$

Hence the phase angle

$$\phi_t = kl\,(1 - \cos \theta)$$

Substituting this value for ϕ_t in the expression for p_θ we have

$$\begin{aligned} p_\theta &= \frac{2p}{kl\,(1 - \cos \theta)} \frac{\sin kl\,(1 - \cos \theta)}{2} \\ &= \frac{p\lambda}{\pi l\,(1 - \cos \theta)} \sin \pi \frac{l}{\lambda}(1 - \cos \theta) \end{aligned}$$

11.6.2. THE DIRECTIONAL CHARACTERISTIC

The polar response or directional characteristic is obtained from the ratio R_θ where

$$R_\theta = \frac{\text{pressure on the diaphragm at angle } \theta}{\text{pressure on the diaphragm at } \theta = 0^\circ}$$

$$= \frac{p_\theta}{p}$$

$$= \frac{\lambda}{\pi l\,(1 - \cos\theta)} \sin \pi \frac{l}{\lambda}(1 - \cos\theta)$$

When values are assigned to l, λ and θ in the expression for R_θ, a directional characteristic is obtained which, in so far as the lobe of

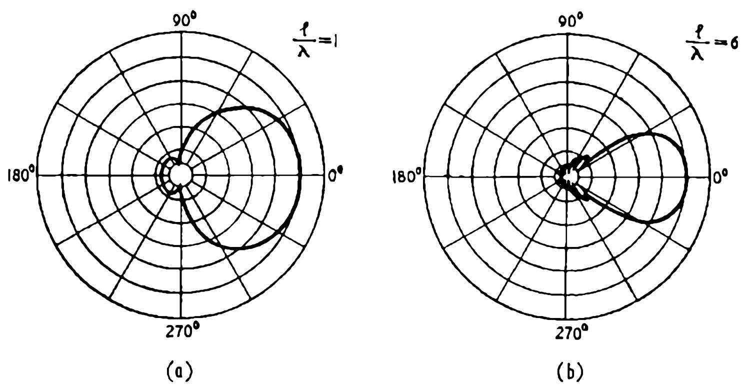


Fig. 11.16.—Polar response of tubular line directional microphone for various values of l/λ. (a) l/λ = 1; (b) l/λ = 6 (courtesy Sennheiser Electronic Company)

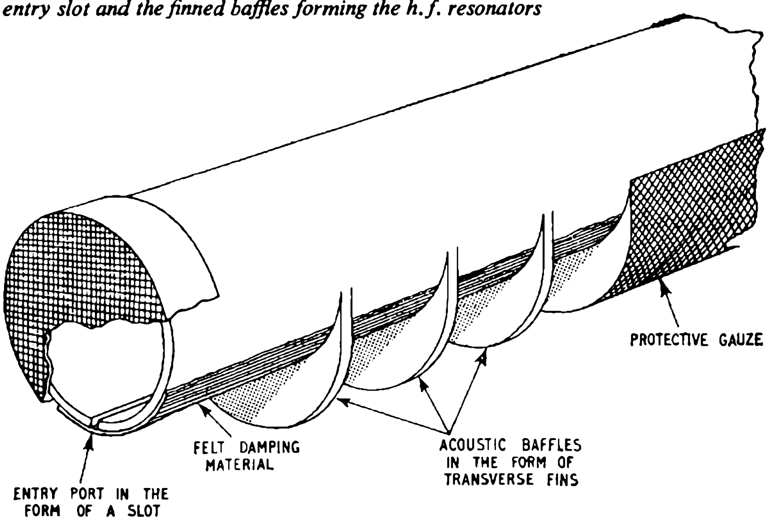
maximum sensitivity is concerned, is very similar in shape to that of the Mason and Marshall microphone: that is, the major lobe narrows as the frequency increases and the microphone becomes more and more directional. A microphone with such a polar response might be satisfactory if used out-of-doors but its usefulness in a studio or concert hall would be limited. To limit the variations in the polar response, an artifice is employed, very similar to that already described in Chapter 9 in connection with multi-path phase-shift microphones. In the Senneheiser microphone the acoustic impedance of the felt through which the sound must pass to reach the interior of the tube is not uniform throughout the length of the slot, but increases progressively towards the distant end of the tube. The effect of the graded impedance is to shorten the active length of the slot as the frequency increases, and so produce a directional characteristic which is appreciably independent of frequency.

The directional characteristic of the microphone with an acoustically corrected tube is shown for two values of l/λ in Fig. 11.16 (a) and (b). If an 80 in. tube is used, the $l/\lambda = 1$ diagram corresponds to a frequency of about 170 c/s. Even at this comparatively low frequency, the microphone is very directional with a front-to-back ratio of the order of 15 dB. As the frequency increases, the width of the major lobe decreases but the $l/\lambda = 6$ characteristic (corresponding to a frequency of 1,000 c/s) shows that the reduction in width of the lobe of maximum sensitivity is not so marked as in microphones having a fixed effective tube length.

11.6.3. SENSITIVITY AND AXIAL FREQUENCY RESPONSE

The presence of soft absorbent damping material within the tube together with the matching section largely eliminates standing waves in the tube and the response is smooth and without irregularities. It must be appreciated, however, that the acoustic impedances which are used to prevent the entry of high-frequency sound pressures into

Fig. 11.17.—Tubular line microphone showing felt-covered entry slot and the finned baffles forming the h.f. resonators



the tube along its whole length also reduce the axial sensitivity of the microphone and affect its high-frequency response. The high-frequency losses are offset by the provision of a series of baffles in the form of transverse fins set at intervals along the underside of the tube (Fig. 11.17). The baffles enclose small volumes of air which resonate at high frequencies, increasing the acoustic pressures in the immediate vicinity of the slot. The cavity resonance is a maximum when sounds arrive along the axis of the slot and a minimum when sounds arrive from the sides. In this way the axial sensitivity of the microphone is increased at high frequencies without unduly affecting the directional response.

11.6.4. OPERATIONAL USE

The open-circuit sensitivity is of the order of − 79 dB with reference to 1 V/dyn/cm^2, which is about 7 dB below that of a conventional moving-coil microphone operating in identical acoustical conditions. Since the tubular microphone normally operates at a greater distance than normal from the sound source, considerable amplification may be required because its output under these circumstances will in general be well below the values quoted. The microphone is light and easily handled and is particularly useful on a site exposed to strong winds.

It may be necessary in certain circumstances to resort to electrical equalisation, in order to compensate for the reduced directivity at low frequencies. In this connection it should be remembered, if the microphone is used indoors, that electrical correction can reduce the output of the microphone at certain frequencies or over a range of frequencies; that is, it can reduce the sum of the direct and indirect energy pick-up, but it cannot alter one without altering the other and so cannot bring about a change in the ratio of direct to indirect sound. It is therefore no substitute for a directional response which is independent of frequency.

11.7. Directional Microphone using a Parabolic Reflector

Concave surfaces and parabolic reflectors have been used for many years to pick up faint sound coming from a distant source and to concentrate the received energy at a point in space called the *focus* of the reflector. If the reflector is large or very large in comparison to the wavelengths, an appreciable increase in acoustic pressure can be obtained at or in the region of the focus.

Before the days of radar, acoustic reflectors were used in England to give early warning of the approach of hostile aircraft. Sound from aircraft is of low pitch and to be effective, the reflecting surfaces had to be of considerable size. Reflectors made of concrete in the shape of paraboloids, with diameters approaching 40 ft, were built into suitable hillsides in the south of England and were used to pick up the noise of aircraft engines over the English Channel. Since they operated on a fixed bearing, they were limited in their usefulness and were soon replaced with more mobile sound-locating equipment.

11.7.1. PARABOLIC REFLECTOR AND PRESSURE UNIT

The simplest and the earliest of the directional microphone systems used in broadcasting in Britain consisted of a parabolic reflector which focused the sound on the diaphragm of a moving-coil pressure-operated microphone positioned at the focus of the paraboloid. Plate 11.2 shows a modern version of the equipment, designed primarily for use in connection with outside broadcasting. The reflector is of aluminium, 3 ft in diameter and 9 in. deep, and is of the type produced for radar. Since it is intended for use out-of-doors, it is backed with ¼ in. thick sponge rubber, to reduce noise from rain to an acceptable level. The microphone is held in an adjustable mounting and can be moved towards or away from the focus of the reflector. When mounted on a standard television tripod, the complete assembly can be rotated about either the vertical or the horizontal axis.

11.7.2. PRINCIPLE OF OPERATION

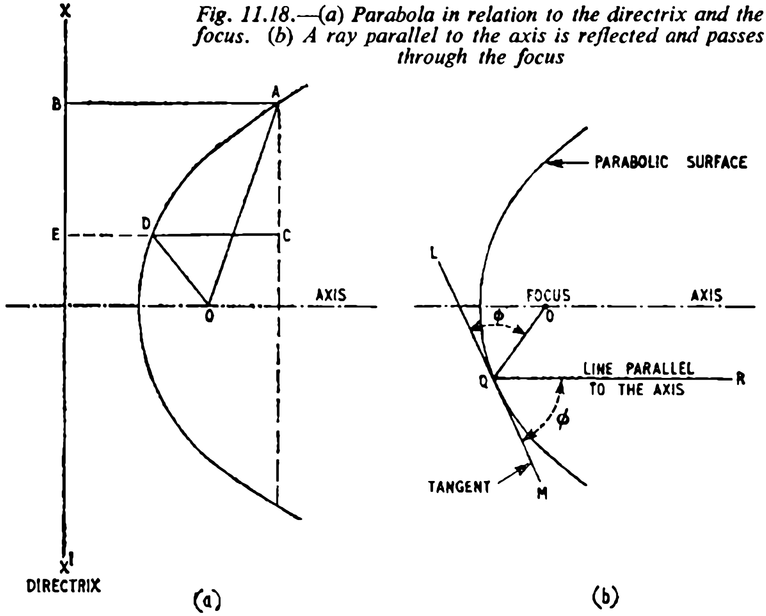
The principle of operation is that sound energy reaching the total area of the reflector from an axial direction is focused or concentrated on the diaphragm of the microphone which faces away from the sound source. The focusing effect depends on two geometrical properties of the parabola.

The first property is this: a parabola is defined as a locus of points which are as far from a fixed point called the focus as from a fixed line called the *directrix*. In Fig. 11.18 (a) the focus is the point O and the directrix the line XX'. AC is a line parallel to the directrix and at right angles to the axis of the parabola. A and D are any two points on the parabola, and by definition $AO = AB$ and $DE = DO$.

$$\begin{aligned} AO &= AB \\ &= CD + DE \\ &= CD + DO \\ \text{i.e., } AO &= CD + DO \end{aligned}$$

This is an important statement and will be referred to later.

Fig. 11.18.—(a) Parabola in relation to the directrix and the focus. (b) A ray parallel to the axis is reflected and passes through the focus



The second important geometrical property of the parabola is that a line drawn as a tangent to a parabola at any point makes equal angles with a line drawn from this point back to the focus and with a line drawn from the point parallel to the axis of the parabola. In Fig. 11.18 (b) LQM is a tangent to the parabola with the point of contact at Q.

$$\therefore\ M\hat{Q}R = L\hat{Q}O$$

This is also an important concept, for if RQ is a beam or pencil of sound arriving parallel to the axis of the parabola, it will on reflection from the surface at Q pass through the focus F.

These two properties are combined in Fig. 11.19, where AA' represents a wavefront whose direction of propagation is along the axis of the parabola. Each element of the front on contact with the parabolic surface is reflected towards the focus and although different elements of the front reach the focus by different paths, the lengths of all the paths are equal. Thus sounds arrive at the focus with the same phase relationship as was present in the original wavefront. Reflecting surfaces with this valuable property are referred to as constant-phase surfaces. Inevitably some sound also reaches the diaphragm by diffraction around the case of the microphone: it is of a different phase, having traversed a shorter path, and tends to reduce the efficiency of the constant-phase surface. Fortunately, the effect is not serious in acoustic techniques.

If the source of sound lies off the axis of the paraboloid, then the point of maximum concentration is also off the axis and the output of the microphone falls in proportion to the increase in the angle of incidence. For a dish-shaped reflector, directivity in both the horizontal and the vertical planes is obtained when the diameter D of the reflector is large relative to the wavelength of the received sound. A 3 ft reflector is effective down to frequencies of the order of 800 c/s

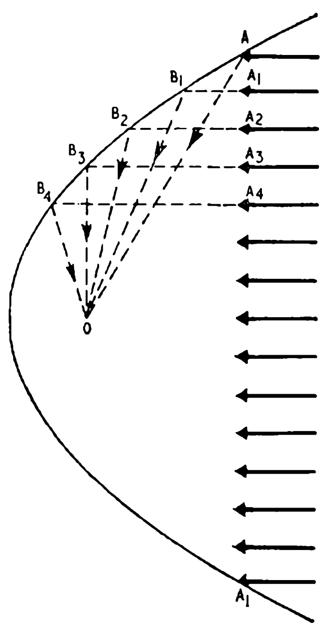


Fig. 11.19.—Plane wave brought to a focus at O

when $D/\lambda \simeq 2$, but it becomes progressively less directional as the frequency is reduced. At low frequencies the combination of reflector and microphone tends to be omni-directional and can pick up sounds of low pitch arriving from sources other than the programme source. To reduce this type of noise, bass cut is provided by a 1 μF series capacitor in the secondary circuit of the microphone transformer.

The polar graphs of Fig. 11.20 (a) and (b) show the directional characteristic for $D/\lambda \simeq \frac{3}{4}$ corresponding to a frequency of 250 c/s, and $D/\lambda \simeq 3$ corresponding to a frequency of 1,000 c/s. One of the difficulties associated with the reflector system is the highly directional nature of the characteristic; when $D > \lambda$, any movement of the sound source which varies the angle of incidence by even a few degrees can produce marked variations in programme quality and level.

To a limited extent, the variations in the directional characteristic with frequency can be controlled by varying the effective size of the reflector. This is done by covering the outer surface of the reflector with a ring-shaped layer of an acoustically absorbent material. The coefficient of absorption of a soft absorber such as felt varies with frequency from about 0·1 at 100 c/s, to about 0·5 at 5,000 c/s. At high frequencies the treated surface of the reflector absorbs rather than reflects the sound and in this way the effective area of the paraboloid is reduced. At low frequencies, little sound is absorbed and there is therefore little change in the effective area. Unfortunately, rain and dampness alter the absorbent properties of the material and can produce uncontrolled changes in both gain and directional characteristic.

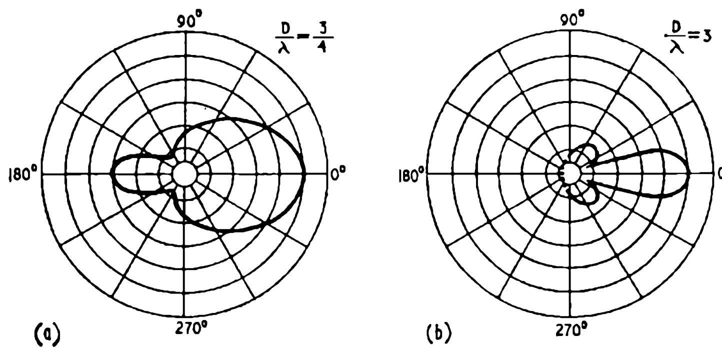


Fig. 11.20.—Polar response of a paraboloid reflector and microphone system for various values of D/λ. *(a)* $D/\lambda = \frac{3}{4}$; *(b)* $D/\lambda = 3$

An alternative to the acoustically corrected reflector (one that is used in the directional microphone system of Plate 11.2) is to displace the microphone some few inches from the geometric focus of the paraboloid. If the microphone is moved away from the focus along the axis of the paraboloid, the directional characteristic widens and becomes less dependent on frequency and at the same time the gain is reduced. Although the output level falls, it also tends to be less dependent on frequency. The effect of defocusing is illustrated in Fig. 11.21, which shows the major lobe of the characteristic at two frequencies, 1 kc/s and 4 kc/s, with the microphone 3·9 in. away from the focus. Over the bandwidth of two octaves the directional

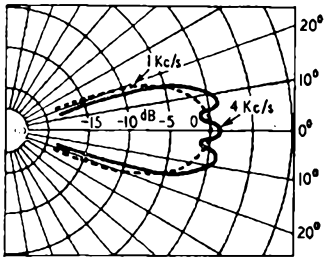


Fig. 11.21.—Diagram showing how the width of the directional characteristic remains sensibly uniform if defocusing is employed

characteristic remains appreciably constant and although there are variations in gain, these are not serious since they amount to only a few dB over an acceptance angle of 20°.

In practice it has been found that if the sound source is more than 35 ft away, the microphone may with advantage be moved closer to the focus. A setting of 1½ in. in front of the geometric focus is generally satisfactory, provided the maximum angle subtended by the sound source does not exceed 10°, and provided that any movement of the sound source will not take it out of the acceptance angle of the reflector, which corresponds to an aperture diameter of 6 ft at the distance quoted.

If the sound source is about 25 ft from the reflector, a greater degree of defocusing is necessary. In general the optimum position of the microphone for this distance is 2·6 in. from the focus, which retains the aperture diameter of 6 ft. If the source is within 15 ft of the reflector, the microphone must be moved still further from the focus to preserve the aperture diameter and it is normal practice to operate, at this close range, with the microphone 3·9 in. away from the focus.

The tripod and mountings of a paraboloid reflector system of this type must be rigid, especially when used out-of-doors, otherwise the reflector would move in a high wind and variations in both level and quality could be produced.

It will be appreciated that with a large paraboloid, some form of sighting arrangement is necessary and the reflector is "aimed" with the aid of a fore-sight in the form of a ring projecting below the microphone. This is viewed from behind the paraboloid through a small circular window of Perspex scribed with cross-lines, which constitutes the back-sight.

11.7.3. AXIAL RESPONSE

The curves in Fig. 11.22 show the axial response of the reflector and microphone system in the open air. With the source at a distance of 75 ft and with the microphone displaced 1·6 in. from the focus, the

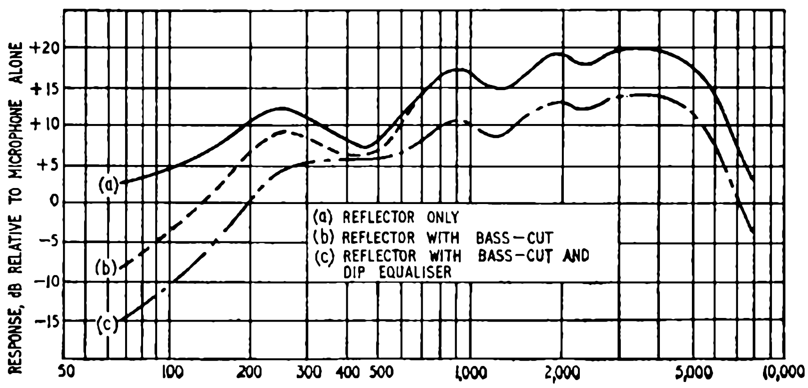


Fig. 11.22.—Showing the axial response of the reflector and microphone system with: (a) reflector only; (b) with bass cut and (c) with bass cut and dip equaliser

full-line curve (a) is the gain/frequency characteristic relative to the response of the microphone without reflector. The effect of introducing a bass cut by means of the 1μF series capacitor is shown by the broken line (b). Both curve (a) and curve (b) show a dip in the response between 400 c/s and 500 c/s, which is followed by an increase between 200 c/s and 300 c/s. A dip equaliser is provided to smooth out the variation and its effect on the gain/frequency characteristic is illustrated by the chain-dot curve (c).

11.8. A Directional Microphone using an Acoustic Lens and Horn

Comparisons of the relative merits of reflectors and lens systems have been made by Kock[5] in connection with microwave aerials, and while many of the criticisms levelled at reflectors are justified in systems which operate at high gain and at a single frequency, they are not necessarily applicable to other techniques. The acoustic reflector is not a single-frequency device but must work satisfactorily over a bandwidth of several octaves and therefore different criteria are involved. For example, while gain is desirable, it is less important than a polar response which is independent of frequency and as we have already seen, gain is invariably sacrificed in order to achieve this end. A valid criticism of the reflector is that it is directional only so long as the diameter is comparable with or larger than the wavelength; furthermore, the side-lobes in the characteristic at certain frequencies are undesirable and can lead to operational difficulties. To a certain extent this criticism also applies to the acoustic lens but a partial solution results if the lens is used in conjunction with a horn as in Fig. 11.27. In this arrangement the horn acts as an effective shield and produces a more acceptable directional characteristic at large angles of incidence.

11.8.1. ACOUSTIC REFRACTORS

Acoustic lenses or refractors are generally of two types, the obstacle type and the path-length delay type. In general construction and in their mode of operation they are very similar to the lenses recently developed in connection with microwave electromagnetic techniques.

Obstacle Refractors

The obstacle type of lens consists of an arrangement of rigid objects, usually in the form of strips or discs, whose size is small compared with the wavelengths. Because the objects are rigid they possess mass and when concentrated in a medium such as air, they increase the refractive index of the medium by increasing its effective density. Since the velocity of sound in a medium is inversely proportional to the square root of the density, the velocity of propagation through the array of objects is less than that in free space. The change in the velocity of sound in the medium can produce refraction or bending of the wavefront (Fig. 11.23 (a)) and by arranging the objects in the appropriate concentration and in suitably shaped groups, divergent or convergent lenses can be constructed, as illustrated in Fig. 11.23 (b).

Lenses of this type work satisfactorily only when the spacing between the objects is large compared with the wavelength, so that no distortion of the sound field occurs in the vicinity of individual objects. If in order to obtain a large refractive index the objects are closely grouped, the operation of the lens will be affected. Furthermore, the objects must not only be suitably spaced but their size and spacing relative to the wavelength must also be considered. At the shorter wavelengths the obstacle size may approach that of the

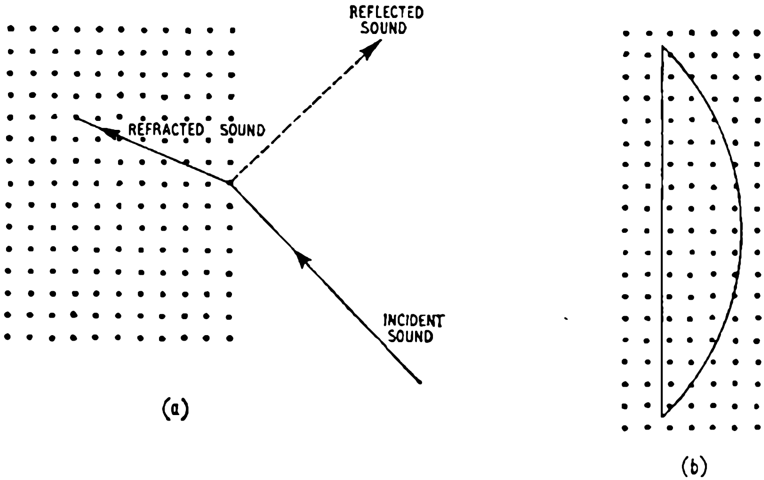


Fig. 11.23.—(a) Reflection and refraction of a sound beam entering a medium of increased density. (b) Objects arranged in appropriate concentration and in suitably shaped groups can be used as convergent or divergent sound lenses

wavelength of the incident sound, in which case the wave will be reflected by the objects setting up resonance effects within the lens. At these frequencies the lens acts as a reflecting surface rather than one which is transparent to sound.

It is not proposed to deal in detail with obstacle lenses, since a discussion of the refractive index[6] of such lenses is outside the scope of this book.

Path-length Refractors

Another method of slowing down a progressive wave and so altering its velocity is to direct different elements of the wavefront along different paths whose lengths are greater than that which the unguided wave would normally take in free space.

The principle of operation of path-length refractors is illustrated in Fig. 11.24. *AB* and *CD* in Fig. 11.24 (a) represent two planes in free space separated by a distance W. If c is the velocity of sound in free space, the time taken t to traverse the distance W is given by

$$t = \frac{W}{c}$$

If a number of flat parallel plates of width W is arranged between the two planes and aligned along the direction of propagation, the wave will pass through this new medium with negligible change in velocity. Hence the refractive index n of a parallel-plate medium arranged as in Fig. 11.24 (b) is unity, i.e. $n = 1$.

If however the plates are inclined at an angle α to the direction of propagation, as in Fig. 11.24 (c), the time taken to pass through the slant-plate medium will be unchanged, since the wave must traverse

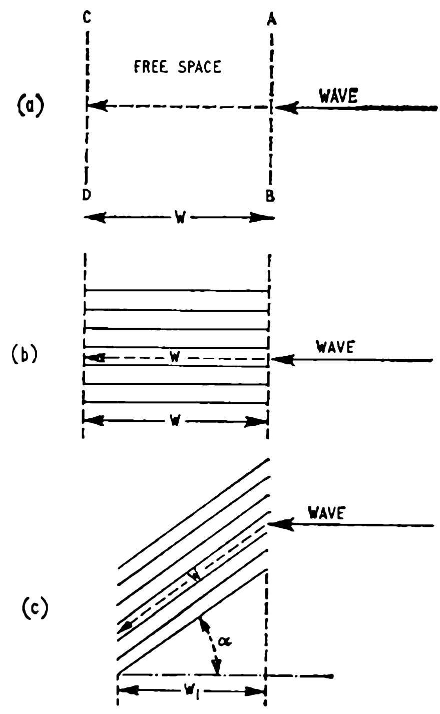


Fig. 11.24.—Diagram showing the apparent change in velocity of a sound wave through a slant-plate medium

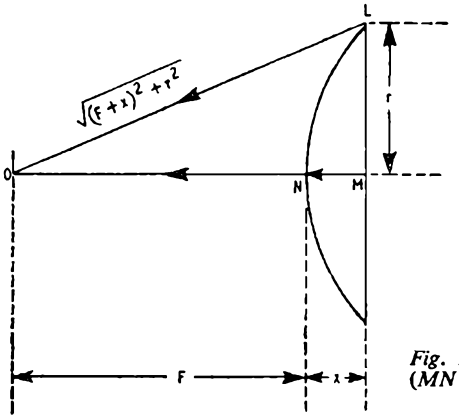


Fig. 11.25.—The two paths LO and (MN + NO) through a plano-convex lens are equal

the same distance W to penetrate the medium, but by slanting the plates the width of the medium has been reduced from W to W_1. Hence the apparent velocity of sound c_1 in the slant-plate medium is less than in free space, because

$$c_1 = \frac{W_1}{t}$$

$$\text{The refractive index } n = \frac{\text{velocity of sound in air}}{\text{velocity of sound in the medium}}$$

$$= \frac{c}{c_1} = \frac{ct}{W_1}$$

$$t = \frac{W}{c}$$

$$\text{Hence } n = \frac{W}{W_1}$$

$$= \frac{1}{\cos \alpha}$$

Certain glasses used in the construction of light lenses have a refractive index of 1·5. A refractive index of this order is obtained for a slant-plate lens when $\alpha = 48{\cdot}3°$.

Equation of the Lens Profile

The equation of the lens profile of the slant-plate plano-convex lens of Fig. 11.25 can be obtained in terms of the refractive index n, the desired focal length F and the radius r of the lens using the concept that if the phase relationship in the wave is to be preserved, the various elements of the wave, after passing through the lens, must arrive at the focus at the same time. That is, the time taken to traverse LO = time taken to traverse $MN + NO$.

Writing the distances LO, MN and NO in terms of the focal length F, the depth of the lens x and the radius r, we have

$$LO = [(F + x)^2 + r^2]^{\frac{1}{2}}$$

$$MN + NO = F + x$$

If c is the velocity of sound in air and c_1 is the velocity of sound through the lens medium, then the time t taken to traverse LO is given by

$$t = \frac{[(F + x)^2 + r^2]^{\frac{1}{2}}}{c}$$

and the time taken to traverse $MN + NO$ is

$$t = \frac{F}{c} + \frac{x}{c_1}$$

Since the two paths are traversed in equal time,

$$\frac{F}{c} + \frac{x}{c_1} = \frac{[(F + x)^2 + r^2]^{\frac{1}{2}}}{c}$$

Multiplying by c and writing n for c/c_1, we have

$$F + nx = [(F + x)^2 + r^2]^{\frac{1}{2}}$$

$$F^2 + 2nFx + n^2x^2 = F^2 + 2Fx + x^2 + r^2$$

$$\text{or } (n^2 - 1)x^2 + 2Fx\,(n - 1) - r^2 = 0$$

This is the equation of a hyperbola.

Variation in path length through optical and acoustical lenses adversely affects their performance but Lord Rayleigh has shown that small departures from the ideal are unimportant, and if the errors in path length are less than $\lambda/4$, the lens is practically perfect. This accuracy, while difficult to achieve optically, can with care be attained in certain acoustical systems. It is therefore claimed for the acoustic lens that unlike the parabola, its performance is not unduly

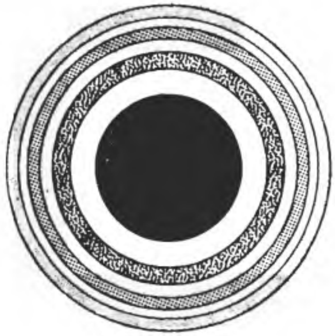


Fig. 11.26.—Diffraction pattern showing central circular area of maximum energy and the concentric rings of diminishing energy

affected by a departure from the optimum profile, and a spherical surface which approximated to the hyperbolic shape would produce an appreciable concentration of sound in the region of the focus.

It will be appreciated however, that unlike light lenses, acoustic lenses are normally expected to operate over a band of frequencies of several octaves. It is therefore something of a misconception to refer to the focus of the lens as a point, especially when the wavelength involved may be measured in inches or even feet. The " image " of a distant point source of sound, brought to a focus by the lens, is not a geometric point but takes the form of a central circular area of finite dimensions surrounded by rings of decreasing maxima and minima (Fig. 11.26). The diffraction pattern so formed was investigated by Airy in connection with light lenses in 1834. He showed mathematically that the central disc received about 84% of the total energy passed by the lens and that the concentric rings received an amount of energy which diminished as their distance from the central disc increased. The ring closest to the centre received 7% and the others 3%, 1·5%, 1%, etc.

For all practical purposes in acoustics the diffraction pattern may be assumed to consist merely of the central circular area of maximum energy. The diameter δ of the central area is given approximately by the following formula:

$$\delta \simeq 2{\cdot}5 \ \frac{F\lambda}{D}$$

where F is the focal length of the lens,
λ is the wavelength,
D is diameter of the lens (*see* Appendix 12).

The equation indicates that the lens has no appreciable focusing action unless its focal length is small or its diameter large compared with the wavelength. In a practical lens, the focal length and the diameter set the low-frequency operating limit. The upper frequency limit is dependent on the spacing s between the slant plates. When $s = \lambda/2$, a second-order mode of propagation is possible between plates, and the normal delay-path relationships no longer apply.

11.9. Slant-plate Lens with Pressure-operated Microphone

M. A. Clarke[6] in 1953 constructed and investigated a lens-horn combination similar to that shown in Fig. 11.27. A plano-convex lens of the slant-plate delay type, having a diameter of 29 in. and a

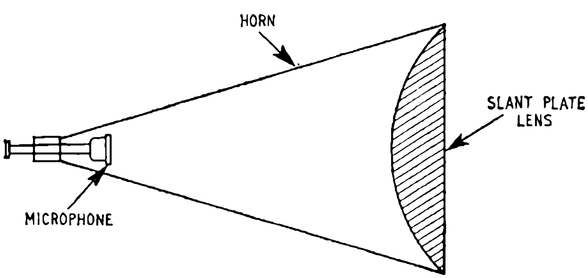


Fig. 11.27.—Lens-horn combination of M. A. Clarke

focal length of 30 in., was used to concentrate sound energy on a pressure-operated microphone placed at the apex of the horn which coincided with the focus of the lens. The mounting which held the microphone was adjustable and allowed it to be moved through the focal " plane " towards or away from the lens.

Clarke claimed that while the directional pattern of the combination was very similar to that of a parabolic reflector system of equivalent diameter, the side and rear lobes were attenuated because

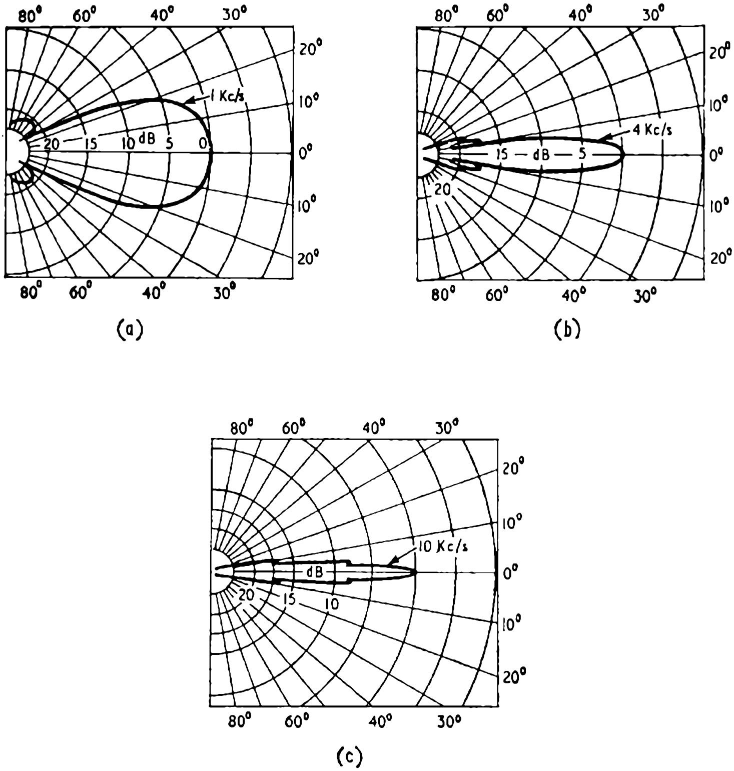


Fig. 11.28.—Directional characteristic of a slant-plate lens:—focal length 30in., aperture 30 in., at three frequencies: (a) 1 kc/s; (b) 4 kc/s; and (c) 10 kc/s (after M. A. Clarke)

the horn reduced diffraction around the lens to a minimum. He also pointed out that unlike the paraboloid combination, the microphone was arranged to face the sound source and hence the housing of the microphone did not block the central portion of the sound beam.

The directional characteristics of Clarke's lens-horn combination are shown for three frequencies in Fig. 11.28 (a), (b) and (c), and were obtained by substituting a small loudspeaker for the microphone and using the lens and horn as a radiator.

The 1,000 c/s characteristic is smooth and the main lobe is wide enough for most practical purposes. At this frequency the sound source can be moved 15° on either side of the normal axis of incidence for a loss of only 3 dB. At 4,000 c/s however, a movement of only 3·5° on either side of the normal axis of incidence reduces the output by the same amount, i.e. 3 dB. At 10 c/s the combination is even more directional, and an angular shift of 1·5° reduces the power output by a half. Unless the sound source is fixed and subtends a very narrow angle, a large measure of defocusing is obviously necessary. Moving the microphone away from the focal plane towards the lens increases the width of the main lobe, and although gain is reduced, the directional characteristic becomes less dependent on frequency.

Internal reflections in a lens-horn system set a practical limit to the refractive index of the lens. At high frequencies, sounds are reflected from the diaphragm and surround of the microphone, and if the refractive index of the lens is high, the abrupt change in impedance results in partial reflections at the lens surface. It is thus possible to set up a standing-wave system involving the microphone housing and the lens, which produces variations in the output. If however the refractive index of the lens is 1·2 or less, the resulting variations in the axial response are not likely to exceed 1·5 dB.

11.10. Higher-order Uni-directional Pressure-gradient Microphones

In Chapter 10, in the discussion on the properties of noise-cancelling microphones, it was shown that a second-order gradient microphone, produced by connecting in opposition two suitably spaced first-order units, has a directional characteristic which is a $\cos^2$ curve in the shape of a narrow figure-of-eight. The polar response is therefore more directional than that of a first-order microphone and within limits, second-order microphones can be used to pick up sounds coming from a greater distance than normal. With this property in mind, it is intended in this section to discuss certain second-order microphones not already covered in the preceding chapter.

As we have already seen, the directional characteristic obtained by connecting in opposition two bi-directional pressure-gradient microphones is also bi-directional and this may lead to operational difficulties in circumstances where a uni-directional characteristic would be more appropriate. If in place of the bi-directional units two microphones having cardioid characteristics are used, the polar response of the combination is uni-directional, and with a directivity greater than that of one unit acting alone.

In Fig. 11.29 (a) the two uni-directional microphones are assumed to be identical in response and sensitivity and to have a directional characteristic of the form $(1 + \cos\theta)$. The units are connected in opposition and so arranged that the distance d' between their acoustic centres is small compared with the wavelength.

11.10.1. AXIAL RESPONSE

For a plane wave incident at $\theta = 0°$, the output of unit A is represented in Fig. 11.29 (b) by the vector V_A and the output of

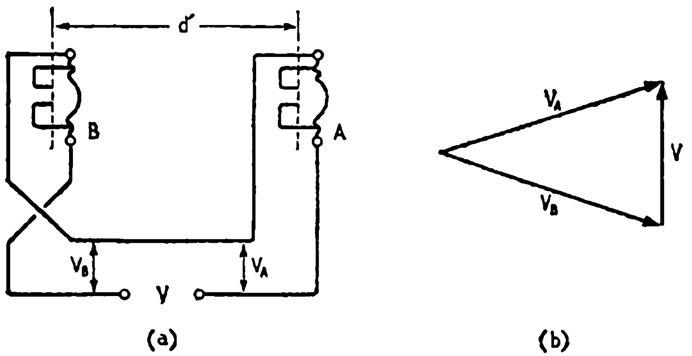


Fig. 11.29.—(a) Schematic view of a second-order pressure-gradient uni-directional microphone. (b) Vector diagram of second-order pressure-gradient microphone

unit B by the vector V_B. The axial output V is the vector difference of V_A and V_B and is given by

$$\begin{aligned} V &= V_A - V_B \\ &= 2V_A \sin kd'/2 \end{aligned}$$

since $V_A = V_B$.

If d' is small compared with the wavelength, then

$$\begin{aligned} V &\simeq V_A \, kd' \\ &= V_A \, 2\pi \frac{f}{c} d' \end{aligned}$$

The equation shows that if the response of V_A and V_B is independent of frequency, then

$$V \propto f$$

If a uniform response is required, some form of electrical equalisation is necessary. To be satisfactory, the equaliser must be capable of introducing, at low frequency, a loss of 6 dB per octave. If however the microphone is required to operate at frequencies where d' is comparable with the wavelength, then the output is no longer proportional to frequency, for the approximation

$$\sin kd'/2 = kd'/2$$

is no longer valid. Over a wide frequency range the axial output is given by

$$V = 2V_A \sin kd'/2$$
$$= 2V_A \sin \frac{\pi d'}{\lambda}$$

Hence $V \propto \sin \dfrac{\pi d'}{\lambda}$.

The response of the microphone as a function of the ratio d'/λ is shown in Fig. 11.30. At small values of the ratio d'/λ, the sensitivity is poor but increases with frequency at the rate of 6 dB per octave. The rate of increase gradually diminishes as the maximum value $2V_A$ is approached. This value is attained when $d'/\lambda = \frac{1}{2}$, i.e., when $\sin \pi d'/\lambda = 1$. The upper limit of the useful frequency range is therefore dependent on the spacing between the units. The first

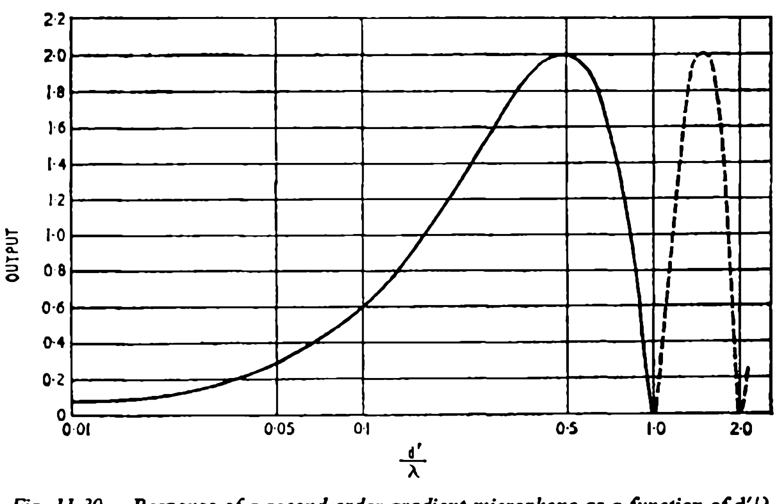


Fig. 11.30.—Response of a second-order gradient microphone as a function of d'/λ

extinction frequency f_{x1} occurs when $f_{x1} = c/d'$. It will be shown later that factors other than sensitivity may affect the choice of the upper frequency limit f_c and set the value at a frequency well below that of f_{x1}. For a practical microphone f_c is likely to lie between the values $c/4d'$ and $c/2d'$. If the spacing d' between the units is reduced in order to achieve a high upper frequency limit, then the sensitivity of the microphone will be adversely affected at low frequencies.

11.10.2. THE DIRECTIONAL CHARACTERISTIC

We have shown that for a plane wave at $\theta = 0°$ the output V of the microphone is given by

$$V = 2V_A \sin kd'/2$$

and if d' is small compared to λ, then

$$V \simeq V_A \, kd'$$

If a plane wave is incident on the combination at θ, the spacing d' between the units is effectively reduced from d' to $d' \cos \theta$. Furthermore, the outputs V_A and V_B of each individual unit alter

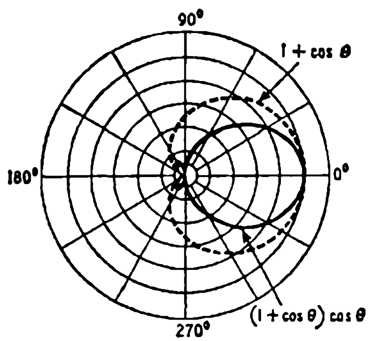


Fig. 11.31.—Directional characteristic of first-order and second-order pressure-gradient uni-directional microphones

in accordance with its directional characteristic, i.e., V_A becomes $V_A (1 + \cos \theta)$ and V_B becomes $V_B (1 + \cos \theta)$. Hence the output V_θ of the combination at any angle θ is

$$V_\theta = kV_A (1 + \cos \theta) d' \cos \theta$$

$$= 2\pi \frac{f}{c} V_A \, d' (1 + \cos \theta) \cos \theta$$

The directional characteristic is of the form $(1 + \cos \theta) \cos \theta$ and this has been plotted for various values of θ in Fig. 11.31, where it can be compared with the normal cardioid characteristic $(1 + \cos \theta)$. Apart from the two small lobes in the second and third quadrants, the characteristic is uni-directional and not as broad as the $(1 + \cos \theta)$ characteristic. The angular shift to the half-power points is of the order of $\pm 40°$ compared with the $\pm 53°$ required to produce a loss of 3 dB in the conventional cardioid characteristic. It must be stressed that a polar response of this shape is only obtained when d' is small compared with λ, that is, when the approximation $\sin kd'/2 = kd'/2$ is valid.

11.10.3. DIRECTIONAL CHARACTERISTIC WHEN d' IS COMPARABLE WITH λ

When d' is comparable with λ, the axial response is given by

$$V = 2V_A \sin kd'/2$$

Hence the response V_θ for a plane wave at angle θ is

$$\begin{aligned} V_\theta &= 2V_A (1 + \cos \theta) \sin \left(\frac{kd'}{2} \cos \theta \right) \\ &= 2V_A (1 + \cos \theta) \sin \left(\frac{\pi d'}{\lambda} \cos \theta \right) \end{aligned}$$

The directional characteristic over a wide band of frequencies is therefore dependent not only on the angle of incidence θ but also on the value of the ratio d'/λ. Fig. 11.32 shows the shape of the characteristic for four values of this ratio.

The curves show that the directivity on the normal axis of incidence is reduced if the microphone operates above $d'/\lambda = \frac{1}{2}$ and is even negative when $d'/\lambda = 1$. Hence the spacing between the units affects both the directivity and the response. If an adequate output at low frequency is required, associated with a high order of directivity, the frequency range of the microphone must be restricted, and in a practical microphone the frequency range is of the order of 25 to 1.

There are however several ways in which the frequency range can be extended. For example, two second-order gradient microphones can be used, each covering 3½ to 4 octaves, one with a relatively large distance between the units to cover the low frequencies and the

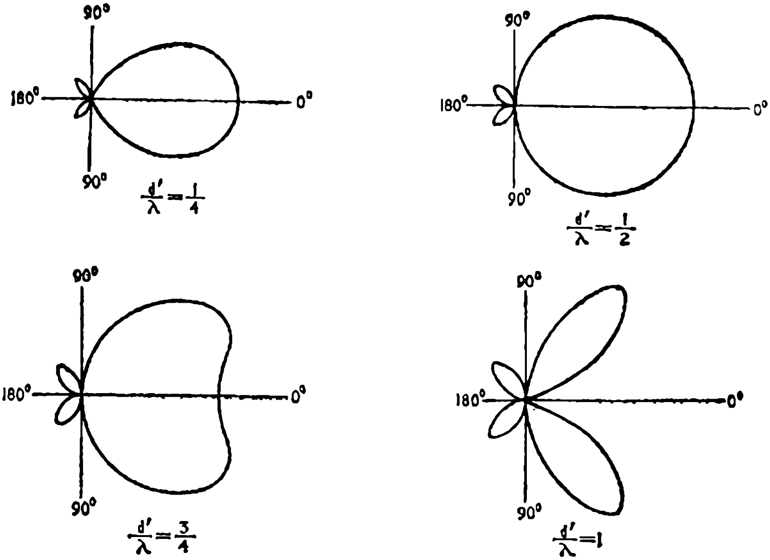


Fig. 11.32.—Showing the change in the shape of the directional characteristic of a second-order pressure-gradient microphone for various values of d'/λ

other with a small distance between the units to cover the high-frequency range. A cross-over network is then required in order to achieve a smooth transition from one unit to the other. Alternatively, a second-order gradient microphone directional at low frequencies can be combined with a line or dimensional microphone designed to cover the high frequencies. Experimental combinations of this type have shown some promise, but they are understandably cumbersome and as yet have not the operational versatility of the conventional microphone and sound boom, which they might hope to displace in television or film studios.

11.11. Bi-gradient Uni-axial Microphones

An alternative to the above systems was the microphone produced by H. F. Olson, J. Preston and J. C. Bleazey[7] in 1956: they described it as a bi-gradient uni-axial microphone. It is a second-order gradient microphone consisting of two phase-shift ribbon microphones, each with a polar response of the form $(0{\cdot}3 + 0{\cdot}7 \cos \theta/3 \cos \theta)$ which is more directional than the normal cardioid.

Fig. 11.33 is a schematic drawing of the microphone. The two phase-shift microphones are located in a tubular metal housing just under 12 in. long, and are separated by a slotted central section called the connector screen. The acoustic separation d' between the units is about 5½ in. This spacing is large enough to give an adequate output at low frequency, and yet allow the combination to operate as a second-order gradient microphone with high directivity up to frequencies in the region of 1,500 c/s. Beyond this frequency, the output of the rear unit of the combination is gradually attenuated and at 4,000 c/s and above the rear unit is completely inoperative. The microphone output is then obtained wholly from the front unit which operates as a normal phase-shift microphone relying on diffraction effects for its directivity.

Following the procedure already used and assuming that the front and rear units have identical sensitivity and response, the axial output voltage V of the combination is given by

$$V = 2V_A \sin kd'/2$$

where V_A is the output of one unit and d' is the spacing between the units, i.e.,

$$V \propto \sin \frac{\pi f d'}{c}$$

Some form of electrical equalisation is therefore necessary if a response independent of frequency is required. The equalisation is provided by a series circuit shunted across the output terminals of the combination (Fig. 11.34).

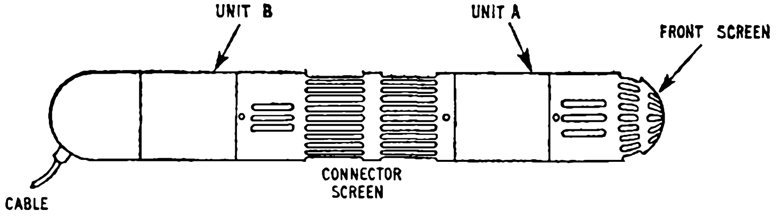


Fig. 11.33.—Schematic view of a bi-gradient uni-axial microphone (H. F. Olson, J. Preston and J. C. Bleazey)

If the directional characteristic of the individual units is of the form

$$(0{\cdot}3 + 0{\cdot}7 \cos\theta/3 \cos\theta)$$

then at low frequency the output voltage V_θ at any angle θ is given by

$$V_\theta = 2V_A\left(0{\cdot}3 + 0{\cdot}7\,\frac{\cos\theta}{3}\cos\theta\right) d' \cos\theta$$

The directional characteristic which is shown in Fig. 11.35 by the full-line curve, exhibits a slightly higher order of directivity than that

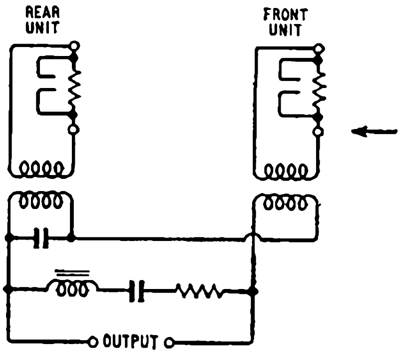


Fig. 11.34.—Showing the electrical correction associated with the bi-gradient uni-axial microphone

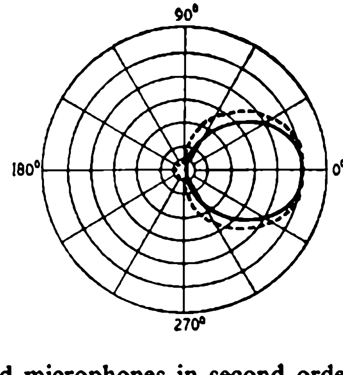


Fig. 11.35.—Theoretical (full-line curve) and practical (dotted curve) polar response of bi-gradient uni-axial microphone (after H. F. Olson, J. Preston and J. C. Bleazey)

obtained by combining two cardioid microphones in second order operation. The dotted curve in Fig. 11.35 is the polar response of the microphone at $d'/\lambda \simeq \frac{1}{2}$ and even at this quite large value of d'/λ the characteristic compares favourably with the theoretical low-frequency characteristic. At high frequency the axial directivity would be negative but a capacitor shunted across the secondary of the transformer associated with the rear unit (Fig. 11.34) reduces the contribution from this source and permits the microphone to function in a first-order condition, directivity being maintained by diffraction effects.

The response of the bi-gradient uni-axial microphone to random sounds is $\frac{1}{9}$ that of an omni-directional microphone and its working

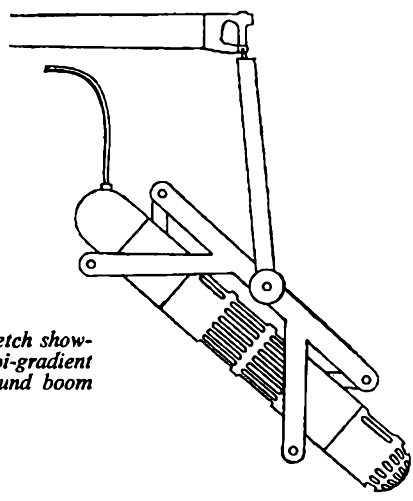


Fig. 11.36.—Diagrammatic sketch showing the method of mounting the bi-gradient uni-axial microphone on a sound boom

distance is therefore three times larger than that of an omni-directional microphone of comparable sensitivity. The increased working distance is not large enough to ensure that the microphone is always " out of shot " in television productions, but its comparatively small size and weight allow it to be mounted on a conventional sound boom where its small angle of pick-up and its ability to discriminate against unwanted noise can be exploited (Fig. 11.36).

11.12. Bi-directional Units in a Uni-directional Combination

An alternative method of obtaining a uni-directional characteristic using two bi-directional units is illustrated in Fig. 11.37 (a). The two units, A and B, are matched in both sensitivity and response and are connected in opposition through an electrical phase-shifting network

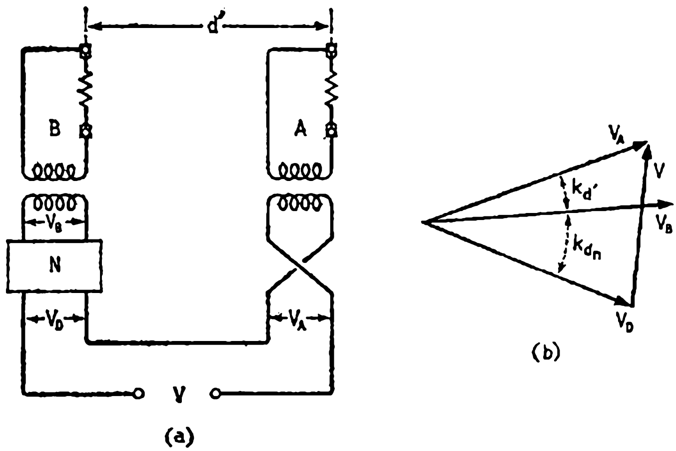


Fig. 11.37.—(a) Two directional pressure-gradient microphones combined through a delay network N to give a second-order uni-directional characteristic. (b) Vector diagram of the bi-directional units in second-order operation

N. The phase shift introduced by the network is equal to that of an acoustic path of length d_n, i.e., kd_n radians.

An expression for the axial response of the combination acting as a second-order gradient microphone may be obtained from the vector diagram of Fig. 11.37 (b). V_A and V_B represent in magnitude and phase the outputs of the units A and B. The output V_B of the rear unit lags on V_A by kd' radians where d' is the acoustic separation between the units. V_D, the output of the electrical phase-shifting network, is obtained from the rear unit by introducing an additional phase shift of kd_n radians, so that V_D lags on V_B by kd_n, and on V_A by k (d' + d_n) radians.

Assuming V_A, V_B and V_D are equal in amplitude and differ only in phase, the output V of the microphone is given by

$$\begin{aligned} V &= V_A - V_D \\ &= 2V_A \sin k \frac{(d_n + d')}{2} \\ &= V_A \, k \, (d_n + d') \end{aligned}$$

for those frequencies where d' and d_n are small compared with the wavelength.

For a plane wave at angle θ, the outputs of the two units are reduced from V_A and V_B to $V_A \cos\theta$ and $V_B \cos\theta$ respectively, and the effective spacing is reduced from d' to $d' \cos\theta$. The phase shift introduced by the electrical network is unaffected by the change in the angle of incidence. Hence the output V_θ of the microphone at any angle θ is given by

$$V_\theta = V_A\, k\, (d_n + d' \cos\theta) \cos\theta$$

A number of directional characteristics is available from the combination, depending on the magnitude of the phase shift introduced by the electrical network. If $d_n = 0$, the characteristic is the familiar $\cos^2$ curve of Fig. 10.6, associated with the combination of two bi-directional units. If $d_n = d'$, the polar response is given by

$$(1 + \cos\theta) \cos\theta$$

(Fig. 11.31), which has already been discussed in this chapter. Maximum directivity is achieved however, when $3d_n = 5d'$, i.e., when the polar response is of the form

$$(0{\cdot}3 + 0{\cdot}5 \cos\theta) \cos\theta$$

The microphone would pick up only $\frac{1}{8}$ of the random sound energy picked up by an omni-directional microphone of comparable sensitivity.

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12

Noise

WE HAVE SEEN in preceding chapters that high sensitivity and a wide frequency range are conflicting factors in microphone design, and where a microphone is required to cover a frequency range of seven or eight octaves, this can be accomplished only by sacrificing sensitivity. It might be thought that sensitivity is comparatively unimportant in microphones used in high-quality sound-reproducing systems, for by careful design an amplifier could be constructed with a high gain, so that a weak signal from the microphone could be amplified to any desired level. It is true that microphones used in sound-broadcasting systems are relatively insensitive, but there is a lower limit to their sensitivity which is determined by the inherent noise present at normal temperatures in all circuits and in all amplifiers.

Noise in sound-reproducing systems can be classified under two headings: inherent or self-noise, and noise produced by sources external to the amplifier system and therefore associated with environment or with a particular location. It is this external or environmental noise with which this chapter is chiefly concerned and only a brief mention will be made of self-noise and the sources which produce it.

12.1. Self-noise

Self-noise voltages are generated by the movements of the free electrons which are present in all conductors or conducting materials. The movements of the electrons in the material are random, but the velocity attained by the electrons is directly proportional to the absolute temperature of the conducting material. The minute currents which result from the electron movement give rise to fluctuating voltages in the conductors and because these currents are random, the frequency spectrum associated with them and with the e.m.f.s they produce is infinitely wide. Self-generated e.m.f.s of this type occurring in the input circuit of an amplifier are amplified in the successive stages of the amplifier and appear in the output as a form of background noise.

12.1.1. VALVE AND POWER-SUPPLY NOISES

Other noise voltages inherent in amplifying systems are associated with the valves and power supply, and like the self-generated e.m.f.s, are particularly objectionable if they occur in the early stages of the amplifier. If an indirectly heated valve is used without special precautions in the first stage of a high-gain amplifier, hum from the heater circuit at the supply mains frequency may be a significant component in the inherent noise produced by the amplifier.

While the designer can reduce hum and power-supply noise to a minimum, there are other sources of noise over which he can exercise little control. The most important of these is shot noise: this is associated with the anode current of the valves in the amplifier and in particular with the valve used in the first stage. The anode current of the valve, while constant over a period of time, varies slightly from instant to instant, because the electrons which constitute the current have random motion, and the number arriving at the anode varies over short intervals of time. These fluctuations in electron flow are equivalent to minute alternating currents superimposed on the mean d.c. anode current, and because they are random, the voltages produced have an infinite frequency spectrum and are heard as noise in the output of the amplifier.

12.1.2. MAGNITUDE OF SELF-NOISE

The actual magnitudes of the noise voltages in a sound-reproducing system with reference to a bi-directional ribbon microphone are given by Olson.[1] The microphone is assumed to have a sensitivity of 600 $\mu V/dyn/cm^2$ at the 250 Ω output terminals of the microphone transformer. The input transformer of the amplifier has a turns ratio of 1:60 and so raises the ribbon impedance to 15,000 Ω at the grid of the first valve. The noise level in the studio is assumed to be only 10 dB above the threshold of audibility and all noise voltages are referred to the 15,000 Ω impedance of the grid circuit.

1. Ambient noise in the studio: 5 μV
2. Self-noise due to thermal agitation of the electrons in the ribbon: 3·5 μV
3. Shot noise associated with the first-stage valve: 1·4 μV
4. Noise due to thermal agitation of the air molecules in the studio: 2·5 μV

If a good signal-to-noise ratio is required, the output of the microphone must be large in comparison with the total self-noise generated in the first stage of the amplifier and its associated circuits. Only noise in the first stage need usually be considered, since in later stages the signals from the microphone have been amplified and are

Table 12.1. WEIGHTED NOISE

Type of microphone	*Mid-band sound equivalent of weighted noise relative to* 2×10^{-4} *dyn/cm*^{2}
Pressure-gradient ribbon	+ 23 dB
Pressure-operated moving-coil	+ 21 dB
Uni-directional electrostatic	+ 26 dB

generally much greater than any noise likely to be generated in the succeeding stages of the amplifier.

To facilitate comparison between the signal-to-noise ratio of microphones having different sensitivities, it is customary to express the noise in terms of the average sound pressures in the mid-band region (i.e., between 300 and 3,000 c/s) which would produce an equal r.m.s. output. The sound pressures are usually given in dB with reference to 2×10^{-4} dyn/cm^2. To allow for variations in aural sensitivity, the noise levels are commonly weighted. This weighting is necessary to provide a basis of comparison with microphones such as the electrostatic and piezo-electric types in which much of the noise occurs at low frequency and is therefore less audible than noise where the energy is uniformly distributed over the frequency spectrum.

Table 12.1 shows the weighted noise, expressed in the manner described, of three microphones, a pressure-gradient ribbon microphone, a pressure-operated microphone and an electrostatic microphone.

12.2. External Noise

Architects and engineers take great care in the design and location of television and broadcasting studios in an attempt to reduce external noise to a minimum but on outside broadcasts the microphone may be subject to noises over which the operational team has little or no control. These noises are not necessarily acoustical in origin but may be produced by a variety of sources: for example, heavy traffic or unbalanced rotating machinery can set up structural vibrations which, if conveyed to the microphone, will lower the signal-to-noise ratio to an extent depending on the microphone in use and on the degree of isolation offered by its shock-absorbing mounting.

12.2.1. MECHANICAL VIBRATION

Certain microphones are more susceptible to structure-transmitted vibrations than others; among the most seriously affected are microphones with mass-controlled mechanical systems and the least susceptible are the compliance- or stiffness-controlled types. The reason for this is obvious if the similarity between a mass controlled microphone and a seismic system is appreciated.

Seismic systems are used to measure absolute displacements or vibrations, and consist basically of a heavy mass suspended on a flexible mounting from a rigid framework (Fig. 12.1). The frame is fixed firmly to the vibrating body or structure and as long as the frequency of the vibrations is higher than the natural frequency of the seismic system, the mass remains stationary in space while the

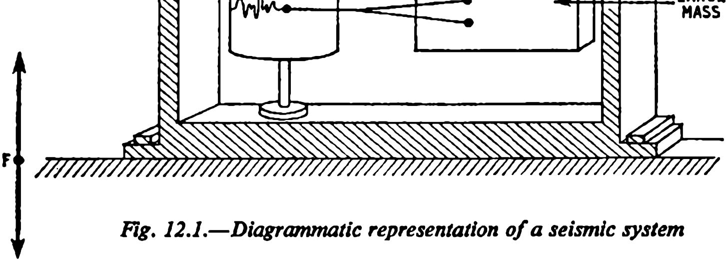

Fig. 12.1.—Diagrammatic representation of a seismic system

frame follows the movements of the structure. The mass carries a pen or stylus, which records on a rotating drum fixed to the frame the displacement waveforms.

Seismographs, used for recording earthquakes or the vibrations of the earth's crust, must have a very low natural frequency, of the order of $\frac{1}{25}$ c/s. Since the vibrations are recorded mechanically, very large masses (20,000 kg) are used to ensure that the mass reactance of the system is large in comparison with the frictional forces inherent in the system of levers which record the displacements.

When a ribbon microphone is subject to a mechanical vibration in a direction normal to the plane of the ribbon, the latter, like the seismic mass, tends to remain stationary in space provided the frequency of the vibrations is higher than the natural frequency of the ribbon system. The magnets and pole-pieces are therefore displaced relative to the ribbon, and a noise voltage is induced proportional to the velocity of a mechanical vibration.

12.2.2. DIRECTIONAL CHARACTERISTIC AND SUSCEPTIBILITY TO MECHANICAL SHOCK

Cardioid microphones which employ a pressure-gradient unit in combination with a pressure unit to obtain a uni-directional characteristic are prone to mechanical shock in the cardioid or bi-directional condition. The shock voltages generated in the mass-controlled bi-directional unit are often referred to as " ribbon

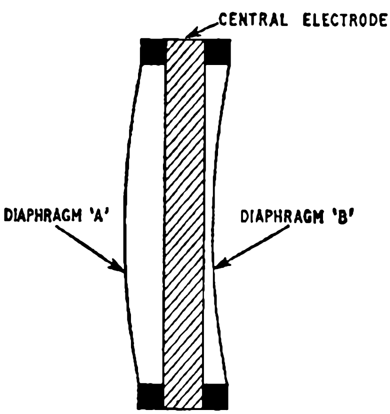

Fig. 12.2.—Diaphragms displaced as a result of acceleration

rumble ". In the omni-directional condition, that is, with the pressure-gradient unit inoperative, the microphone's susceptibility to shock is at a minimum.

Although twin-diaphragm electrostatic microphones have resistance-controlled diaphragm systems, they too are to some extent susceptible to mechanical shock, for the mass of the diaphragm is not negligible. Their susceptibility to shock varies very markedly with the directional characteristic in use.

If a microphone of the twin-diaphragm type experiences an acceleration from left to right, as indicated by the arrow in Fig. 12.2, the diaphragms are displaced relative to the central electrode. The shock voltage e_a produced by the displacement of diaphragm A away from the central electrode is equal in magnitude to, but 180° out of phase with, the shock voltage e_b produced by the displacement of diaphragm B toward the electrode.

In the omni-directional condition both diaphragms are electrically operative and are connected in parallel (Section 8.3.4). The total shock voltage e_s is therefore:

$$
\begin{aligned}
e_s &= e_a + (-e_b) \\
&= 0
\end{aligned}
$$

Hence in the omni-directional condition the microphone is resistant to mechanical shock.

In the cardioid condition, although both diaphragms are displaced by the acceleration, only one diaphragm is electrically operative and no out-of-phase voltage is available to cancel the voltage produced by the mechanical movement. Hence the shock voltage e_s is

$$e_s = e_a \quad \text{or} \quad e_s = -e_b$$

depending on the electrically operative diaphragm.

In the bi-directional condition, both diaphragms are operative but are electrically connected in opposition. The output of the microphone is the difference of the e.m.f.s produced by diaphragms A and B. Hence the shock voltage e_s is given by

$$
\begin{aligned}
e_s &= e_a - (-e_b) \\
&= 2e_a
\end{aligned}
$$

assuming $e_a = e_b$.

Thus in the bi-directional condition, twin-diaphragm electrostatic microphones are most susceptible to mechanical vibration and shock.

Microphones which are least susceptible to shock are those with diaphragms whose mass is small in comparison with the stiffness of their mechanical systems: that is, compliance-controlled microphones. That is why electrostatic microphones are used where severe vibrations are likely to be encountered.

When a microphone is subject to vibrational accelerations in a plane normal to the diaphragm, a force is exerted on the diaphragm which is proportional to its mass and to the acceleration. If m is the mass per unit area of the diaphragm and α is the r.m.s. value of the acceleration, then the force per unit area on the diaphragm is given by

$$F = m\alpha$$

If the sensitivity of the microphone in $\mu V/dyn/cm^2$ is known, an indication of the noise level produced by the mechanical vibrations can be obtained.

Rule, Suellentrop and Perls[2] used two pressure-operated electrostatic microphones to measure low-intensity sounds in the presence of severe mechanical vibrations. The microphones were connected in parallel and arranged in such a way that the vibrations transmitted to each unit produced equal and opposite vibrational voltages which

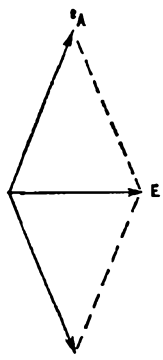

Fig. 12.3.—Vector diagram showing the output of two spaced pressure units when d′ *is comparable with* λ

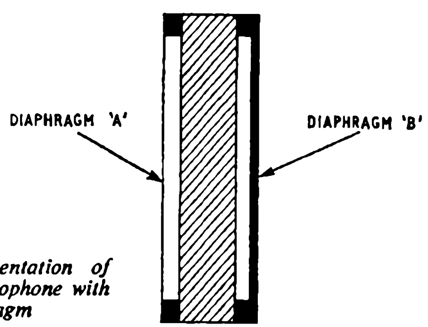

Fig. 12.4.—Diagrammatic representation of pressure-operated electrostatic microphone with vibration-cancelling diaphragm

cancelled each other. This will be recognised as an application of the vibration-cancelling property of the twin-diaphragm microphone which has already been discussed.

When microphones are arranged in this way to reduce vibrational voltages, the sensitivity and response of the combination to acoustic pressures may be affected at high frequency, especially if the spacing

d' between the units is comparable with the wavelength of the sound to be measured.

Since the units are connected in parallel but spaced by a distance d', the output of the combination is the vector sum of the outputs of the units A and B. If the units have equal sensitivity, the combined output E can be obtained from Fig. 12.3 and is given by

$$E = 2e_A \cos \frac{kd'}{2}$$

where e_A is the output of one diaphragm or one unit.

To avoid the reduction in sensitivity occurring when d' is comparable with $\lambda/2$, a twin-diaphragm microphone (shown diagrammatically in Fig. 12.4) is used. Although the diaphragms have equal vibrational sensitivity, only one responds to acoustic pressures. This is achieved by arranging for both diaphragms to have the same natural frequency but at the same time making the mass of one diaphragm many times greater than the other. If m and C_m are the mass and compliance of diaphragm A and if the mass of diaphragm B is nm, then if both diaphragms are to have the same natural frequency, the compliance of diaphragm B must be decreased in proportion to the increase in its mass, i.e., the compliance of diaphragm B is C_m/n. The natural frequency f_{oA} of diaphragm A is given by

$$f_{oA} = \left(\frac{1}{mC_m}\right)^{\frac{1}{2}}$$

The natural frequency f_{oB} of diaphragm B is given by

$$\begin{aligned} f_{oB} &= \left(\frac{n}{nmC_m}\right)^{\frac{1}{2}} \\ &= \left(\frac{1}{mC_m}\right)^{\frac{1}{2}} \\ &= f_{oA} \end{aligned}$$

12.2.3. DISPLACEMENT OF DIAPHRAGMS AS A RESULT OF VIBRATIONAL ACCELERATION

If the r.m.s. value of the acceleration is α, then the vibrational force acting on diaphragm A is given by

$$F_a = m\alpha$$

where m is the mass of diaphragm A.

The displacement x_a of diaphragm A is given by

$$x_a = \frac{F_a}{\omega Z_{ma}}$$

where Z_{ma} is the mechanical impedance of diaphragm A. Since the diaphragm is compliance-controlled,

$$Z_{ma} = \frac{1}{\omega C_m}$$

hence

$$x_a = \frac{F_a}{\omega/\omega C_m}$$

$$= F_a C_m$$

$$= m\alpha C_m$$

If the mass of diaphragm B is nm and it is also subject to the same acceleration α, then the force F_b acting on the diaphragm is given by

$$F_b = nm\alpha$$

and the displacement x_b is

$$x_b = \frac{F_b}{\omega Z_{mb}}$$

If the diaphragm is compliance-controlled,

$$Z_{mb} = \frac{n}{\omega C_m}$$

hence

$$x_b = \frac{F_b}{\omega n/\omega C_m}$$

$$= \frac{F_b C_m}{n}$$

$$= \frac{nm\alpha C_m}{n}$$

$$= m\alpha C_m$$

The displacements of the diaphragms are equal, hence the voltages produced by mechanical vibration are also equal but are 180° out of phase, since one diaphragm is displaced towards the central electrode and the other away from it. If the diaphragms are connected in parallel, cancellation occurs and the sensitivity of the microphone to mechanical vibration is much reduced.

12.2.4. ACOUSTIC OUTPUT

The acoustic outputs of the diaphragms are proportional to the displacement produced by an acoustic pressure. If P is the acoustic pressure acting on diaphragm A, then the displacement x_a of diaphragm A is given by

$$x_a = \frac{P}{\omega/\omega C_m} = PC_m$$

where C_m is the compliance of diaphragm A.

Applying an acoustic pressure P to diaphragm B produces a deflection x_b given by

$$x_b = \frac{P}{\omega n/\omega C_m} = \frac{1}{n}(PC_m)$$

where $\frac{C_m}{n}$ is the compliance of diaphragm B.

The output of diaphragm B is $1/n$ that of diaphragm A. By making n sufficiently large, diaphragm B can be made insensitive to sound pressures without reducing its noise-cancelling properties. The acoustic output of the microphone is then derived almost entirely from diaphragm A and the loss in sensitivity at high frequency is avoided, since the output is now independent of the spacing between the diaphragms.

12.3. Principles of Vibration-free Mountings

With the exception of the twin-diaphragm electrostatic microphones already mentioned, vibration-cancelling diaphragms are not fitted to microphones used in sound-reproducing systems; instead, it is customary to incorporate some form of vibration-isolating device, either in the microphone stand or in the microphone mounting, or both. Because of the obvious limitations on both space and weight, it is difficult to design a shock-free mounting which will provide adequate isolation under all circumstances. It is well known that the simplest way of preventing vibrations from reaching a sensitive instrument is to interpose between the instrument and the vibrating surroundings a resilient system. This can take the form of a set of helical springs, a rubber cushion or rubber blocks but even when this is done the results are sometimes disappointing, and in some cases, far from isolating the instrument, the resilient system may make it even more susceptible to shock than before.

12.3.1. PASSIVE ISOLATION

Isolation which is intended to prevent vibrations from reaching a sensitive instrument is known as " passive isolation " to distinguish it from the active isolation often applied at the source of troublesome vibrations to prevent their transmission into the surroundings or into the foundations.

The susceptibility of a resilient system to mechanical vibrations can best be explained by reference to its analogous electrical circuit. In the simple mechanical system shown diagrammatically in Fig. 12.5 (a), the microphone or sensitive instrument is represented by the mass *m* and is isolated from the vibrating foundation by the perfect spring *s*, which possesses compliance C_m but no mass or friction.

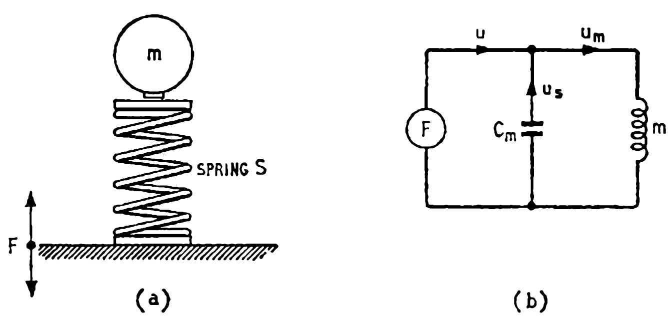

Fig. 12.5.—Passive isolation: (a) a spring used to prevent movement of the foundations being transmitted to the mass m*; (b) equivalent circuit of (a)*

The velocity in the mass or in the spring is equivalent to the current in the analogous branch of the equivalent circuit, shown in Fig. 12.5 (b). If the vibrating foundation applies to the spring a force which is independent of frequency, then the velocity u_s in the spring depends on its compliance reactance $1/\omega C_m$ and the velocity u_m in the mass *m* depends on the mass reactance ωm.

Below the natural frequency f_0 of the mechanical system, the compliance reactance is large and the spring behaves as if it were rigid, transferring the velocity of the foundations to the mass with virtually no loss. The vector diagram in Fig. 12.6 (a) shows the

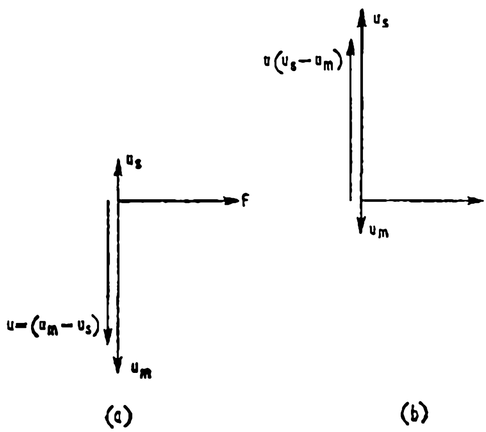

Fig. 12.6.—Vector diagrams showing the magnitude and phase of the velocities of the microphone system of Fig. 12.5: (a) at a frequency below the natural frequency of the system; (b) at a frequency above the natural frequency of the system

magnitude and phase of the velocities for this condition. It will be seen that the velocity of the mass m is large and comparable with the velocity u of the foundations. The resilient system is therefore ineffective in shielding the instrument or microphone from shock. It should be noted that where the isolating effect of the mounting is poor, the ratio $u_m : u \simeq 1$.

At a frequency well above that of the natural frequency of the mechanical system, the compliance reactance of the spring is small but the reactance of the mass is large, hence u_m is small in comparison to u_s and to u, as indicated in Fig. 12.6 (b). Under these conditions the mounting is shock-free and the ratio $u_m : u < 1$.

12.3.2. THE PASSIVE ISOLATION FACTOR

The ratio of the velocity of the mass to the velocity of the foundations is defined as the passive isolation factor i_p and is a measure of the effectiveness of the resilient mounting. In calculating the ratio, the phases of the voltages u_m and u are unimportant and only their magnitude need be considered.

$$|i_p| = \frac{u_m}{u}$$

$$= \frac{u_m}{u_s - u_m}$$

If the vibrational force applied by the foundations to the system is F, then

$$u_m = \frac{F}{\omega m}$$

and

$$u_s = \frac{F}{1/\omega C_m}$$

$$= F\omega C_m$$

Hence

$$|i_p| = \frac{F/\omega m}{F\omega C_m - F/\omega m}$$

$$= \frac{F/\omega m}{F/\omega m\,(\omega^2 m C_m - 1)}$$

$$= \frac{1}{\omega^2 m\, C_m - 1}$$

At the natural frequency f_o of the mechanical system,

$$\omega_o^2 = \frac{1}{mC_m}$$

or

$$mC_m = \frac{1}{\omega_o^2}.$$

Hence

$$i_p = \frac{1}{\omega^2/\omega_o^2 - 1}$$

$$= \frac{1}{f^2/f_o^2 - 1}$$

The variations of i_p as a function of f/f_o are shown in Fig. 12.7. The graph indicates that to be effective, the natural frequency of the

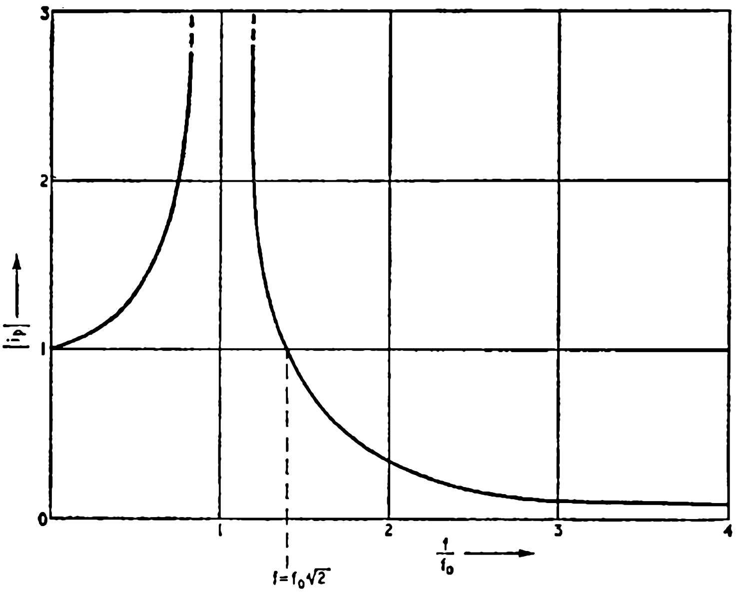

Fig. 12.7.—Variations in the passive isolation factor i_p *plotted against* f/f_0

resilient system must be low and well below the lowest frequency of the interfering vibrations. The mounting only begins to be effective at frequencies above $f_0\ \sqrt{2}$. At frequencies lower than $f_0\ \sqrt{2}$ the resilient system is at a disadvantage, especially if the natural frequency f_0 of the mounting corresponds to or is close to the frequency of the interfering vibrations. In these circumstances it would be better to employ a rigid mounting.

12.3.3. THE EFFECT OF DAMPING ON THE ISOLATION FACTOR

The simple shock-free mounting just described is not likely to be satisfactory in practice for it possesses two serious faults. If the microphone or sensitive instrument is displaced from its mean position by accident or by mechanical shock it will vibrate at the natural frequency of the system and since the mounting is loss-free, it will continue to do so for a long time. Furthermore, if the frequency of an interference vibration is at or near the natural frequency f_0 of the system, the velocity imparted to the mass is very large, and under these conditions a sensitive instrument can be damaged. Although

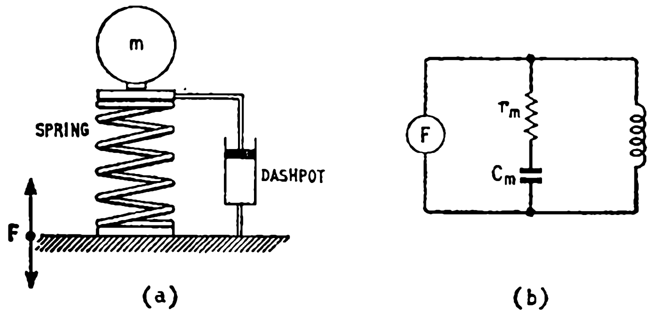

Fig. 12.8.—Passive isolation: (a) damped mechanical system; (b) equivalent electrical circuit of (a)

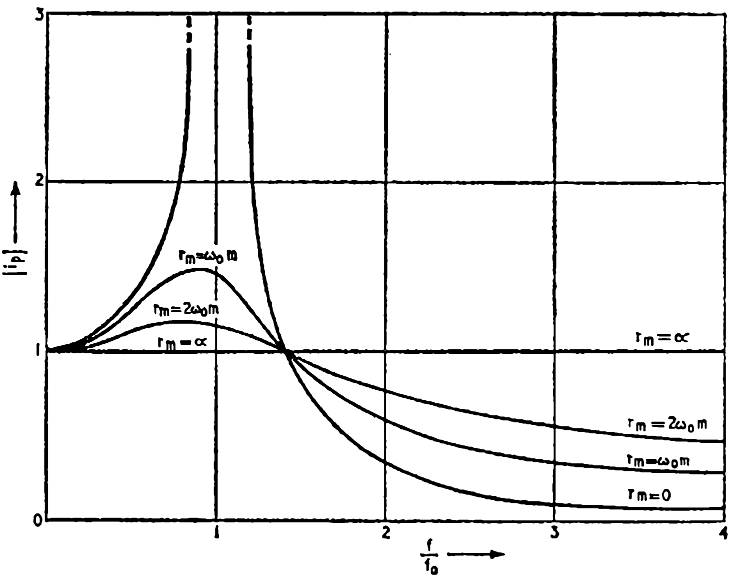

Fig. 12.9.—Isolation factor i_p plotted against f/f_0 for various degrees of damping

there is always some damping in a practical system, it is usually necessary to increase it, either by means of a so-called " shock absorber " or dashpot employing fluid friction, or by replacing the spring by a resilient material which is not loss-free. The most commonly used materials of this type are rubber and certain plastics.

A resilient mounting with auxiliary damping is shown in Fig. 12.8 (a) and its equivalent electrical circuit in Fig. 12.8 (b). If the damping r_m is very large, the system behaves as if the mass were rigidly connected to the foundation, i.e. $r_m \gg 1/\omega C_m$. The mounting is therefore ineffective since $u_m = u$ and the isolation factor $i_p = 1$ at all frequencies. This condition is represented in Fig. 12.9 by the straight line $r_m = \infty$. The effect on the isolation factor for some other values of r_m, defined in terms of $m\omega_o$, is also shown in the diagram. It will be seen that the larger the damping, the greater is the reduction in the resonance peak but the less effective is the mounting as an isolating device.[3] The actual value of the damping employed is a compromise and will depend on an estimate of the conditions likely to be encountered in practice. The curves show that if a high degree of isolation is required with a large value of r_m, the natural frequency of the mechanical system must be well below the interfering frequency region. This is difficult to achieve in certain microphone mountings because of the limitations on space and weight.

12.4. Anti-vibration Mounting for Microphone

Designers deliberately reduce the sizes of microphones to ensure that their dimensions are small compared with the wavelength of audible sounds and as a consequence of the restriction in size, microphones are comparatively light in weight. It is difficult therefore to produce an anti-vibration mounting with sufficient compliance to ensure that the natural frequency of the system is arbitrarily low and will thus provide adequate isolation in all circumstances. Moreover, microphones are used in a variety of conditions and a mounting which is satisfactory in one situation may be quite inadequate in another. In recording or sound-broadcasting studios where the microphone is static, the problems are simpler than in film and television studios.

If the microphone can be suspended, adequate isolation can generally be obtained by the use of rubber cords of appropriate length and thickness but when a floor or table stand is required, a more elaborate form of anti-vibration mounting may be necessary. An anti-vibration unit is designed for use with a pressure-gradient microphone and is intended to be interposed between the microphone and its stand. Two Silentbloc resilient mountings in the form of circular diaphragms (Plate 12.1) isolate the microphone holder from the microphone stand. The use of two resilient diaphragms, as in Fig. 12.10 (a), doubles the stiffness of the mechanical system, hence the compliances C_{m1} and C_{m2} together with

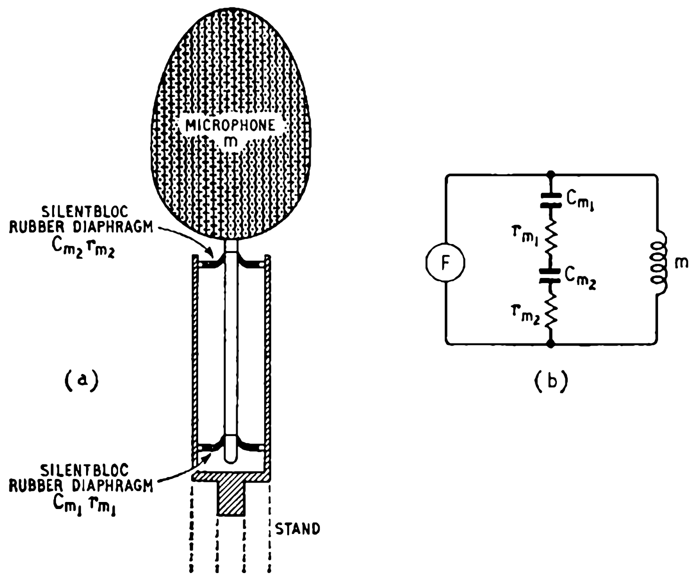

Fig. 12.10—(a) Diagrammatic representation of the anti-vibration unit of Plate 12.1. (b) Equivalent circuit of the unit

their associated resistances r_{m1} and r_{m2} are shown in series in the equivalent circuit of Fig. 12.10 (b).

While an anti-vibration mounting of the type described is generally satisfactory in sound studios, a more elaborate mounting is necessary if a pressure-gradient ribbon microphone is to be used satisfactorily on a sound boom in a television production. The boom is usually telescopic and its shaft is supported on rubber rollers which in the course of time develop flats as a result of uneven wear. During the extension or retraction of the boom, vibrations are produced which can, if the isolation is inadequate, shock-excite the moving system of the microphone.

The combined windshield and anti-vibration mounting of Plate 12.2 is designed primarily for a pressure-gradient ribbon microphone and provides the necessary isolation without interfering with the positive action of the remote turning and tilting controls associated with the sound boom. Because ribbon microphones are susceptible to draughts as well as shock, a rather more elaborate windshield is necessary. The microphone is rigidly fixed to the windshield and their combined mass is represented by m_2 in the simplified mechanical diagram of Fig. 12.11 (a), and in the equivalent electrical circuit of Fig. 12.11 (b). Four rubber mountings shown in Plate 12.2 attach the microphone and windshield to the remote tilting mechanism of the sound boom. Although the four mountings are effectively in series, they are represented for simplicity in the equivalent electrical circuit by a single compliance C_{m2}, r_{m2} in Fig. 12.11 (a) and (b). The remote tilting mechanism, whose mass is represented by m_1, is attached to the

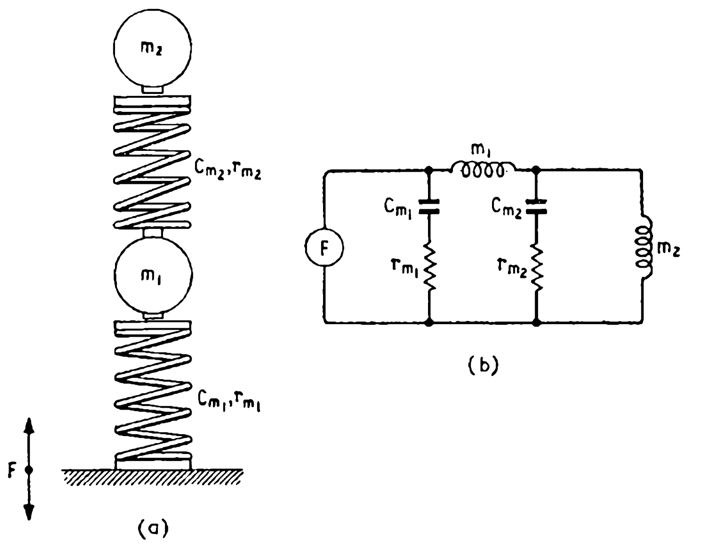

Fig. 12.11.—(a) Simplified mechanical system of Plate 12.2. (b) Equivalent electrical circuit of Plate 12.2

rotating head of the sound boom by two polythene strips in the form of a stirrup whose compliance and friction are shown as C_{m1} and r_{m1}.

The equivalent circuit of Fig. 12.11 (b) shows that the anti-vibration unit is analogous to a low-pass filter terminated, not in its characteristic impedance, but by the mass m_2 of the microphone. Mechanical damping is therefore necessary to avoid undesired resonances and for this reason rubber and polythene are employed as the compliant materials in preference to steel, since their internal damping is high.

12.5. Effect of Winds and Air Streams

Another type of noise associated with environment is that produced by the action of the wind or of an air stream on the microphone. The noise which results is low-pitched and rumbling, varying with the wind speed and with the microphone in use. In general, pressure-gradient microphones, electrostatic as well as electrodynamic, become unusable in winds of only a few miles per hour and if they are subjected to winds in excess of 20 m.p.h., may suffer permanent damage unless specially protected. Compliance-controlled microphones and those with critically damped moving systems, i.e., resistance-controlled types, are to a much lesser extent affected by winds and are not likely to be damaged, even under abnormal conditions. The problem of screening microphones from winds and draughts is one that is frequently encountered when microphones have to be used out of doors; the subject does not appear to have received the attention which its importance merits, there being little published information.

12.5.1. BERNOULLI WINDSHIELD

If the air stream or wind is constant in direction, a windshield can be constructed which takes advantage of the pressures and phase differences acting over the surface of the microphone to reduce the effective wind pressure on the diaphragm. The Bernoulli windshield is an example of a screen designed on this principle and its action is illustrated in Fig. 12.12 (a), (b) and (c).

It is well known that when an air stream is directed against the leading edge of an aircraft's wing, as illustrated in Fig. 12.12 (a), the aerodynamic pressures produced are not uniform in either amplitude or phase over the surface of the aerofoil. The air passing under the wing is slowed down by the action of the wing surface, and the reduction in air-stream velocity causes the particles of the air to crowd closer together, thus increasing the air pressure below the wing. The air flowing over the deeply curved upper surface of the wing follows a path which is longer than that traversed by the air under the wing. In order to preserve the laminar flow of the air stream, the velocity of the air passing over the upper surface of the wing increases. The increase in velocity is accompanied by a reduction in air pressure because of the increased distance between the

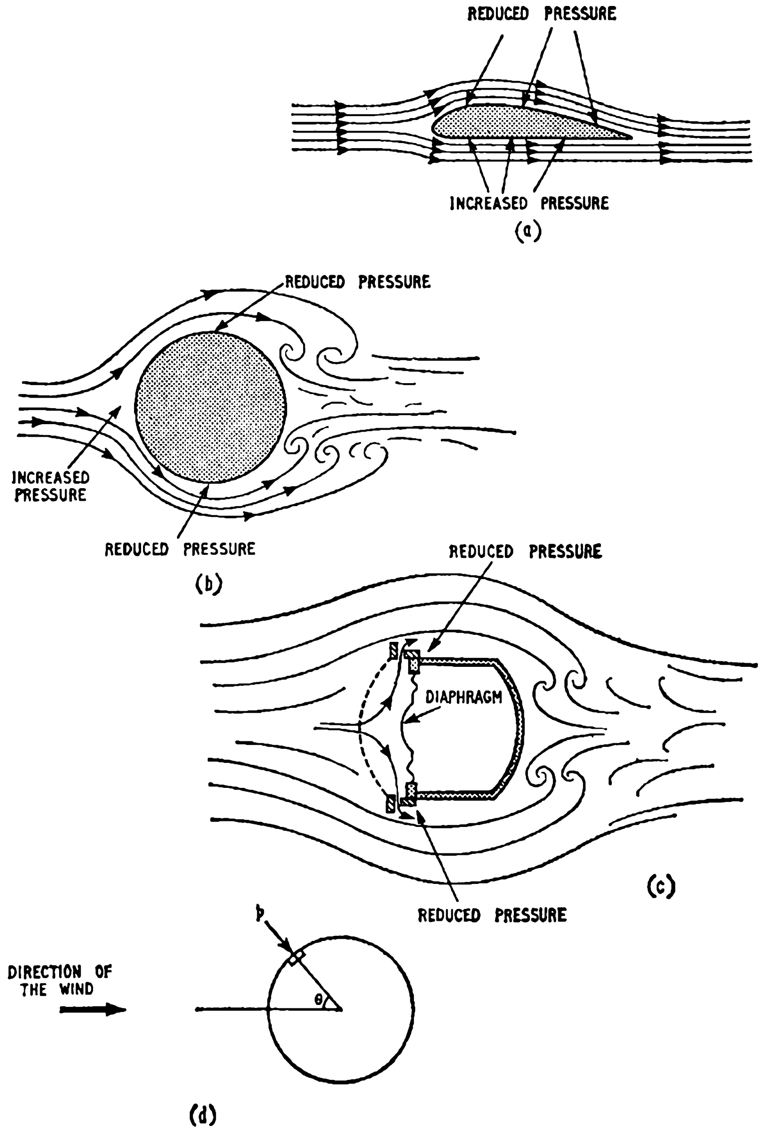

Fig. 12.12.—Aerodynamic pressures (a) on an aerofoil; (b) on a sphere; (c) on a microphone equipped with a Bernoulli windshield; (d) Point on the surface of a sphere relative to the direction of the wind

particles in the fast-moving air stream. The reduction in pressure on the upper surface of the aerofoil is considerable and may contribute about 75% of the total "lift".

It is therefore clear that the position of the pressures and rarefactions on the surface of an obstacle depends on the direction of the wind and the shape of the obstacle. Assuming the shape of the microphone and windshield shown in Fig. 12.12 (d) to be approximately spherical, Phelps[4] derived an expression for the air pressure at any point on the surface of a sphere and showed that if the fluctuations in wind velocity are small and of very low frequency, the pressure p at a point on a sphere depends solely on the position of the point and is independent of the volume or size of the sphere.

If a radius drawn through the point on the surface of the sphere makes an angle θ with the direction of the wind, as in Fig. 12.12 (c), then the pressure p at the point is given by

$$p = a\,(9\cos^2\theta - 5)$$

where a is a constant depending on the mean velocity of the wind and the density of the air.

The equation shows that the pressure is a maximum at the point where the wind is normal to the surface of the sphere, i.e., at $\theta = 0°$, and falls to zero or more correctly to atmospheric pressure when

$$9\cos^2\theta = 5 \quad \text{or} \quad \theta = 41{\cdot}8°$$

At $\theta = 90°$ the pressure is negative, i.e., below atmospheric pressure, and the reduction in pressure at $\theta = 90°$ is greater than the increase in pressure at $\theta = 0°$, in the ratio 5:4.

The Bernoulli windshield is shown in Fig. 12.13 and schematically in Fig. 12.12 (c). The front of the windshield is perforated and covered with silk and the narrow ring which serves to attach the

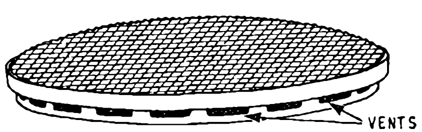

Fig. 12.13.—A Bernoulli windshield showing the vents

shield to the microphone is provided with a number of vents. The vents are also covered with a layer of silk and are arranged to coincide with the areas of reduced pressure on the surface of the microphone case. While the shield has little effect on sounds and acoustic pressures, the wind stream which may penetrate the shield is diverted as shown in Fig. 12.12 (d), so reducing the effective wind pressure on the diaphragm of the microphone. In spite of its small size, a reduction of about 12 dB in wind noise is obtained and this compares favourably with the ellipsoidal windshield used in the late 1930s. It must be remembered that the Bernoulli shield is designed on the assumption that the microphone can be positioned so as to face the wind which is also assumed to be constant in direction. These conditions seldom obtain in practice and if they are not satisfied, the shielding properties of the screen are much reduced. For this and other reasons Bernoulli shields are not now in common use.

Modern microphone windshields consist of a layer of acoustically porous material having a high ratio of resistance to inertance; the material is supported if necessary on a light metal frame and arranged to shield or enclose the diaphragm or sound entrance of the microphone. The shield, in a way not yet fully understood, discriminates against the air stream produced by the wind but allows the acoustic pressures which constitute sound to pass through the shield with negligible attenuation. The ability to discriminate against one form of particle movement in favour of another may be due in part to the vibratory nature of the movement of the particles in the sound field compared with their uni-directional motion in the air stream, and in part to the relatively high velocities attained by the particles in an air stream compared with their low velocities in a sound field. The mean velocity of an air particle in a comparatively light wind of some 10 m.p.h. is of the order of 170 in./sec while the r.m.s. velocity of a particle in a sound field of quite high intensity is of the order of 0·2 in./sec.

The effectiveness of a shield in reducing wind noise while at the same time offering negligible attenuation to sounds has been shown to depend on its size and shape and on the acoustic resistance of the porous material employed. When a sound wave is transmitted through an acoustically porous material such as silk or cotton cloth, energy is lost. The loss is due to viscosity effects in the air and for a given air flow, depends on the material used. It is convenient to regard the conversion of kinetic energy into heat energy as being due to the acoustical resistance r_A of the material.

r_A is defined in acoustical ohms as

$$r_A = \frac{p}{U}$$

where p = the pressure in dynes across the material and
U = the volume current in cm^3/sec.

r_A is a basic property of the acoustical material and is constant, provided that the velocity of the particles in the air stream is comparable with the particle velocities in a typical sound field: that is, less than 1 in./sec. If the velocity of the air stream exceeds 4 in./sec, the flow resistance ceases to be constant and begins to increase. Turbulent flow theory predicts that if complete turbulence is present in the air stream, the increase in flow resistance is proportional to the first power of the particle velocity. Brown and Bolt[5] have verified the theory experimentally and have shown that the flow resistance of

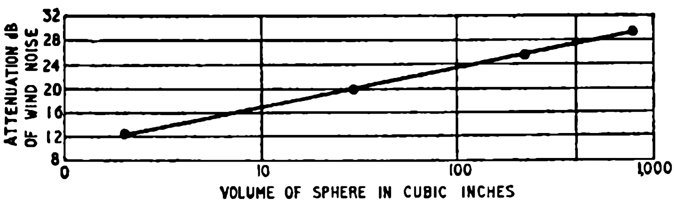

Fig. 12.14.—Spherical attenuation of wind noise in dB plotted against volume of shield (after J. C. Bleazey)

fabrics increases as predicted by turbulent flow theory, provided the velocity of the air stream is in excess of 10 cm/sec, that is, greater than 4 in./sec.

Since the attenuation introduced by a fabric shield is proportional to its flow resistance, it follows that the attenuation must increase directly as the velocity. Taking as a reference velocity $V_2 = 4$ in./sec, the attenuation in dB associated with a wind velocity V_1 is

$$20 \log \frac{V_1}{V_2} \text{ dB}$$

For a wind speed of 170 in./sec (about 10 m.p.h.) the attenuation is

$$20 \log \frac{176}{4} \text{ dB}$$
$$= 30 \text{ dB}$$

An attenuation of this magnitude is not likely to be obtained in practice, unless the volume enclosed by the wind screen is very large.

Bleazey[6] obtained experimentally an expression for the attenuation introduced by a spherical windshield as a function of its volume. He made measurements in a " dead room " under controlled conditions using a wind machine of the paddle type. An unscreened pressure-operated microphone was placed 4 ft from the machine and its response to the wind noise recorded for 3 minutes. The microphone was then equipped with spherical windshields, 2 to 8 in. in diameter, and the test was repeated to find the attenuation introduced by each screen. Tests were also made out of doors, comparing a screened and an unscreened microphone.

It was found the attenuation in dB introduced by the shield, when plotted against the log of the effective volume of the shield, approximates to a straight line (Fig. 12.14). From this, an empirical expression for the effectiveness of the shield is obtained.

$$\text{Attenuation in dB} = 6{\cdot}77 \log V + 10{\cdot}4$$

where V, the effective volume, is measured in cubic inches. The effective volume is the volume of the shield, less the volume of the microphone, or the part of it extending into the shield.

12.5.2. SCREENING MATERIALS

The attenuation introduced by a windshield is affected to a slight extent only by the number of layers of fabric used in its construction. The curve[6] of Fig. 12.15 represents the attenuation for maximum wind-noise peaks of a spherical screen of 2 in. diameter as a function

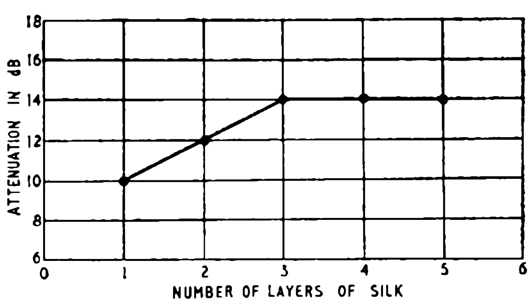

Fig. 12.15.—Attenuation of wind noise as a function of fabric thickness (after J. C. Bleazey)

of the number of layers of silk used. It will be seen that only a slight increase in attenuation is achieved by increasing the layers from one to three. Thereafter, no reduction in wind noise is obtained by increasing the fabric thickness. This suggests that of the total wind noise picked up by a microphone equipped with a shield, only a small part is due to penetration of the shield by air stream particles. The greater part of the noise results from the eddies and turbulences produced by the obstacle effect of the shield itself. Increasing the size of the shield does not increase the aerodynamic pressure per unit area on it but if the diameter of the screen exceeds 6 in., the pitch and frequency of the turbulent tones are likely to be below the pass-band of most microphones and amplifiers. Unfortunately, large windshields are often unacceptable in broadcasting and television production because they are difficult to use and obtrusive in appearance, and small ones are preferred. In adverse weather conditions it is necessary to use electrical equalisation in conjunction with the small shield to ensure that the wind noise is reduced to an acceptable level. If the equalisation introduces progressive bass cut below 200 c/s, speech of acceptable quality can be obtained in wind speeds of up to 40 m.p.h.

Plate 12.3 shows some of the smaller windshields used with various types of studio and outside broadcast microphones. Shield A was used in the late 1930s to screen the diaphragm of a pressure-operated moving-coil microphone. The acoustically porous material consisted of a fabric supported on an open mesh of copper gauze. An attenuation in wind noise of some 12 dB was obtained and this could be improved on if the back of the microphone was arranged to face the wind so that advantage could be taken of the streamlined shape of microphone and windshield. This screen is now obsolete, but is included for comparison with shield B.

Shield B could be described as a modern version of A, since it is intended to screen the diaphragm of a contemporary pressure-operated moving-coil microphone. Instead of acoustically porous fabric, this shield consists of a thick dome-shaped pad of rubberised hair protected by a finely woven wire-mesh screen. The attenuation introduced by shield B is comparable with that of shield A and has the advantage of being much smaller.

12.5.3. EFFECT ON FREQUENCY RESPONSE

Both shields to some small extent alter the frequency response of the microphones (Fig. 12.16), but the frequencies at which the response is affected are determined by the inertance of the porous material used and on the volume of air enclosed by the shield. The equivalent circuit of a typical microphone shield (shown in Fig. 12.17) is in the form of a low-pass filter, the series inductive impedance representing the fabric or rubberised hair, and the shunt capacitance the compliance of the air enclosed by the shield. It should be appreciated that

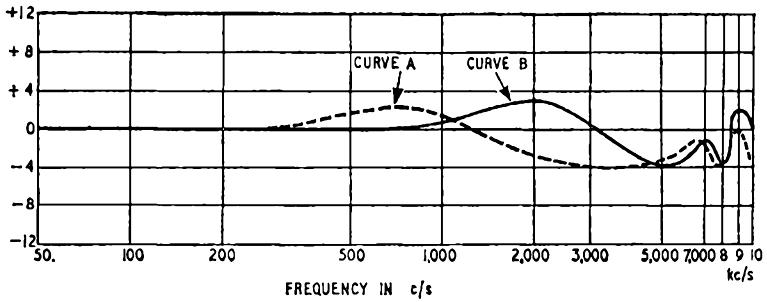

Fig. 12.16.—The effect on the response of the microphone of Shield A and Shield B of Plate 12.3

although pressure-operated moving-coil microphones are substantially resistance-controlled, the filter is not likely to be properly terminated over the operating range of the microphone.

Windshields and fabric materials used to protect the diaphragm and sound ports of the microphone will eventually become blocked if exposed to rain or spray for lengthy periods. The wire-mesh protection screens are often treated with a water-repellent silicone varnish, but if the fabric material becomes sodden as the result of exposure, the response of the microphone is seriously affected.

The three windshields of group C (Plate 12.3) would, if used out of doors, attenuate wind noise to some extent, but are primarily intended for use in studios on occasions when close microphone techniques are employed. Close-range speech or singing subjects a microphone to air streams of high velocity. This is particularly true of the shock waves associated with the explosive consonants " b " or " p ", which produce air particle velocities in excess of 300 in./sec, or 20 m.p.h. When close techniques are employed it should be noted that the air

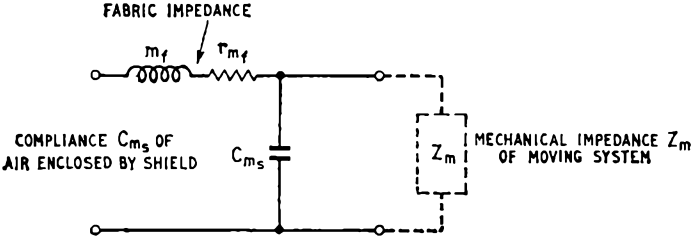

Fig. 12.17.—Equivalent circuit of windshield and microphone

streams from the mouth and nose are much more directional than the acoustic pressure associated with them and the microphone can often be placed to avoid the direct air stream without too great a loss in sound quality. The wind shields of group C use stainless steel gauze to protect and support the fabric material in contrast to shield D which is of plastic, having a large number of holes of very small diameter which provide the necessary acoustic resistance.

12.6. Induction and Radio Interference

On outside broadcasts, microphones are sometimes used in the vicinity of power-supply sub-stations or very close to television monitoring equipment, where they can be subjected to stray magnetic

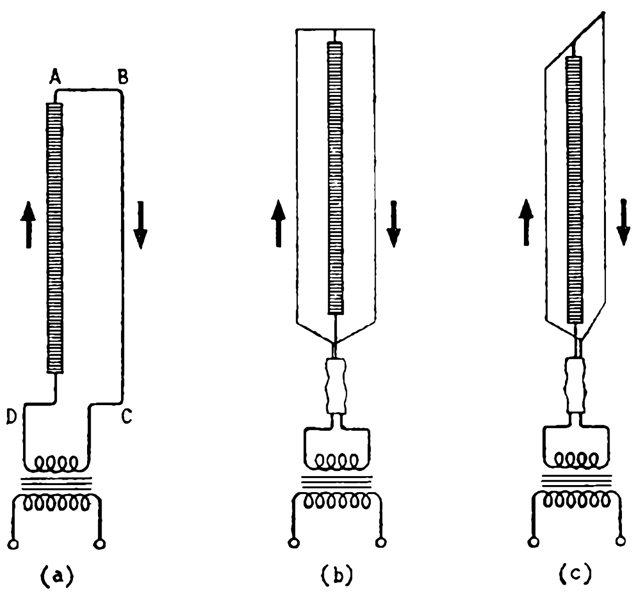

Fig. 12.18.—Hum neutralising connections for ribbon microphone

fields whose strength may be of the order of 2 or 3 gauss. Low-impedance microphones of the pressure-gradient and moving-coil types are particularly susceptible to magnetic induction. The effect of the stray field on the microphone transformer can be minimised by the use of a Mumetal case but it is also necessary to take special precautions with the disposition and arrangements of the leads which connect the ribbon to the primary windings of the transformer.

If the connections are made as in Fig. 12.18 (a), the loop ABCD encloses a considerable area and may link a flux of a magnitude sufficient to cause serious interference. The interference can be reduced by decreasing the size of the loop but this is not always effective because a large part of the stray field tends to be concentrated in or near the pole-pieces and so some pick-up by the loop is inevitable. An alternative method is to use two leads from the top of the ribbon arranged to form two loops of equal area. If the stray field is uniform, the loops each enclose an equal flux and the voltages which result cancel out.

Two possible arrangements for the hum-neutralising loops are shown in Fig. 12.18 (b) and (c). In Fig. 12.18 (b), the loops are located in the same plane as the ribbon or in a plane parallel to it. It is essential that they lie precisely in the same plane, otherwise complete cancellation is not likely to occur. Disposed as in Fig. 12.18 (c), the loops lie in a plane normal to the plane of the ribbon and therefore in the acoustic path. In order to reduce obstruction effects to a minimum it is customary to use uninsulated conductors of a diameter just sufficient to ensure the required rigidity in the loop.

Before mounting the microphone in its case it is subjected to a uniform magnetic field and the wires which form the loops are bent and adjusted to give minimum hum output. A reduction of some 30 dB in hum pick-up can be achieved with balanced conductors compared with the single-conductor arrangement of Fig. 22.18 (a).

High-impedance microphones of the electrostatic type, if imperfectly screened, are susceptible to radio-frequency interference. While the degree of interference depends to some extent on the strength of the radio-frequency field, it is also influenced by the disposition and screening of the microphone leads and can be greatly increased if the length of the metal microphone stand is such that it resonates at the frequency of the interfering field.

Experiments have shown that the interfering signals enter the head amplifier at radio frequency, via the screened microphone lead, and are likely to do so if the lead passes into the case of the amplifier through a hole with more than adequate clearance. The cable within the case may be in the form of a loop or portion of a loop which acts as an inductive coupling. Voltages large enough to cause the valve or valves to operate on the non-linear part of their characteristic can be induced, and any amplitude modulation present in the signal will then appear as audio-frequency interference. The interference can be reduced by as much as 50 dB by the use of a concentric type of gland nut which grips the metallic screen of the cable tightly and at the same time makes a low impedance connection at radio frequency to the case of the amplifier.

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Appendix 1

Simple Harmonic Motion[1] and a Proof of the Relationship $u = j\omega x$

Sound is conveyed through the air by a vibratory movement of the air particles. The displacement of the particles is in the direction of propagation of the " wave " and if the sound is a pure tone, the particles execute simple harmonic motion.

If the maximum amplitude of the displacement is x_m, then the instantaneous displacement x_i is given by:

$$x_i = x_m \sin \omega t$$

where $\omega = 2\pi f$,
f = number of vibrations per second,
t = time in seconds.

The displacement of a particle with reference to time is shown in Fig. A.1.1. The time t is in seconds, but is on a scale marked $\frac{\text{vibrations}}{f}$. This is permissible, since $f = \frac{\text{vibrations}}{t}$, hence $t = \frac{\text{vibrations}}{f}$.

The instantaneous velocity u_i is rate of change of displacement, that is, $\frac{dx_i}{dt}$ and u_i is given by

$$u_i = \frac{dx_i}{dt}$$
$$= \omega x_m \cos \omega t$$

The time at which the velocity is a maximum in either the positive or negative direction can be obtained by differentiating the expression for u_i and equating the result to zero.

Differentiating, we have

$$\frac{du_i}{dt} = -\omega^2 x_m \sin \omega t$$

The slope of this curve is zero at a maximum, and this occurs when

$$\omega^2 x_m \sin \omega t = 0$$

that is, when $\sin \omega t = 0$, or when $\omega t = 0, \pi, 2\pi$, etc.

$$\frac{du_i}{dt} = 0$$

when $t = 0, \frac{1}{2f}, \frac{1}{f}$, etc.

By differentiating once more and then substituting the above values for t in the equation, it is possible to distinguish between the positive and negative maxima. A substitution which indicates that

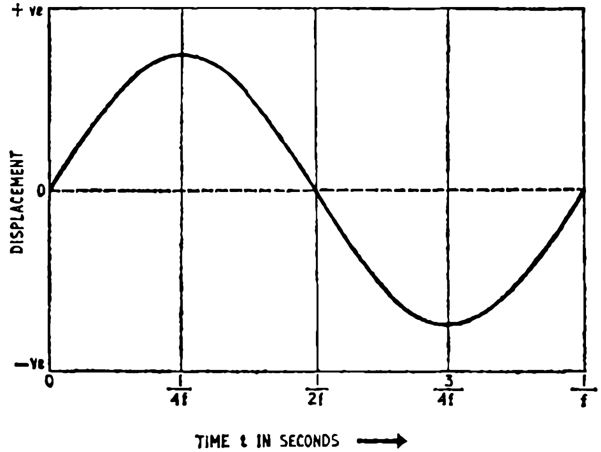


Fig. A.1.1.—Sinusoidal displacement curve: simple harmonic motion

the slope of the velocity curve is negative corresponds to a positive velocity maximum, and conversely, a positive rate of change of slope indicates a negative velocity maximum. Differentiating, we have

$$\frac{d^2u_i}{dt^2} = -\omega^3 x_m \cos \omega t$$

When $t = 0$

$$\frac{d^2u_i}{dt^2} = -\omega^3 x_m \cos 0$$
$$= -\omega^3 x_m$$

This indicates that when $t = 0$, the velocity is a maximum and positive.

When $t = \frac{1}{2f}$

$$\frac{d^2u_i}{dt^2} = -\omega^3 x_m \cos \pi$$
$$= \omega^3 x_m$$

This indicates that when $t = \frac{1}{2f}$, the velocity is a maximum and negative.

When $t = \frac{1}{f}$

$$\frac{d^2u_i}{dt^2} = -\omega^2 x_m \cos 2\pi$$
$$= -\omega^3 x_m$$

For this value of t, the velocity is a maximum and positive.

In Fig. A.1.2 the graphs of displacement and velocity are plotted against the same time scale. It will be seen that the velocity maxima $u_m = \omega x_m$ occur when $t = 0$, $1/2f$, $1/f$, etc.

The r.m.s. values of displacement or velocity can be obtained from the relationship

$$\text{r.m.s.} = \frac{\text{maximum value}}{\sqrt{2}}$$

Since velocity and displacement curves are of the same shape,

$$u_{rms} = \omega x_{rms}$$

The statement is only true for the magnitude of the velocity and does not state precisely the relationship between displacement and

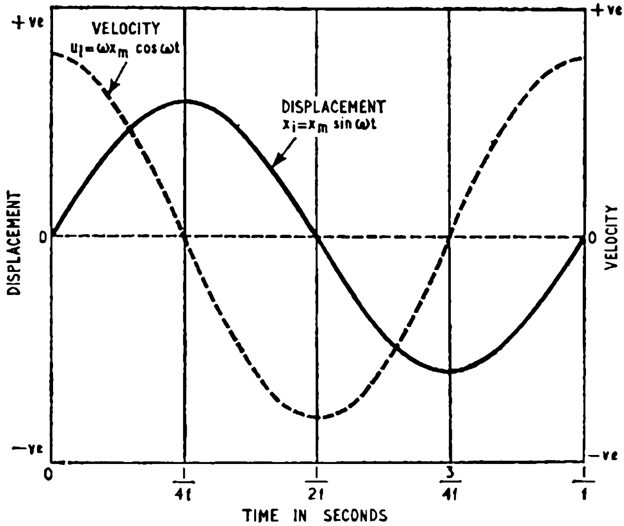


Fig. A.1.2.—Phase relationship between displacement and velocity: simple harmonic motion

velocity. Fig. A.1.2 indicates that the positive value of the displacement curve occurs later in time than the positive value of the velocity curve, the time difference between the maxima being $1/4f$. This time corresponds to a quarter of a wavelength, or to an angular displacement of $\pi/2$ radians, i.e., 90°. The displacement curve lags on the velocity curve by 90° and we could specify the relationship between them more accurately by using j notation. That is,

$$u = j\omega x$$

where u and x may both be r.m.s. values or maximum values. Alternatively, vectors could be used to show the magnitude and the phase relationship between displacement and velocity.

Appendix 2

Longitudinal Wave Motion[2,3,4]

IN THE PARTICULAR type of wave motion associated with sound propagation, the disturbance in the medium takes the form of an oscillatory motion of the air particles in the direction of propagation of the wave. Since the particle displacement is along the direction of propagation, the waves are known as longitudinal waves. The wave of disturbance travels through the air with a velocity many times greater than that attained by the particles in their longitudinal vibrations. It is the purpose of this appendix to find a simple expression for the velocity of the particles at a point on a plane wavefront together with an expression for the variations in atmospheric pressure created by the oscillatory movement of the air particles.

A plane wave is one in which the disturbance is constant over all points of a plane drawn perpendicular to the direction of propagation. Such a plane is called a wavefront.

In order to simplify the expressions and the problem, it is assumed that:

1. The velocity of propagation is constant and independent of frequency.
2. The medium through which the disturbance travels is *lossless*, that is, the sound does not die away as it gets further away from the sound source.
3. The motion of the particles is sufficiently small to allow variations of the second order to be neglected.

It is difficult to produce perfect plane waves in free air but a plane wave can be obtained if the air is confined in a tube and set in motion by a vibrating piston (Fig. A.2.1). The diameter of the tube must be small compared with the wavelengths of the sounds produced, otherwise the tube may act as a wave guide. The piston imparts to the air particles in contact with it a vibratory motion, which is transmitted through the air in the tube without loss. In any plane

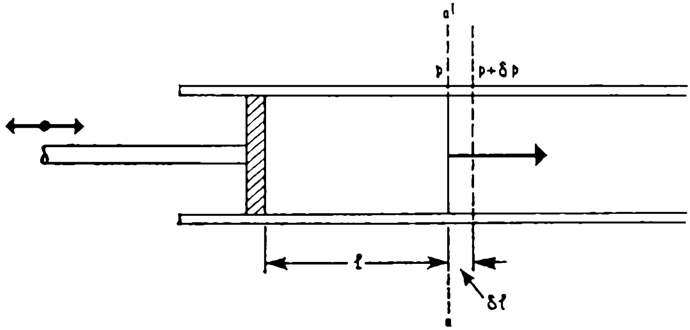


Fig. A.2.1.—Vibrating piston producing plane waves in a tube

aa' at right angles to the tube axis, all the air particles in the plane oscillate with the same velocity and with the same amplitude as the particles in contact with the piston, but in a different phase. The phase difference is dependent on the distance l of the wavefront *aa'* from the piston.

A.2.1. Particle Displacement

If the piston executes simple harmonic motion, the instantaneous velocity u_o is given by

$$u_o = u_m \sin \omega t$$

The time taken for the wave to travel the distance l to the plane *aa'* is l/c sec; hence the instantaneous particle velocity at *aa'* lags by this time interval on the particle velocities at the piston end of the tube.

At time t the instantaneous velocity at *aa'* is

$$u_i = u_m \sin \omega (t - l/c)$$

Now, since $\omega l/c = \dfrac{2\pi f l}{c} = \dfrac{2\pi l}{\lambda} = kl,$

u_i can be expressed in the form:

$$u_i = u_m \sin (\omega t - kl)$$

The particle velocity at *aa'* can be integrated to give the particle displacement x at the wavefront *aa'*.

$$x = -\frac{u_m}{\omega}\cos(\omega t - kl)$$

The minus sign indicates that the movement of the particle is in the opposite direction to that of the propagation.

A.2.2. Pressure at a Point in the Wavefront

To find the pressure in the wavefront, use is made of the relationship *force* = *mass. acceleration.* It is therefore necessary to find the mass of a small element of gas situated at aa' and also the forces acting on it.

Let the thickness of an elemental section at aa' be δl. If A is the cross-sectional area of the tube, the volume of element is $A\delta l$ and if ρ is the density of air, then

$$\text{mass} = \rho A \delta l$$

Let the instantaneous values of the pressures on either side of the element be p and $(p + \delta p)$. Then the force acting on the element is

$$Ap - A(p + \delta p) = -A\delta p$$

Hence the acceleration α is given by

$$\alpha = \frac{\text{force}}{\text{mass}}$$

$$= \frac{-A\delta p_i}{\rho A \delta l}$$

$$= -\frac{1}{\rho}\frac{\delta p_i}{\delta l}$$

where p_i is the instantaneous pressure.

The acceleration of the air particles can also be obtained by differentiating the expression for velocity.

$$\text{Particle acceleration } \frac{\mathrm{d}u_i}{\mathrm{d}t} = \omega u_m \cos(\omega t - kl)$$

Hence $$-\frac{1}{\rho}\frac{\delta p_i}{\delta l} = \omega u_m \cos(\omega t - kl)$$

which, in terms of the differential, becomes

$$dp_i = -\rho\omega u_m \cos(\omega t - kl)\,dl$$

$$\therefore p_i = -\int \rho\omega u_m \cos(\omega t - kl)\,dl$$

$$= \rho \frac{\omega}{k} u_m \sin(\omega t - kl)$$

Since $\omega/k = c$, we have for the instantaneous values of the pressure

$$p_i = \rho c\, u_m \sin(\omega t - kl)$$

If u is the r.m.s. value of the particle velocity, then the r.m.s. pressure p is given by

$$p = \rho c u$$

Table A.2.1

PLANE-WAVE ACOUSTIC PRESSURE, PARTICLE VELOCITY, AND PARTICLE DISPLACEMENT AT 20°C AND FOR $\rho c = 41{\cdot}4$

Pressure in dB relative to 0·0002 *dyn/cm²*	*Pressure in dyn/cm²* p	*Particle velocity in cm/sec* $u = \frac{p}{\rho c}$	*Particle displacement in cm at* 1,000 *c/s* $x = \frac{p}{\rho c\omega}$
0	0·0002	0·0000048	$0{\cdot}76 \times 10^{-9}$
20	0·002	0·000048	$7{\cdot}6 \times 10^{-9}$
30	0·006	0·000152	24×10^{-9}
40	0·02	0·00048	76×10^{-9}
50	0·063	0·00152	240×10^{-9}
60	0·2	0·0048	760×10^{-9}
70	0·63	0·0152	$2{\cdot}4 \times 10^{-6}$
80	2·0	0·048	$7{\cdot}6 \times 10^{-6}$
90	6·32	0·152	24×10^{-6}
100	20·0	0·48	76×10^{-6}
110	63·2	1·52	240×10^{-6}
120	200·0	4·80	760×10^{-6}
130	632·0	15·2	$2{\cdot}4 \times 10^{-3}$
140	2,000·0	48	$7{\cdot}6 \times 10^{-3}$
150	6,320·0	152	24×10^{-3}
160	20×10^{3}	480	76×10^{-3}
170	$63{\cdot}2 \times 10^{3}$	1,520	240×10^{-3}
180	200×10^{3}	4,800	760×10^{-3}
190	632×10^{3}	15,200	$2{,}400 \times 10^{-3}$
200	$2{,}000 \times 10^{3}$	48,000	$7{,}600 \times 10^{-3}$

This can be written in the form

$$u = \frac{p}{\rho c}$$

The similarity to the electrical expression $I = \frac{E}{R}$ should be noted. ρc is the acoustic resistance per unit area and is termed the specific acoustic resistance of the medium. In CGS units ρc is dimensionally in dyne-seconds per gramme or newton-seconds per kilogramme, but in honour of Lord Rayleigh, it is more commonly known as the rayl or the MKS rayl.

The equations show that in a plane progressive wave the pressure and particle velocity are in phase, and the particle displacement is 90° out of phase. Furthermore, if the frequency and either p, u_t or x are known, the other parameters of the wave can be calculated. For example, if p is the r.m.s. value of the pressure, then the r.m.s. velocity $u = \frac{p}{\rho c}$ and the r.m.s. particle displacement $x = \frac{p}{\omega \rho c}$.

The particle velocity and particle displacement for various values of the acoustic pressure p are shown in Table A.2.1. It will be seen that the particle velocities are small in comparison to the velocity of sound.

A.2.3. The Velocity of Sound[1]

The velocity of sound is a scalar quantity and was first investigated by Newton, who showed mathematically that the velocity c of a longitudinal disturbance or strain in a medium is given by

$$c = \left(\frac{E}{\rho}\right)^{\frac{1}{2}}$$

where E is the modulus of elasticity for the particular type of strain and ρ is the density of the material.

In a gas such as air, longitudinal waves are propagated by compression and rarefaction and hence the bulk modulus is used in the expression.

If a mass of gas at constant temperature has a volume V at a pressure P then, according to Boyle's Law, the product PV is a constant. If the pressure increases from P to $(P + p)$, the volume decreases from V to $(V - v)$, but since the product of pressure and volume is constant,

$$(P + p)(V - v) = PV$$
$$pV - vP - pv = 0$$

For sounds of normal intensity, p and v are small and their product pv may be neglected without serious error. Hence,

$$pV - vP = 0 \quad \text{or } P = \frac{pV}{v}$$

The bulk modulus E_t of a gas at constant temperature is therefore equal to the pressure P to which it is subjected.

Hence the velocity of sound c is given by

$$c = \left(\frac{P}{\rho}\right)^{\frac{1}{2}}$$

Since one atmosphere P equals $76 \times 13{\cdot}6 \times 981$ dyn/cm^2, and $\rho = 0{\cdot}0013$ g/cm^3,

$$c = 280 \text{ m/sec}$$

Experiments show that the velocity of sound in air is of the order of 331 m/sec at 0°C, and the value of c as calculated by Newton is obviously incorrect. About a century later Laplace pointed out that the compression and rarefaction in the air which constitute sound, produce changes in the temperature in the air, increasing the temperature in a compression and decreasing the temperature in a rarefaction. Hence the constant-temperature or isothermal equation PV = constant does not apply to sound waves: the adiabatic relationship should be used. That is,

$$PV^{\gamma} = constant$$

where γ is the ratio of the specific heat of the air at constant pressure to the specific heat at constant volume.

If a volume V of air at pressure P is compressed adiabatically to a volume $(V - v)$ by an increase in pressure P to $(P + p)$, then

$$PV^{\gamma} = (P + p)(V - v)^{\gamma}$$
$$P = (P + p)\left(1 - \frac{v}{V}\right)^{\gamma}$$

Now

$$\left(1 - \frac{v}{V}\right)^{\gamma} = \left[1 - \frac{\gamma v}{V} + \text{terms containing higher powers of} \left(\frac{v}{V}\right)\right]$$

Since v is small in comparison to V, the higher powers of $\left(\frac{v}{V}\right)$ may be neglected. Hence

$$P \simeq (P + p)\left(1 - \frac{\gamma v}{V}\right)$$

$$1 - \frac{p}{P} = 1 - \frac{\gamma v}{V}.$$

$$\frac{p}{P} = \frac{\gamma v}{V}$$

$$\gamma P = \frac{Vp}{v}$$

$$= \frac{p}{\frac{v}{V}}$$

Now, the adiabatic modulus E_a is defined as

$$\frac{\text{increase in stress}}{\text{increase in strain}} = \frac{p}{v/V}$$

$$E_a = \gamma P$$

and the velocity c of sound is given by

$$c = \left(\frac{E_a}{\rho}\right)^{\frac{1}{2}}$$

$$= \left(\frac{\gamma P}{\rho}\right)^{\frac{1}{2}}$$

$$= (1{\cdot}4)^{\frac{1}{2}} \times 280$$

$$= 331 \text{ m/sec at } 0°\text{C}$$

Greely found the velocity of sound at low temperatures and in very still air is given approximately by

$$c = (332 + 0{\cdot}6t) \text{ m/sec}$$

where t is the temperature in degrees Centigrade.

Since the value of c = 34,400 cm/sec, or 344 m/sec at 20°C and ρ = ·0013 g/cm³ or 1·3 kg/cm³, $\rho c \simeq$ 41 CGS or 410 MKS.

Appendix 3

Energy in a Plane Progressive Wave[2,3,4]

THE ENERGY ASSOCIATED with the propagation of a sound wave through a gas such as air is in two forms:

1. Kinetic energy, that is, the energy associated with the movement of the particles in the gas.
2. Potential energy associated with the compression or rarefaction in the gas.

Consider a volume element of air of cross-sectional area A and of length δl (Fig. A.3.1). The size of δl is such that all the air particles

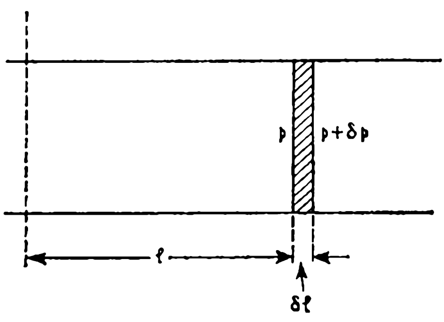


Fig. A.3.1.—Pressures p *and* p + δp) *acting on volume element of length*

in the element are considered to have the same instantaneous displacement and velocity. If the volume element is at a distance l from the sound source, then from Appendix 2,

$$\text{instantaneous velocity } u_i = u_m \sin(\omega t - kl)$$

$$\text{instantaneous displacement } x_i = \frac{-u_m}{\omega} \cos(\omega t - kl)$$

A.3.1. Kinetic Energy

If ρ is the density of the air, then the mass of the volume element is $\rho A\delta l$ where A is the cross-sectional area and δl the length of the element.

$$\text{Instantaneous velocity } u_t = u_m \sin(\omega t - kl)$$

Hence the kinetic energy $\frac{1}{2} mu^2$ is given by

$$\tfrac{1}{2}\rho A\delta l u_m^2 \sin^2(\omega t - \mathrm{k}l)$$

A.3.2. Potential Energy

If the pressures on either side of the element are p and $(p + \delta p)$, the resultant pressure on the element is

$$Ap - A(p + \delta p) = -A\delta p.$$

The elastic restoring force is therefore $+ A\delta p$, and from Appendix 2,

$$\delta p = -\rho\omega u_m \cos(\omega t - kl)\,\delta l$$

Hence $$A\delta p = -\rho A\omega u_m \cos(\omega t - kl)\,\delta l$$

The potential energy in the slice $= \frac{1}{2}$ force $\times$ displacement.
Potential energy

$$= \tfrac{1}{2}\left[-A\rho\omega\delta l u_m \cos(\omega t - kl)\right]\left[-\frac{u_m}{\omega}\cos(\omega t - kl)\right]$$
$$= \tfrac{1}{2} A\rho\delta l\, u_m^2 \cos^2(\omega t - kl)$$

The total energy = potential energy + kinetic energy.

$\therefore$ Total energy

$$= \tfrac{1}{2} A\rho\delta l u_m^2 \cos^2(\omega t - kl) + \tfrac{1}{2}A\rho\delta l u_m^2 \sin^2(\omega t - kl)$$
$$= \tfrac{1}{2} A\rho\delta l u_m^2[\cos^2(\omega t - kl) + \sin^2(\omega t - kl)]$$
$$= \tfrac{1}{2} A\rho u_m^2\delta l$$

The total energy in a length $= \frac{1}{2}\int \rho A u_m^2 \mathrm{d}l,$

$$= \rho A u^2 l,$$

where u is the r.m.s. velocity.

The energy density is

$$\frac{\text{total energy}}{\text{volume}} = \frac{\rho A u^2 l}{A l}$$
$$= \rho u^2$$
$$= \frac{p^2}{\rho c^2}$$

since $u^2 = \dfrac{p^2}{\rho^2 c^2}$.

A.3.3. Acoustic Intensity

The acoustic intensity I of a sound wave is defined as the average rate of flow of energy through a unit area normal to the direction of propagation of the wave. The fundamental units are erg/sec/cm^2 and since the intensity has the dimensions of power transmitted per unit area, it can also be expressed in W/cm^2 by introducing the factor 10^{-7} into the equation given below.

Since the wavefront moves with velocity c, a column of length c will pass through a cross-sectional area A in one second. The energy density in the wave is $\dfrac{p^2}{\rho c^2}$. Hence the total energy passing through area A in one second is given by

$$\frac{p^2}{\rho c^2} A c = \frac{A p^2}{\rho c}$$

The power transmitted per unit area is $\dfrac{p_2}{\rho c}$ and since $p^2 = u^2 \rho^2 c^2$, the power transmitted per unit area may be written

$$\frac{u^2 \rho^2 c^2}{\rho c} = u^2 \rho c$$

The similarity to the electrical expression for power $P = I^2 R$ should be noted.

It will be appreciated that the expressions are valid only for plane waves and cannot be applied to spherical waves unless the distance from the source is such that the wavefront can be considered as plane.

A.3.4. Intensity Standards

Intensity can be expressed in a variety of ways. In microphone techniques intensity is usually specified in terms of acoustic pressure rather than particle velocity. Since intensity specified in this way involves ρc, the acoustic resistance of the medium (which in turn depends on atmospheric pressure and temperature) must also be specific or understood. At a temperature of 20°C and at standard atmospheric pressure the density of air is 0·00121 g/cm³ and the velocity of sound is 34,300 cm/sec, giving a standard specific acoustic resistance for air

$$(\rho c)_{20} = 41{\cdot}5 \text{ g/cm}^2 \text{ sec}$$

The intensity of a sound corresponding to an acoustic pressure r.m.s. of 1 dyn/cm² under standard conditions of air temperature and atmospheric pressure is given by

$$\begin{aligned} I &= \frac{p}{\rho c} \\ &= \frac{1}{41{\cdot}5} \\ &= 0{\cdot}0241 \text{ erg/sec/cm}^2 \end{aligned}$$

Appendix 4

Stationary Waves[2,3,4]

STATIONARY OR STANDING waves are produced by the interaction of two or more progressive-wave systems of the same frequency. In order to produce the nodes or partial nodes which are a feature of standing-wave systems, the interfering waves must have components which travel in opposite directions. Two sound sources of identical frequency are not needed to set up a standing-wave system, for these are more frequently produced as the result of the reflection of a single progressive wave from a rigid surface or by reflection from an abrupt discontinuity in the medium through which the wave is propagated.

When a progressive wave impinges on an infinitely rigid surface, the wave is reflected. Since no movement of the air particles is possible at the rigid surface, the particle velocity of the reflected wave must, at the surface, be 180° out of phase with the advancing wave in order to produce velocity cancellation. As shown in Appendix 2, the particle velocity of a wave travelling along the positive direction, that is, from left to right, can be represented by

$$u_m \sin(\omega t - kl)$$

where l is a distance measured from left to right. The angle kl radians signifies a progressive lag in phase as the wave moves away from the origin. Conversely, a wave travelling from right to left is advancing in respect to the cycle of events and can be represented by

$$u_m \sin(\omega t + kl)$$

In Fig. A.4.1 the incident wave is represented by the wavefront A and its particle velocity by the expression

$$u_i = u_m \sin(\omega t + kl)$$

If no sound absorption occurs at the rigid surface, the particle velocity at the surface must be zero, hence the amplitude of the reflected wave must be equal to, but 180° out of phase with the advancing wave: that is, the amplitude of the particle velocity of the reflected wave is $-u_m$. In Fig. A.4.1 the reflected wave is represented by the wavefront *B* and since it is moving in the positive direction, the expression for its instantaneous particle velocity is

$$u_i = -u_m \sin(\omega t - kl)$$

The equations for the particle velocities of the two waves are linear: that is, u_m occurs in the first degree only, hence by the principle

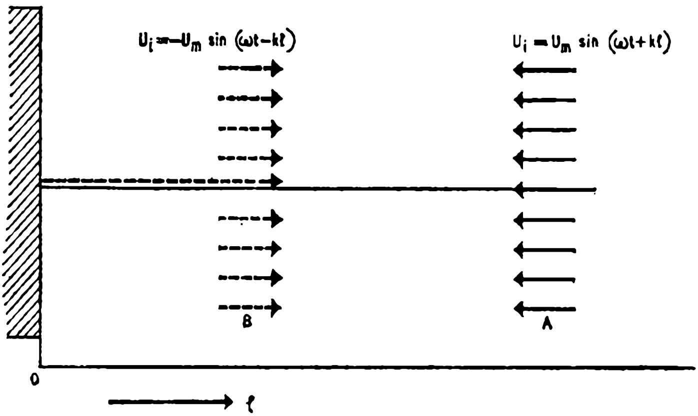


Fig. A.4.1.—Incident and reflected waves at a rigid boundary

of superposition, the equations may be added to obtain the expression for the resultant wave or waves. The resultant particle velocity is therefore the algebraic sum of the waves

$$u_m \sin(\omega t + kl) - u_m \sin(\omega t - kl)$$

Since $\sin A - \sin B = 2 \cos \frac{A+B}{2} \sin \frac{A-B}{2}$ the resultant instantaneous particle velocity u_r is given by

$$u_r = 2\, u_m \cos \omega t \sin kl$$

In a plane wave, acoustic pressure and particle velocity are in phase and the amplitude of the pressure p is given in terms of the particle velocity by

$$p_i = \rho c u_i$$

The expression for the pressure of the incident wave p_i is

$$p_i = \rho c u_m \sin(\omega t + kl)$$

and for the reflected wave is

$$p_i = \rho c u_m \sin(\omega t - kl)$$

There is no negative sign associated with the amplitude of the reflected pressure wave, for pressure is a scalar quantity and possesses magnitude but not direction. The resultant pressure is the arithmetical sum of the two pressures

$$p_r = \rho c u_m \sin(\omega t + kl) + \rho c u_m \sin(\omega t - kl)$$

Since $\sin A + \sin B = 2 \sin \frac{A+B}{2} \cos \frac{A-B}{2}$ the resultant pressure p_r at any distance l from the rigid surface is

$$p_r = 2\rho c\, u_m \sin \omega t \cos kl$$

The equations for particle velocity and pressure in the resultant wave indicate that both parameters vary not only with time but also with the distance l from the reflecting surface.

A.4.1. Variations in Particle Velocity with Distance l and Time t

At a time t the variations in particle velocity with distance from the reflecting surface can be obtained by substituting the appropriate value for l in the equation

$$u_r = 2\, u_m \cos \omega t \sin kl$$

At $l = 0$, $\sin kl = \sin 0° = 0$. Hence $u_r = 0$.

The particle velocity is always zero at the reflecting surface, irrespective of time.

At $l = \lambda/2$, or at $l = n\lambda/2$,

$$\sin kl = 0 \text{ if } n \text{ is an integer}$$

Hence the particle velocity is always zero at $l = 0$, $\lambda/2$, λ, etc. These positions of zero velocity are called nodes.

At $l = \lambda/4$ or at $l = q\lambda/4$,

$$\sin kl = \pm 1 \text{ if } q \text{ is an odd integer}$$

The particle velocity $u_i = 2u_m \cos \omega t$ varies with time and has a maximum amplitude of twice the value of the incident wave. These positions $l = \lambda/4$, $3\lambda/4$, $5\lambda/4$, etc., are velocity maxima called antinodes.

Fig. A.4.2 is a family of curves which show the particle velocity at certain instants of time, $t = 1/6f$, $1/4f$, $1/3f$, $1/2f$, plotted against the ratio l/λ. It will be seen that velocity maximum $2u_m$ occurs when $\cos \omega t = 1$, that is when $t = 1/2f$, $1/f$, $3/2f$, etc. At the instants when $\cos \omega t = 0$, that is, when $t = 1/4f$, $3/4f$, the particle velocity is zero throughout the space occupied by the complex wave.

A.4.2. Variations in Acoustic Pressure with Distance l and time t

At any time t the variations in pressure with distance from the reflecting surface can be obtained by substituting the appropriate value for l in the equation

$$p_r = 2p_m \sin \omega t \cos kl$$

At $l = 0$, $\cos kl = \pm 1$. Hence the pressure

$$p_r = 2p_m \sin \omega t$$

varies sinusoidally with time, having its maximum positive or negative values $2p_m$ when $t = 1/2f$, $3/4f$, $5/4f$, etc.

At $l = \lambda/2$ or at $l = n\lambda/2$,

$$\cos kl = \pm 1 \text{ if } n \text{ is an integer}$$

and these positions correspond to the pressure antinodes.

At $l = \lambda/4$, or at $l = q\lambda/4$,

$$\cos kl = 0 \text{ where } q \text{ is an odd integer.}$$

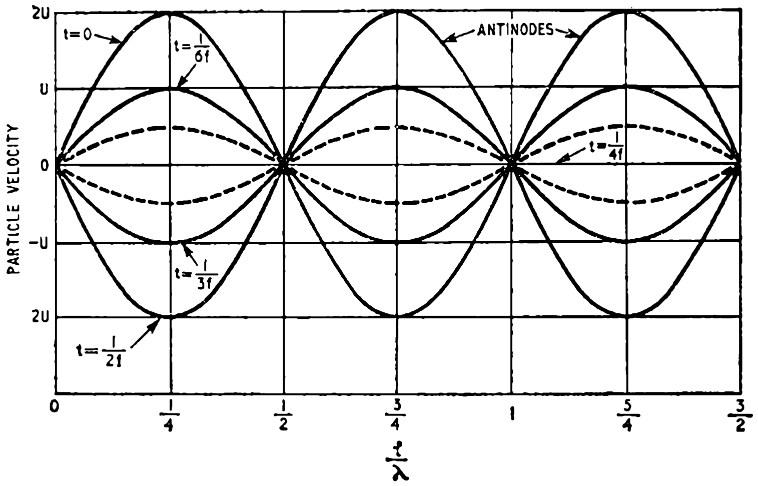


Fig. A.4.2.—Stationary wave (particle velocity) due to complete reflection from a rigid boundary

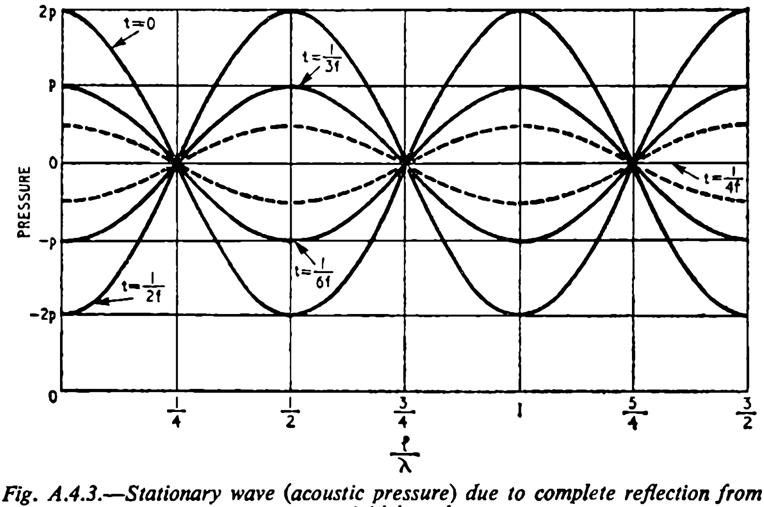


Fig. A.4.3.—Stationary wave (acoustic pressure) due to complete reflection from a rigid boundary

Hence the pressure is always zero at $l = \lambda/4$, $3\lambda/4$, $5\lambda/4$, etc. These positions are the pressure nodes of the system.

Fig. A.4.3 is a family of curves showing the acoustic pressures at certain time instants $t = 1/6f$, $1/4f$, $1/3f$, $1/2f$ plotted against the ratio l/λ. It will be seen that a pressure maximum occurs at the reflecting surface and its amplitude is twice that of the incident wave. (This is the pressure doubling referred to in Chapter 5, when the diameter of the sphere is large enough to ensure complete reflection of the incident wave at $\theta = 0°$.)

At the instants when $\sin \omega t = 0$ the acoustic pressure is zero throughout the space occupied by the complex wave.

A.4.3. Stationary Waves with Partial Reflection

If partial absorption occurs on contact with the boundary, the amplitude of the reflected wave is reduced and can be represented by gu_m where g is a fraction less than unity. The amplitude of the wave transmitted through or into the boundary is therefore $(1 - g)\, u_m$. If the expression for the particle velocity of the incident wave is

$$u_m \sin (\omega t + kl)$$

the expression for the reflected wave is then

$$- gu_m \sin (\omega t - kl)$$

and the resultant particle velocity is the algebraic sum of

$$u_m \sin (\omega t + kl) - gu_m \sin (\omega t - kl)$$

Writing A for $(\omega t + kl)$ and B for $(\omega t - kl)$, this becomes

$$\begin{aligned} & u_m (\sin A - g \sin B) \\ &= u_m (\sin A - g \sin A + g \sin A - g \sin B) \\ &= u_m [(1 - g) \sin A + g (\sin A - \sin B)] \\ &= u_m \left[(1 - g) \sin A + 2\, g \cos \frac{A + B}{2} \sin \frac{A - B}{2}\right] \\ &= (1 - g)\, u_m \sin (\omega t + kl) + 2\, gu_m \cos \omega t \sin kl \end{aligned}$$

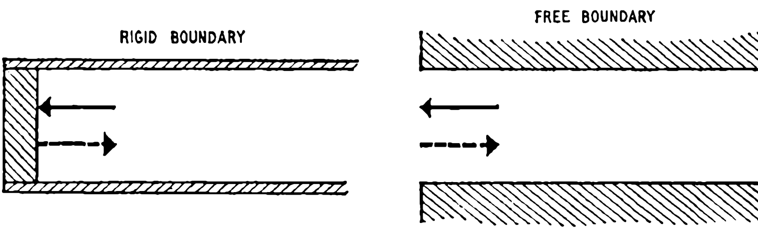
The first term of the expression represents a wave travelling from right to left and is that portion of the incident wave which is transmitted through the boundary. The second term will be recognised as a stationary wave having a maximum amplitude $2\,gu_m$ and since g is less than unity, $gu_m < u_m$. The stationary wave has no nodal points unless $g = 1$, but has a series of maxima and minima, the maxima occurring at $l = \lambda/4$, $3\lambda/4$, $5\lambda/4$, etc., and the minima at $l = 0$, $\lambda/2$, λ, $3\lambda/2$, etc.

A.4.4. Rigid and Free Boundaries

At a rigid boundary the acoustic pressure in the reflected wave is in phase with the pressure in the incident wave: that is, a compression or positive excess of pressure in the incident wave is returned as a compression in the reflected wave. If, however, the boundary is free rather than rigid, as would be the case at the open end of an organ pipe or at the opening of a Helmholtz resonator, a compression is reflected as a rarefaction and a rarefaction as a compression. The condensations are therefore reversed.

Table A.4.1

CONDITIONS AT RIGID AND FREE BOUNDARIES



Displacement reversed	Displacement unchanged
Particle velocity reversed	Particle velocity unchanged
Condensation unchanged	Condensation reversed
Pressure unchanged	Pressure difference reversed

In general, when the characteristic impedance of the medium in which the incident wave is propagated is greater than the characteristic impedance of the medium encountered at the boundary, the condensations are reversed. Conversely, if the characteristic impedance of the medium through which the incident wave is transmitted is less than the characteristic impedance encountered at the boundary, the condensations remain unchanged.

Two conditions must be satisfied at any boundary, whether it be free or rigid:

1. If the two media at the boundary are to remain in contact, the particle velocities normal to the boundary must be equal.
2. Since pressure in a fluid medium such as air is a continuous scalar quantity, the acoustic pressures on the two sides of the boundary must be equal.

From the above, the conditions at rigid and free boundaries can be deduced and are summarised in Table A.4.1.

Appendix 5

Electromechanical Analogies[3,5]

IN GENERAL, mechanical systems possess mass, compliance and friction, in combinations which can be simple or complex, depending on the nature of the mechanism. A simple arrangement is shown in Fig. A.5.1.

The mass m which is coupled by a spring of stiffness S to a fixed outer bracket is capable of movement between the guides, but as a

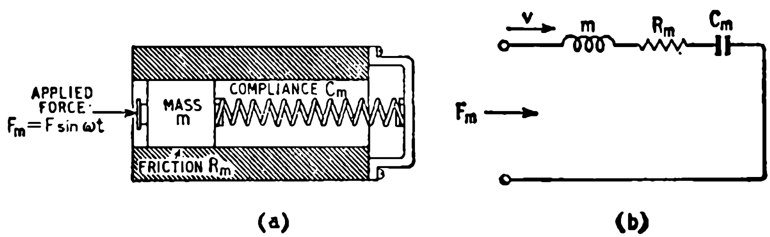


Fig. A.5.1.—(a) simple mechanical system: (b) equivalent circuit

result of its movement it is subject to friction R_m. The stiffness of the spring is S dyn/cm displacement.

When the system is subjected to a sinusoidal force F_m dyn applied as shown in Fig. A.5.1, the mass is displaced sinusoidally. If the amplitude of the displacement is x, then the applied force is absorbed in overcoming the component forces within the system as follows.

The force to overcome the inertia of the mass is

$$\text{mass} \times \text{acceleration} = m\frac{d^2x}{dt^2}$$

The force to overcome the frictional forces in the system is

$$R_m \times \text{velocity} = R_m \frac{dx}{dt}$$

The force to overcome the stiffness of the system is

$$S \times \text{displacement} = Sx$$

Expressing the result in the form of a differential equation, we have

$$F_m = m\frac{d^2x}{dt^2} + R_m\frac{dx}{dt} + Sx$$
$$= F_m \sin \omega t$$

This may be rewritten as follows

$$F_m = m\frac{du}{dt} + Ru + Sudt$$
$$= F \sin \omega t$$

Writing for S the reciprocal of the compliance, i.e., $S = \frac{1}{C_m}$, we have

$$F_m = m\frac{du}{dt} + Ru + \frac{1}{C_m}udt$$
$$= F \sin \omega t \quad (1)$$

When an electrical circuit having inductance L, resistance R and capacitance C in series is subjected to a sinusoidal voltage V, this voltage is absorbed in overcoming the induced and resultant voltages within the system, as follows

$$V = L\frac{di}{dt} + Ri + \frac{1}{C}\ idt$$
$$= V_{max} \sin \omega t \quad (2)$$

Equations 1 and 2 are similar in form, and the steady-state solution is given by

$$u = \frac{F_m}{R_m + j\left(\omega m - \frac{1}{\omega C_m}\right)} \quad \text{for the mechanical system}$$

and $$I = \frac{V}{R + \mathrm{j}\left(\omega L - \frac{1}{\omega C}\right)}$$ for the electrical circuit

where u and F_m and I and R are r.m.s. values.

By comparing Equations 1 and 2 we may draw the following conclusions:

Force is analogous to *voltage.*
Mass is analogous to *inductance.*
Damping is analogous to *resistance.*
Displacement $\int u\,dt$ is analogous to *charge* $\int i\,dt$.

Comparison of the two final equations shows that:

Velocity is analogous to *current.*
Compliance is analogous to *capacitance.*
Mechanical impedance is analogous to *electrical impedance.*

Using these analogies, the mechanical system of Fig. A.5.1 (a) may be represented by the equivalent circuit of Fig. A.5.1 (b).

Appendix 6

Electro-acoustical Analogies[3,5]

THE ACOUSTICAL NETWORKS associated with microphones consist in general of cavities and small tubes or slits. In nearly all cases the dimensions of the tubes and cavities are small compared with the wavelength of sound: then the flow or movement of the air in the network is analogous to current in an electrical circuit with lumped elements of inductance, capacitance and resistance.

The analogue of pressure in an acoustical circuit is voltage in the electrical circuit, and the analogue of current in the electrical circuit is total air flow in the acoustical circuit. Total air flow can be defined as the particle velocity multiplied by the cross-sectional area of the acoustical circuit at the appropriate point or plane.

This concept of total air flow is best explained by considering an infinitely light piston which can move freely in a tube or constriction whose cross-sectional area is A. If an acoustic pressure p on one side of the piston causes it to move with velocity u in phase with the pressure, then the total power supplied is pAu. This is analogous to the power VI in a resistive electrical circuit. The total pressure pA can be taken as analogous to voltage, in which case u is analogous to the current I. As we have seen, this is the concept employed in electromechanical analogies where

$$\text{mechanical resistance} = \frac{\text{total force}}{\text{velocity}} = \frac{pA}{u} = R_m \text{ mechanical ohms.}$$

A.6.1. Acoustical Resistance

An alternative method is to take p as analogous to voltage V, and Au, the volume current, as analogous to current I. This is the concept used in electro-acoustical analogies, where it is expressed as follows:

$$\text{acoustical resistance} = \frac{\text{pressure}}{\text{volume current}} = \frac{p}{Au} = \frac{p}{U}$$

where $U = Au$ *in the volume current.* Hence

$$\text{acoustical resistance } R_a = \frac{\text{dyn/cm}^2}{\text{cm}^3/\text{sec}} = \frac{\text{dyn}}{\text{cm}^2} \times \frac{\text{sec}}{\text{cm}^3} = \text{dyn sec/cm}^5$$

A.6.2. Inertance

If a tube or constriction in which air is moved by an acoustic pressure has an effective length l_e and a cross-sectional area A, then the total mass of air moved is $\rho A l_e$ where ρ is the density of air. The actual length l of the tube is not the effective length, for the air beyond the ends of the tube is also set in motion by the movement of the air in the tube. The effective length l_e in terms of the actual length l is given by:

$$l_e \simeq l + 0{\cdot}58r$$

where r is the radius of the tube. (A hole in a very thin plate may have an appreciable effective length, especially if the cross-sectional area of the hole is large.)

$$\text{Inertance} = \frac{\text{sound pressure}}{\text{rate of change of volume current}}$$

This is analogous to mass, which is defined by the ratio

$$\frac{\text{force}}{\text{rate of change of velocity}}.$$

$$\text{Inertance } M = \frac{p}{\frac{du}{dt}}$$

$$= \frac{\text{dyn/cm}^2}{\text{cm}^3\text{/sec/sec}}$$

$$= \frac{\text{dyn}}{\text{cm}^2} \times \frac{\text{sec}^2}{\text{cm}^3}$$

$$= \frac{\text{dyn (sec)}^2}{\text{cm}^5}$$

Since 1 dyne is the force required to produce an acceleration of 1 centimetre per second in a mass of 1 gramme,

$$\frac{1 \text{ dyn (sec)}^2}{\text{cm}} = 1 \text{ g}$$

and

$$M = \text{g/cm}^4.$$

A.6.3. Capacitance

Since $U = Au$ where u = rate of change of displacement, i.e., $u = \frac{dx_a}{dt}$, the capacitance C_a is equal to the volume displacement x_a divided by the sound pressure

$$C_a = \frac{x_a}{p}$$

$$= \frac{\text{cm}^3}{\text{dyn/cm}^2}$$

$$= \frac{\text{cm}^3}{\text{dyn}} \times \text{cm}^2$$

$$\therefore C_a = \text{cm}^5\text{/dyn}$$

A.6.4. Simple Acoustical System (Helmholtz Resonator)

The Helmholtz resonator shown in Fig. A.6.1 (a) has a neck of effective length l and a cross-sectional area A and a cavity volume V_o. If the dimensions of the resonator are small in comparison with the

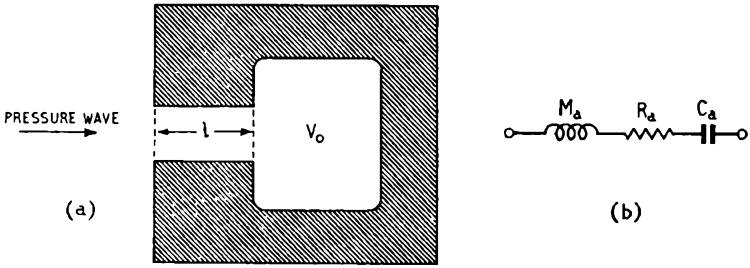


Fig. A.6.1.—(a) Simple acoustical system (Helmholtz resonator); (b) equivalent circuit

wavelength of sound, the mass and compliance of the system are provided by the air contained in the neck and cavity of the resonator.

From Appendix 2

$$\textit{velocity of sound } c = \left(\frac{E}{\rho}\right)^{\frac{1}{2}}$$

where E is the adiabatic bulk modulus and ρ is the density of the air. Hence

$$E = \rho c^2$$

if an increase in pressure p produces a displacement x of the air in the neck of the resonator, and a change in volume v. Then the condensation s is defined as the incremental change in volume to the original volume. That is,

$$s = -\frac{v}{V_o}$$

The negative sign indicates that an increase in pressure produces a decrease in volume. Since $v = -Ax$, the sign being negative because the displacement causes a decrease in volume, the condensation is

$$s = -\frac{v}{V_o}$$

$$= \frac{Ax}{V_o}$$

A.6.5. Acoustical Mass

The mass of air in the neck of the resonator is ρAl and the force required to accelerate this mass is given by

$$\rho A l \ddot{x} \text{ where } \ddot{x} = \frac{d^2x}{dt^2}$$

A.6.6. Acoustical Stiffness

$$\begin{aligned} p &= Es \\ &= \rho c^2 s \\ &= \rho c^2 A \frac{x}{V_o} \end{aligned}$$

The force required to overcome stiffness is

$$pA = \rho c^2 A^2 \frac{x}{V_o}$$

A.6.7. Acoustical Resistance

Sound is dissipated at the mouth of the resonator, and Lord Rayleigh has shown that the resistive force is given by

$$\frac{\rho \omega k}{2\pi} A^2 \dot{x}$$

where $k = \frac{2\pi}{\lambda}$ is the wavelength constant.

A.6.8. Equation of Motion

Equating the combined forces overcoming inertia, resistance and stiffness to the actuating force $AP_m \sin \omega t$,

$$AP_m \sin \omega t = \rho A l \ddot{x} + \left(\frac{\rho \omega k}{2\pi}\right) A^2 \dot{x} + \rho c^2 \frac{A^2}{V_o} x$$

Multiplying by $\frac{1}{A}$ and rearranging,

$$P_m \sin \omega t = \left(\frac{\rho l}{A}\right) A\ddot{x} + \left(\frac{\rho \omega k}{2\pi}\right) A\dot{x} + \left(\frac{\rho c^2}{V_o}\right) Ax$$

Writing

$$M_a = \frac{\rho l}{A}, \ R_a = \frac{\rho\omega k}{2\pi} \text{ and } C_a = \frac{1}{\frac{V_o}{\rho c^2}}$$

and if $A\ddot{x} = \dot{U}$, $A\dot{x} = U$ and $Ax = \int U\mathrm{d}t$ where U is the volume velocity, then

$$P_m \sin \omega t = M_a\dot{U} + R_aU + \int\frac{U\mathrm{d}t}{C_a}$$

from which,

$$U = \frac{P_m \sin \omega t}{R_a + \mathrm{j}\left(\omega M_a - \frac{1}{\omega C_a}\right)}$$

This is the steady-state solution; hence

$$U_{rms} = \frac{P}{R_a + \mathrm{j}\left(\omega M_a - \frac{1}{\omega C_a}\right)}$$

$$= \frac{P}{\left[R_a^2 + \left(\omega M_a - \frac{1}{\omega C_a}\right)^2\right]^{\frac{1}{2}}}$$

The denominator is the acoustical impedance and the resonator can be represented by the equivalent circuit of Fig. A.6.1 (b).

By analogy, $M_A = \frac{\rho l}{A}$ is the acoustic mass and is equal to $\frac{\rho l A}{A^2}$, that is, to the $\frac{\text{actual mass}}{A^2}$.

$$C_A = \frac{V_o}{\rho c^2}$$

$$= \frac{V_o}{E}$$

is the acoustical capacitance.

$$R_A = \frac{\rho\omega k}{2\pi}$$

$$= 2\pi\rho\frac{f^2}{c}$$

is the radiation resistance (acoustical).

Note that when the dimensions are small compared with the wavelength, the radiation resistance of the mouth of the resonator is proportional to f^2.

A.6.9. Resonance Frequency

Maximum volume current or air flow in the neck of the resonator occurs at a frequency which makes the total reactance zero, that is, when

$$\frac{\omega\rho l}{A} = \frac{\rho c^2}{\omega V_o}$$

$$\omega^2 = \left(\frac{c^2}{V_o}\right)\frac{A}{l}$$

The resonance frequency is thus given by

$$f_o = \frac{c}{2\pi}\left(\frac{A}{l V_o}\right)^{\frac{1}{2}}$$

A.6.10. Formula for Acoustic Impedances

A tube of small diameter has an acoustical impedance which is due partly to inertance and partly to acoustical resistance. The acoustical resistance, which may be large, results from viscous friction between the air particles confined in the narrow tube. If the diameter of the tube is small compared with its length, the end correction may be neglected.

The impedance of the tubular path (Fig. A.6.2) whose diameter is small compared with its length and whose length is small compared with the wavelength of the pressure wave, is given by

$$Z_a = \frac{1}{\pi r^2}\left(\frac{8\mu}{r^2} + j\omega\frac{4\rho}{3}\right)$$

where r = radius of the tube in centimetres,
l = length of the tube in centimetres,
$\omega = 2\pi f$,
μ = viscosity coefficient $1{\cdot}83 \times 10^{-4}$ g/cm/sec for air.

A narrow slot of the type shown in Fig. A.6.3 has an acoustical impedance similar to that of a narrow tube. The end correction may be neglected if the thickness of the slot is small compared with the length.

The impedance of a slot whose thickness is small compared with its length and with a length small compared with wavelength is given by

$$Z_a = \frac{12\mu l}{t^3 W} + j\omega^6 \frac{\rho l}{5tW}$$

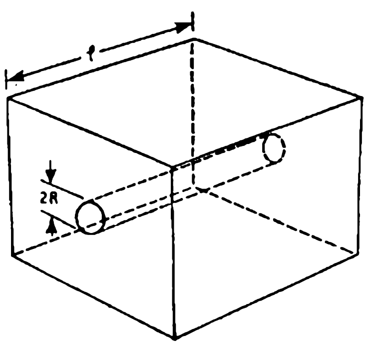
where l = length of slot in centimetres measured in the direction of flow,

t = thickness of slot in centimetres normal to the direction of flow,

W = width of slot in centimetres normal to the direction of flow.

The resistive part of the impedance is inversely proportional to the cube of the thickness t of the slot. The inertance is inversely

Fig. A.6.2.—Tubular path of small diameter



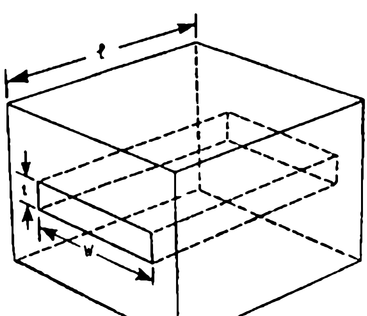


Fig. A.6.3.—Slot with the thickness small compared to its length

proportional to t. By suitable choice of t, the ratio of inertance to resistance may be made to have almost any desired value. The magnitude of the impedance is determined by the choice of the ratio of length to width that is, l/W. The slot is a most useful form of acoustical impedance and is of particular value in microphone design techniques.

A.6.11. Equivalent Mechanical Values

The moving system of a microphone may be complex and may include both mechanical and acoustical elements. The schematic drawing of Fig. A.6.4 (a) shows a mechanical system coupled by means of a piston of cross-sectional area A to a Helmholtz resonator.

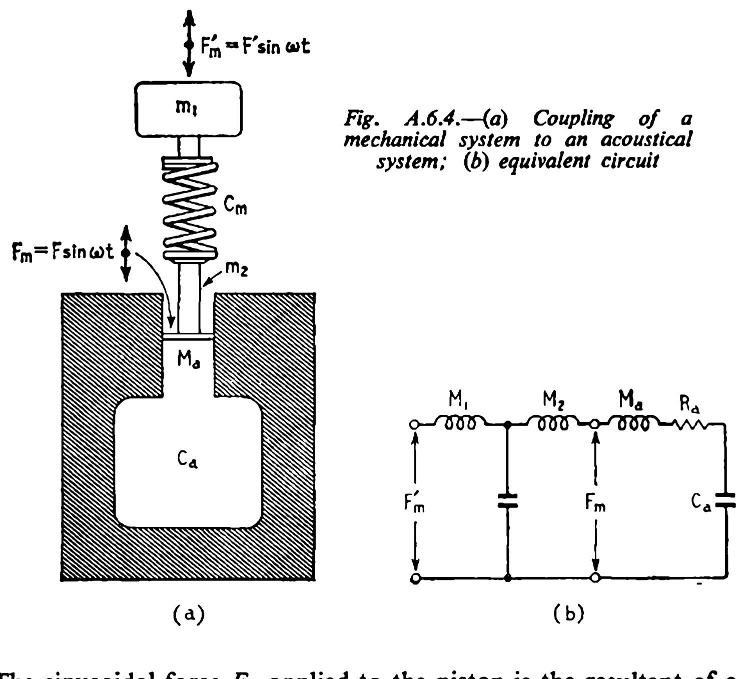


Fig. A.6.4.—(a) Coupling of a mechanical system to an acoustical system; (b) equivalent circuit

The sinusoidal force F_m applied to the piston is the resultant of an external force F_m' applied to the mass m_1 in the mechanical system. The equivalent circuit of Fig. A.6.4 (b) shows that the combined system may be represented by a network comprising mechanical and acoustical elements. The equivalent mechanical values of the acoustical qualities may be obtained from the following relationships.

(1) Force = pressure × area

$$F_m = pA$$

(2) $$\text{Velocity} = \frac{\text{volume current}}{\text{area}}$$

$$u = \frac{U}{A}$$

(3) $$\text{Displacement} = \frac{\text{volume displacement}}{\text{area}}$$

$$x = \frac{x_a}{A}$$

(4) $$\text{Resistance} = \text{acoustical resistance} \times \text{area squared}$$

$$R_m = R_a A^2$$

(5) $$\text{Mass} = \text{inertance} \times \text{area squared}$$

$$m = MA^2$$

(6) $$\text{Compliance} = \frac{\text{acoustical capacitance}}{\text{area squared}}$$

$$C_m = \frac{C_a}{A^2}.$$

Appendix 7

Stretched Membranes and Thin Plates[4,5,6]

A.7.1. Circular Membranes

THE DIAPHRAGM of an electrostatic microphone and the stretched parchment skins of drum heads are examples of circular membranes. A membrane, as distinct from a thin plate, is a two-dimensional surface which possesses mass but no friction, and whose stiffness is

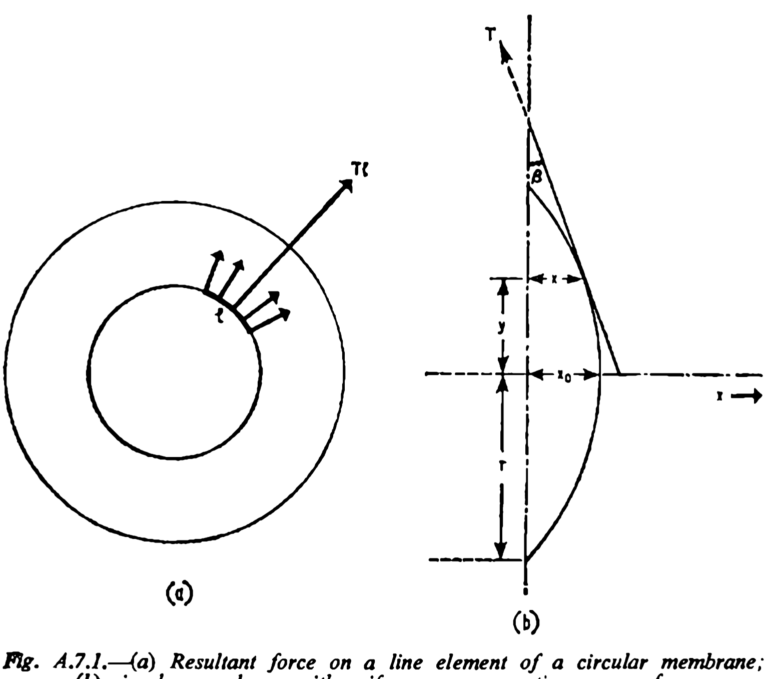


Fig. A.7.1.—(a) Resultant force on a line element of a circular membrane; (b) circular membrane with uniform pressure acting on one face negligible in comparison with the restoring force, due to the tension applied to the system.

In the analysis it will be assumed first, that the tension T per unit length is uniformly distributed throughout the membrane, so that the material on opposite sides of a line segment of length l will tend to be pulled apart by a force Tl (Fig. A.7.1 (a)) and secondly, that the tension remains constant and independent of any small deflection of the membrane.

A uniform pressure P per unit area acting on one face deflects the membrane from its normal flat form, as in Fig. A.7.1 (b). Let the displacement of a point at radius y be x.

The component of the tension along the x axis is $-T\sin\beta$.

For small displacements $-T\sin\beta \simeq -T\tan\beta$.

In the notation of the calculus $-T\tan\beta = -T\dfrac{dx}{dy}$.

The restoring force around a circle of radius y having a circumference $2\pi y$ is therefore

$$-2\pi y\, T\,\frac{dx}{dy}$$

The external force applied to a circle of radius y and area πy^2, is given by

$$\pi y^2 P$$

For equilibrium, the applied force = the restoring force. Hence,

$$\pi y^2 P = -2\pi y T\,\frac{dx}{dy}$$

$$\text{or } dx = -\frac{P}{2T}\, y\,dy$$

$$x = -\frac{P}{2T}\int y\,dy$$

$$= -\frac{Py^2}{4T} + A$$

The constant A can be evaluated from the boundary conditions: $y = r$ when $x = 0$.

$$0 = -\frac{Pr^2}{4T} + A \quad \text{or } A = \frac{Pr^2}{4T}$$

$y = 0$ when $x = x_o$, that is,

$$x_o = A = \frac{Pr^2}{4T}$$

The expression for the deflection x of the membrane under the action of a uniform force P is given by

$$\begin{aligned} x &= -\frac{Py^2}{4T} + \frac{Pr^2}{4T} \\ &= \frac{P}{4T}(-y^2 + r^2) \\ &= \frac{Pr^2}{4T}\left(-\frac{y^2}{r^2} + 1\right) \\ &= x_o\left(1 - \frac{y^2}{r^2}\right) \end{aligned}$$

The membrane therefore assumes a parabolic shape when a uniform pressure is applied to one face. If the pressure is alternating, the expression for displacement is complex, for the membrane can exhibit a number of vibrational modes in which different parts of the membrane are displaced in anti-phase. The complex movements

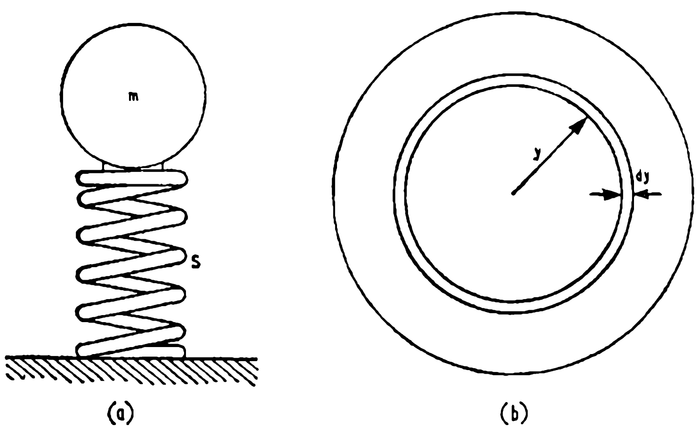


Fig. A.7.2.—(a) Simple mechanical equivalent of stretched membrane ; (b) ring element of stretched membrane produce displacement patterns characterised by circumferential and radial nodes. These vibrational modes occur at frequencies above the fundamental frequency f_{o1} of the membrane and since it is customary to operate the diaphragms of electrostatic microphones below f_{o1} in order to obtain a stiffness-controlled system, the complex modes are not normally excited by sounds in the audio-frequency range.

A.7.2. Fundamental Frequency

An approximate expression for the fundamental frequency f_{o1} can be obtained by comparing the kinetic energy of the membrane with the kinetic energy of the simple equivalent system, as in Fig. A.7.2 (a). In the simple system the equivalent dynamic mass of the membrane is represented by m and the resilience due to tension is represented by the spring S. If an alternating force P applied to the simple system produces an alternating displacement x_o, then the velocity of the mass is ωx_o and the kinetic energy is therefore given by

$$\begin{aligned} KE &= \tfrac{1}{2}mu^2 \\ &= \tfrac{1}{2}m\,(\omega x_o)^2 \end{aligned}$$

The kinetic energy in the membrane can be obtained by the summation of an infinite number of rings concentric with the centre of the membrane. The area of a ring (Fig. A.7.2 (b)), of radius y and small width $\mathrm{d}y$ is $2\pi y\,\mathrm{d}y$. If ρ' is the mass per unit area of the membrane, the mass of the ring is

$$2\pi\rho' y\,\mathrm{d}y$$

If the displacement of the ring under the action of the alternating pressure P is x, then the velocity of the ring is ωx.

It has been shown that

$$x = x_o\left(1 - \frac{y^2}{r^2}\right)$$

Hence the kinetic energy in the ring is

$$\begin{aligned} \tfrac{1}{2}mu^2 &= \tfrac{1}{2}\,(2\pi\rho' y\,\mathrm{d}y)\,(\omega x)^2 \\ &= \pi\rho' y\,\mathrm{d}y\,\omega^2 x_o^{\,2}\left(1 - \frac{y^2}{r^2}\right)^2 \\ &= \omega^2 x_o^{\,2}\,\pi\rho' y\left(1 - \frac{y^2}{r^2}\right)^2 \mathrm{d}y \end{aligned}$$

The kinetic energy of the membrane may be obtained by integrating this expression

$$\text{Kinetic energy of membrane} = \omega^2 x_o^2\, \pi\rho' \int_o^r y\left(1 - \frac{y^2}{r^2}\right)^2 dy$$

$$= \omega^2 x_o^2\, \pi\rho' \int_o^r \left(y - \frac{2y^3}{r^2} + \frac{y^5}{r^4}\right) dy$$

$$= \omega^2 x_o^2\, \pi\rho' \left[\tfrac{1}{2}y^2 - \frac{2}{4}\frac{y^4}{r^2} + \frac{1}{6}\frac{y^6}{r^4}\right]_o^r$$

Substituting first $y = r$, and then subtracting when $y = 0$, yields

$$KE = \omega^2 x_o^2\, \pi\rho' \left(\tfrac{1}{2}r^2 - \tfrac{1}{2}r^2 + \frac{1}{6}r^2\right)$$

$$= \omega^2 x_o^2\, \pi\rho' \frac{r^2}{6}$$

Writing this in the form

$$KE = \tfrac{1}{2}mu^2$$

we have

$$KE = \tfrac{1}{2}\left(\frac{2\pi r^2 \rho'}{6}\right)(\omega x_o)^2$$

$$= \tfrac{1}{2}\left(\frac{\pi r^2 \rho'}{3}\right)(\omega x_o)^2$$

Thus the term $\frac{\pi r^2 \rho'}{3}$ represents the dynamic mass of the vibrating membrane and is $\frac{1}{3}$ that of the static mass.

The fundamental frequency of the simple system in terms of the mass m and the tension T is given by

$$f_{o1} = \left(\frac{T}{2\pi m}\right)^{\frac{1}{2}}$$

Substituting for m the dynamic mass $\frac{\pi r^2 \rho'}{3}$, the fundamental frequency f_{o1} of the membrane is given by

$$
\begin{aligned}
{}_{o1} &= \left\{\frac{3T}{2\pi\,(\pi r^2 \rho')}\right\}^{\frac{1}{2}} \\
&= \left(\frac{3T}{2\pi^2 r^2 \rho'}\right)^{\frac{1}{2}} \\
&= \frac{1}{\pi r}\left(\frac{3T}{2\rho'}\right)^{\frac{1}{2}}
\end{aligned}
$$

Both the frequency of the fundamental resonance and the displacement of the membrane are affected when the membrane is surrounded by air and coupled to a mechanical or an acoustical system.

A.7.3. Clamped Circular Plate

If the restoring force on a clamped circular plate is due solely to bending stiffness, it can be shown, using the notation of the membrane, that the displacement x at any radius y is given by

$$x = x_o\left(1 - \frac{y^2}{r^2}\right)^2$$

The equivalent dynamic mass of the clamped plate is one-fifth of its static mass and the fundamental frequency f_{o1} is given by

$$f_{o1} = \frac{0{\cdot}467t}{r^2}\left[\frac{E}{\rho\,(1-\sigma^2)}\right]^{\frac{1}{2}}$$

where t = thickness of the plate in cm,

r = radius of the clamped boundary in cm,

ρ = density in g/cm³,

σ = Poisson's ratio,

E = Young's modulus in dyn/cm³.

The clamped plate and the stretched membrane exhibit similar vibrational modes, but at different frequencies relative to the fundamental resonance f_{o1}. Some of the less complex modes are shown in Table A.7.1 where the shaded areas in the figures indicate a displacement in the opposite sense to the unshaded sections.

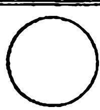
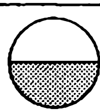
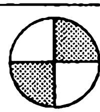
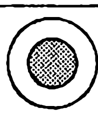
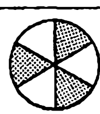
It is instructive to compare the displacement/frequency curve of a thin clamped plate with that of a stretched membrane. The average displacement, plotted against the ratio f/f_{o1} of a clamped circular plate and a clamped circular membrane, is shown in Fig. A.7.3. The parameters are adjusted so that the fundamental resonance is the same for both plate and membrane, and it is also assumed that they have the same mass per unit area.

A.7.4. Membrane

Below $f/f_{o1} = 1$ the curves are very similar in shape but the sensitivity of the membrane shown by the larger displacement amplitude is greater than the sensitivity of the plate. At a frequency

Table A.7.1

VIBRATIONAL MODES OF CIRCULAR MEMBRANES AND PLATES

MODE	CIRCULAR MEMBRANE	CLAMPED CIRCULAR PLATE
	f_{01}	f_{01}
	$1{\cdot}59\ f_{01}$	$2{\cdot}09\ f_{01}$
	$2{\cdot}14\ f_{01}$	$3{\cdot}43\ f_{01}$
	$2{\cdot}30\ f_{01}$	$3{\cdot}91\ f_{01}$
	$2{\cdot}65\ f_{01}$	$4{\cdot}95\ f_{01}$

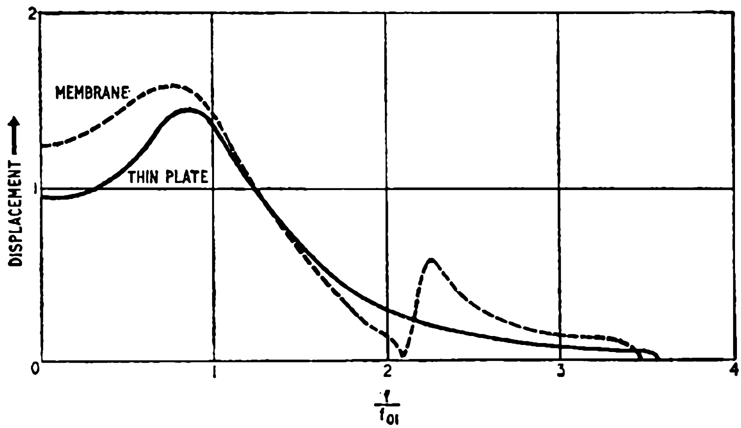


Fig. A.7.3.—Displacement of a membrane and a thin plate plotted against the ratio f/f_{o1}

above $f/f_{o1} = 1$, a circular node forms at the outer edge of the membrane and as the frequency increases, the diameter of the nodal circle decreases. Since the centre of the membrane now moves in anti-phase with the outer portion, the displacements tend to cancel and the average displacement and sensitivity decrease. At $f = 2{\cdot}3 f_{o1}$, complete cancellation occurs and the average displacement is zero.

A.7.5. Plate

Because of the stiffness associated with its clamped edge, the plate has a smaller average displacement than the membrane but it has a larger frequency range below the first frequency of zero response. Unfortunately it is difficult in practice to obtain a plate thin enough to have the same mass per unit area as the membrane, yet stiff enough to have the same fundamental frequency of resonance. Plate diaphragms have been used successfully for miniature electrostatic microphones, but in general diaphragms have a characteristic midway between the stretched membrane and the clamped plate.

Appendix 8

Force on a Completely Exposed Diaphragm[5]

WHEN A DIAPHRAGM, small compared with the wavelength of sound, is exposed to acoustic pressures on both front and rear surfaces, the resultant pressure on the diaphragm depends on the phase as well as the magnitude of the pressures acting on the two faces. In order to obtain an expression for the force on the diaphragm, it will be assumed that the front and rear surfaces are separated one from the other by an acoustic distance *d*, and that the diameter of the diaphragm is small compared to the wavelength so that the pressure acting on a surface is uniform over that surface.

Fig. A.8.1 shows a diaphragm in a plane-wave field at a time when the instantaneous pressure p_i at the acoustic centre of the diaphragm system is

$$p_i = p_m \sin \omega t$$

$$= p_m \sin 2\pi f t$$

Since $f = c/\lambda$ where c is the velocity of sound and λ the wavelength, we may also express the instantaneous pressure p_i at the acoustic centre of the system as

$$p_i = p_m \sin \frac{2\pi}{\lambda} ct$$

$$= p_m \sin kct$$

where $k = \dfrac{2\pi}{\lambda}$ is the wavelength constant.

The product ct represents a distance, being the product of velocity and time. Hence in a plane-wave field the instantaneous distribution of pressure along the axis of propagation is given by

$$p_i = p_m \sin k\,(ct - x)$$

where x is a distance measured from some reference point or plane.

Taking the acoustic centre of the diaphragm system as the reference plane, the back of the diaphragm is at a distance $d/2$ from

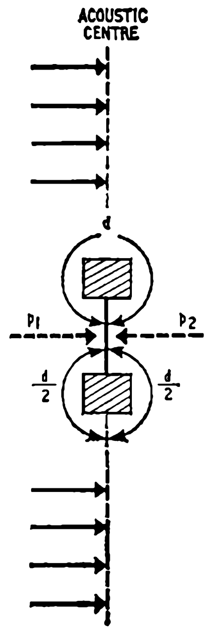


Fig. A.8.1.—Pressures acting on a completely exposed diaphragm

the reference plane measured in the direction of propagation. Hence the pressure p_2 on the rear surface of the diaphragm is given by

$$p_2 = p_m \sin k\left(ct - \frac{d}{2}\right)$$

The front of the diaphragm with reference to the acoustic centre is at a distance $d/2$ measured against the direction of propagation of the wave; hence the pressure p_1 on the front of the diaphragm is given by

$$p_1 = p_m \sin k\left(ct + \frac{d}{2}\right)$$

The resultant pressure p on the diaphragm is $p_1 - p_2$. Hence,

$$p = p_m \sin k\left(ct + \frac{d}{2}\right) - p_m \sin k\left(ct - \frac{d}{2}\right)$$

$$= p_m\left[\sin k\left(ct + \frac{d}{2}\right) - \sin k\left(ct - \frac{d}{2}\right)\right]$$

Since $\sin A - \sin B = 2 \cos \frac{A+B}{2} \sin \frac{A-B}{2}$,

$$p = 2p_m \cos kct \sin k\frac{d}{2}$$

The force F on the diaphragm is pA where A is the area of one side.

Hence, $$F = 2Ap_m \cos kct \sin k\frac{d}{2}$$

Since $k = 2\pi/\lambda$, and $c = f\lambda$ we may write $\cos 2\pi ft$ in place of $\cos kct$, and

$$F = 2Ap_m \cos 2\pi ft \sin k\frac{d}{2}$$

Cos $2\pi ft$ has a maximum positive value of 1·0 when t is any integral multiple of $1/f$. At such instants the force has a positive value, given by

$$F = 2Ap_m \sin k\frac{d}{2}$$

When t is an integral odd multiple of $1/2f$, then $\cos 2\pi ft$ has a negative value $-$ 1·0 and the force F is negative, that is, in the opposite sense or direction.

Appendix 9

Effect of Diffraction on Pressure-Gradient Operation

THE EXTENT to which pressure-gradient operation is affected by diffraction can be illustrated by comparing the response/frequency characteristics of two pressure-gradient systems having equal path lengths at low frequency, but differing at high frequency as the result of diffraction effects.

A.9.1. The Space Phase-shift System

The two pressure units which constitute the space phase system of Fig. A.9.1 (a) are differentially connected and are assumed to have equal sensitivity and an identical response. The units are also assumed to be small compared with the wavelength at all frequencies and are separated by a distance d measured along the axis of propagation. The effective pressures p_1 and p_2 on the units are therefore equal to the free-space pressure p_o and the phase angle ϕ between p_1 and p_2 depends solely on the phase shift introduced by the path length d, that is, $\phi = kd$ radians.

A.9.1.1. BAFFLE SYSTEM

Fig. A.9.1 (b) represents a pressure-gradient system employing two differentially connected pressure units mounted back to back in a thin circular baffle, so that the air path between the two units is equal to the diameter of the baffle. It is assumed that while the diameter of the baffle is small in comparison with the wavelengths at low frequency, it is comparable with or larger than the wavelengths at high frequency.

A.9.1.2. COMPARISON FOR PLANE WAVE AT NORMAL INCIDENCE

For the space phase system the pressures p_1 and p_2 are equal for a plane wave at normal incidence and the phase angle ϕ between

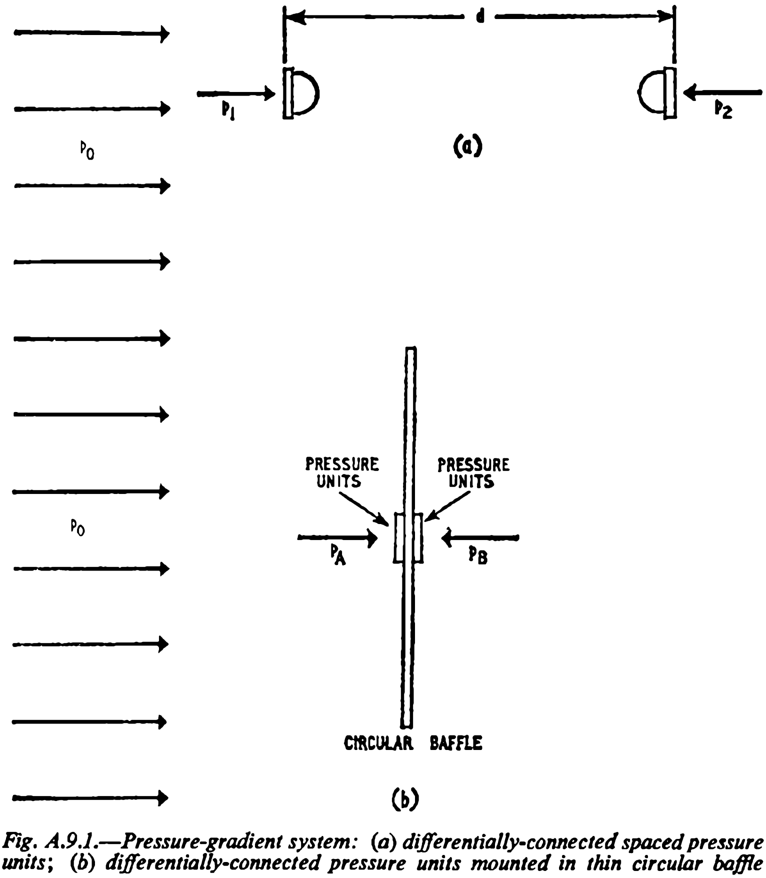


Fig. A.9.1.—Pressure-gradient system: (a) differentially-connected spaced pressure units; (b) differentially-connected pressure units mounted in thin circular baffle

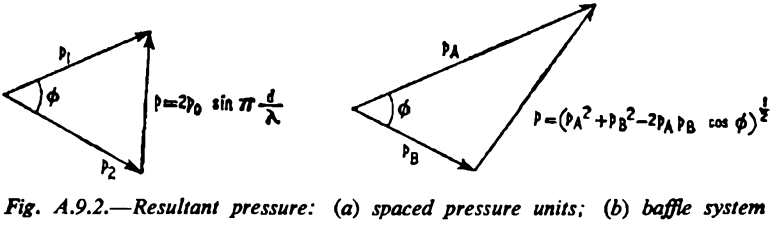


Fig. A.9.2.—Resultant pressure: (a) spaced pressure units; (b) baffle system

p_1 and p_2 is kd radians. The resultant pressure p is the vector difference of p_1 and p_2 and from Fig. A.9.2 (a) is given by

$$p = (p_1{}^2 + p_2{}^2 - 2p_1\, p_2 \cos \phi)^{\frac{1}{2}} \tag{1}$$

Since p_1 and p_2 are equal,

$$p = 2p_o \sin \frac{\pi d}{\lambda} \tag{2}$$

This corresponds to the equation in Appendix 8 for the force on a completely exposed diaphragm. The resultant pressure p obtained from this expression is shown as a function of the ratio d/λ by the dotted curve of Fig. A.9.4.

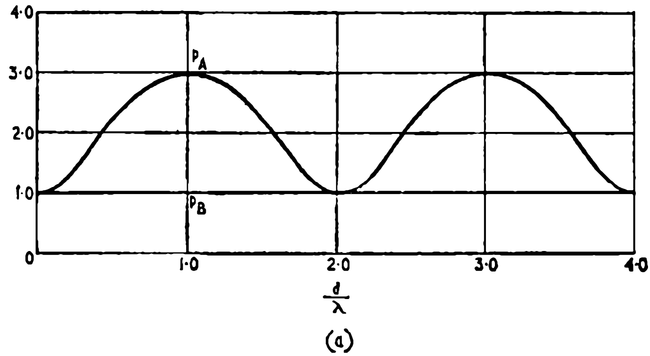
In the baffle system the pressures at front and rear are not equal throughout the operating range, except at very low frequency or at particular frequencies, where the ratio d/λ is an even integer. The variations in the amplitude of p_A and p_B with reference to the free-space pressure p_o are shown in Fig. A.9.3 (a). The pressure p_B on the unit at the back of the baffle is equal to the free-space pressure, hence $\frac{p_B}{p_o} = 1$, but because of obstacle effects, the pressure p_A on the unit mounted in front of the baffle varies with frequency between the values $p_A = p_o$ and $p_A = 3p_o$. Furthermore, the phase angle ϕ between p_A and p_B is not strictly proportional to frequency as in the space phase system.

The phase of the pressures p_A and p_B relative to the free-space pressure p_o is shown in Fig. A.9.3 (b). The phase angle β of p_B relative to p_o is proportional to frequency and is given by $\beta = kr$ where r is the radius of the baffle. The phase of the pressure p_A relative to p_o is not proportional to frequency, but is given by

$$\alpha = \frac{\tan^{-1} \sin kr}{(2 - \cos kr),}$$

as in Fig. A.9.3 (b). It has a maximum value of the order of 30° and may lag or lead the free-space pressure, depending on frequencies, i.e., on the value of d/λ.

The phase angle between p_A and p_B is $\phi = \alpha - \beta$ and may be obtained from Fig. A.9.3 (b). It will be found that it is only at very low frequency ($d/\lambda = 0{\cdot}1$), where α and β are of opposite sign, that the phase angle $\phi = \alpha - \beta$ corresponds to the phase shift associated



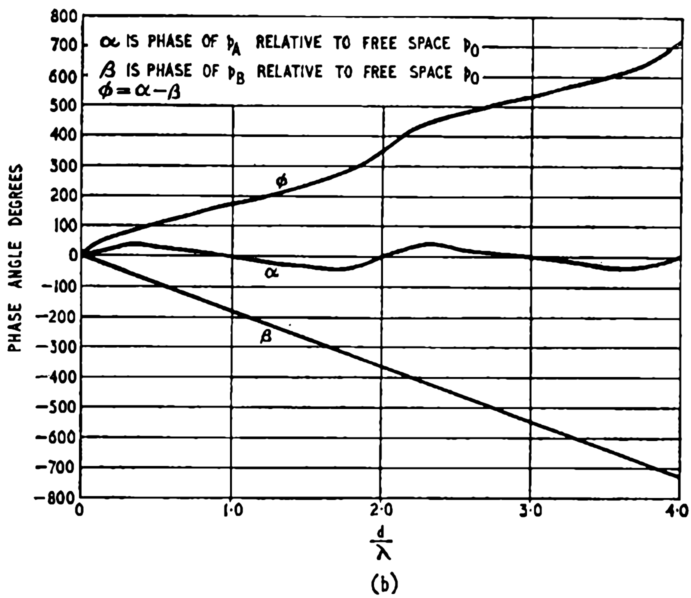


Fig. A.9.3.—(a) Variations in the amplitude of the pressures p_A and p_B plotted against the ratio α/λ; (b) phase angle of p_A and p_B relative to free-space pressure

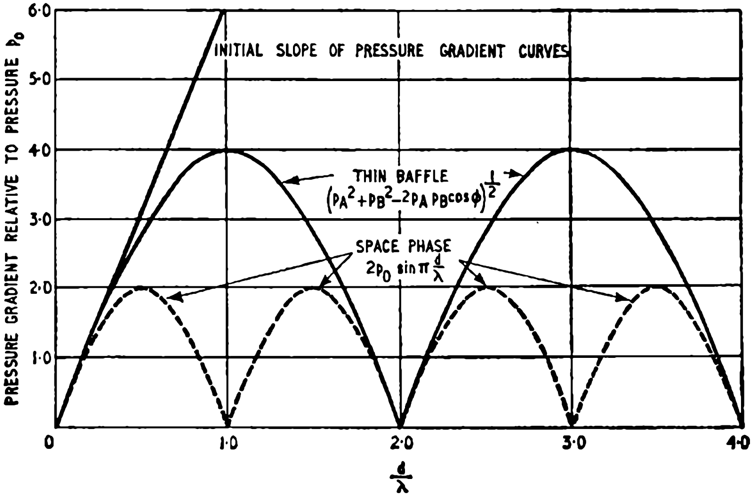


Fig. A.9.4.—Pressure gradient: dotted curve spaced system; full line curve baffle system

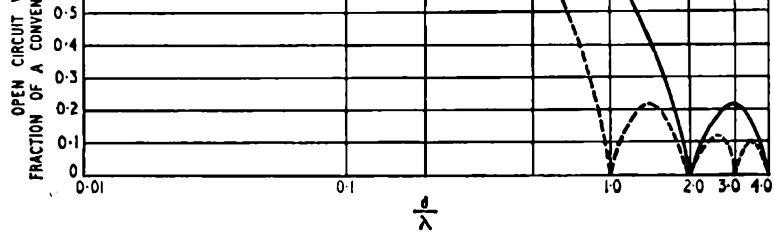


Fig. A.9.5.—Frequency response of spaced and baffle system with a path of length $d = 2r$. At higher frequencies α is only a small fraction of the total phase difference and hence at high frequencies $\phi \simeq \beta = kr$ and the system behaves as if the pressure units were separated by an acoustic distance r rather than by the air path d.

To estimate the resultant pressure p for the baffle system, values of p_A and p_B are obtained from the graphs, together with the appropriate value of the phase angle ϕ and substituted in the expression

$$p = (p_A{}^2 + p_B{}^2 - 2p_A\, p_B \cos \phi)^{\frac{1}{2}}$$

The graph of the resultant pressure is plotted against the ratio d/λ and shows that in the baffle system the maximum value of p is twice the maximum value of the space phase system (Fig. A.9.4).

The two curves are similar in shape at very low frequencies where $d/\lambda < 1$. At these frequencies the pressures p_A and p_B at front and rear of the baffle are approximately equal to the free-space pressure p_o and since $\phi \simeq 2\pi d/\lambda$ at low frequency, the resultant pressures and the initial slopes of both curves are equal.

The slope can be obtained by differentiating the expression for $p = 2p_o \sin \pi d/\lambda$ with respect to d/λ. Differentiating gives

$$2\pi\, p_o \cos \frac{\pi d}{\lambda} \text{ for the slope}$$

At $d/\lambda = 0$ the initial slope is equal to $2\pi\, p_o$. The straight line in Fig. A.9.4 is drawn with a slope of 2π and indicates the manner in which the resultant pressure should increase with frequency in order to obtain a constant output, on the assumption that the pressure-sensitive units are coupled to a mass-controlled system.

The divergence of the curves from the straight line indicates that the output of a mass-controlled system falls when the applied force is no longer proportional to frequency. The height of any ordinate drawn to the straight line which represents the initial slope is $2\pi d/\lambda$, hence the response of space phase system is given by

$$\left(2p_o \sin \frac{\pi d}{\lambda}\right) \Big/ \left(\frac{2\pi d}{\lambda}\right)$$

Similarly the response of the baffle system is given by

$$\left(p_A{}^2 + p_B{}^2 - 2p_A\, p \quad \cos \phi\right)^{\frac{1}{2}} \Big/ \left(\frac{2\pi d}{\lambda}\right)$$

The response of the two systems calculated in this way and plotted against the log of the ratio d/λ is shown in Fig. A.9.5.

It will be seen that the response/frequency range of the baffle system is twice as great as that of the space phase system. The first extinction frequency for the baffle system occurs at $d/\lambda = 2$, that is, when the wavelength is equal to the radius of the circular baffle, while the first extinction frequency for the space phase system occurs at $d/\lambda = 1$. This suggests that with a pressure-gradient microphone having a path length between the diaphragm surfaces which is due mainly to the transverse dimension, a first extinction frequency is to be expected at a wavelength equal to half the lesser frontal dimension.

Appendix 10

First and Second Order Pressure Gradients in Plane and Spherical Waves

AN INDICATION of the characteristics of pressure-gradient microphones in plane-wave and spherical-wave sound fields can be obtained by differentiating the appropriate wave equation. In interpreting the results or in applying them to practical microphones it should be remembered that a pressure-gradient microphone acts as a true differential device only when its path length is very small compared with the wavelength, and while the results obtained by differentiation are valid for the low frequencies, they are inaccurate at high frequency.

A.10.1. First-order Pressure-gradient Microphones

In a spherical wave the pressure is inversely proportional to the distance from the source of sound. If p_{max} is the maximum pressure at the source, the pressure p' at a distance r from the source is

$$p' = \frac{p_{max}}{r}$$

If the pressure varies sinusoidally, the instantaneous pressure p is

$$p = \frac{p_{max}}{r} \cos k\,(ct - r)$$

The pressure gradient is

$$\frac{dp}{dr} = \frac{p_{max}}{r} k \sin k\,(ct - r) - \frac{p_{max}}{r^2} \cos k\,(ct - r)$$

$$= \frac{p_{max}}{r}\left[k \sin k\,(ct - r) - \frac{1}{r} \cos k\,(ct - r)\right]$$

This is a complex wave of the form

$$A \sin \omega t + B \cos \omega t$$

The amplitude of the resultant of these quadrature components is therefore $(A^2 + B^2)^{\frac{1}{2}}$ and the r.m.s. value of the complex wave is given by

$$\text{r.m.s.} = \left(\frac{A^2 + B^2}{2}\right)^{\frac{1}{2}}$$

Hence

$$\frac{\mathrm{d}p}{\mathrm{d}r_{rms}} = \frac{p_{max}}{r}\left(\frac{k^2 + 1/r^2}{2}\right)^{\frac{1}{2}}$$

$$= \frac{p_{max}\, k}{r\sqrt{2}}\left(1 + \frac{1}{r^2 k^2}\right)^{\frac{1}{2}}$$

In a plane wave of the same amplitude $\left(p' = \frac{p_{max}}{r}\right)$ the instantaneous pressure p is

$$p = p' \cos k\,(ct - r)$$

The pressure gradient is

$$\frac{\mathrm{d}p}{\mathrm{d}r} = p'\, k \sin k\,(ct - r)$$

$$= \frac{p_{max}}{r} k \sin k\,(ct - r)$$

$$\frac{\mathrm{d}p}{\mathrm{d}r_{rms}} = \frac{p_{max}\, k}{r\sqrt{2}}$$

The ratio of the pressure gradients is

$$\frac{\text{pressure gradient r.m.s. spherical wave}}{\text{pressure gradient r.m.s. plane wave}}$$

$$= \frac{\dfrac{p_{max}\, k}{r\sqrt{2}}\left(1 + \dfrac{1}{r^2 k^2}\right)^{\frac{1}{2}}}{\dfrac{p_{max}\, k}{r\sqrt{2}}}$$

$$= \left(1 + \frac{1}{r^2 k^2}\right)^{\frac{1}{2}}$$

Thus the pressure gradient in a spherical wave is greater than the pressure gradient in a plane wave by a factor

$$\left[1+\left(\frac{c}{2\pi fr}\right)^2\right]^{\frac{1}{2}}$$

This factor is a measure of the discrimination exercised by the microphone in favour of the spherical-wave source (close speech) and against the plane-wave source (ambient noise) on the assumption that the path length of the microphone $d \ll \lambda$.

If the noise is random in incidence and the microphone has a bi-directional characteristic, then the average discrimination is

$$\sqrt{3}\left(1+\frac{1}{k^2r^2}\right)^{\frac{1}{2}} \text{ in favour of the point source.}$$

A.10.2. Second-order Pressure-gradient Microphones

Second-order pressure-gradient microphones, especially in their present stage of development, are unlikely to be used generally in broadcasting but because of their directional characteristics and their ability to discriminate against a high level of ambient noise, they may eventually be used to advantage in difficult acoustic conditions.

An indication of their characteristics can be obtained by differentiating

$$\frac{dp}{dr} = \frac{p_{max}}{r} k \sin k\,(ct - r) - \frac{p_{max}}{r^2} \cos k\,(ct - r) \text{ with respect to } r$$

$$\therefore \frac{d^2p}{dr^2} = \frac{-p_{max}}{r} k^2 \cos k\,(ct - r) - \frac{p_{max}}{r^2} k \sin k\,(ct - r) + $$
$$+ \frac{2p_{max}}{r^3} \cos k\,(ct - r) - \frac{p_{max}}{r^2} k \sin k\,(ct - r)$$

$$= \frac{-p_{max}}{r} k^2 \cos k\,(ct - r) - \frac{2p_{max}}{r^2} k \sin k\,(ct - r) + $$
$$+ \frac{2p_{max}}{r^3} \cos k\,(ct - r)$$

Rearranging the equation,

$$\frac{d^2p}{dr^2} = \left(\frac{2p_{max}}{r^3} - \frac{p_{max}\,k^2}{r}\right)\cos k\,(ct - r) - \frac{2p_{max}\,k}{r^2}\sin k\,(ct - r)$$

This is a complex wave and its r.m.s. value is the root of the sum of the squares of the r.m.s. values of its components, i.e.,

$$\left(\frac{A^2 + B^2}{2}\right)^{\frac{1}{2}}$$

Hence $$\frac{d^2p}{dr^2}_{rms} = \left[\frac{\left(\frac{2p_{max}}{r^3} - \frac{p_{max}\,k^2}{r}\right)^2 + \left(\frac{2p_{max}\,k}{r^2}\right)^2}{2}\right]^{\frac{1}{2}}$$

$$= \frac{p_{max}}{\sqrt{2}}\left(\frac{4}{r^6} + \frac{k^4r^4}{r^6} - \frac{4k^2}{r^4} + \frac{4k^2}{r^4}\right)^{\frac{1}{2}}$$

$$= \frac{p_{max}}{r^3\sqrt{2}}\,(4 + k^4r^4)^{\frac{1}{2}}$$

Now differentiating $\frac{dp}{dr} = p'k \sin k\,(ct - r)$ with respect to r, we have

$$\frac{d^2p}{dr^2} = p'k^2 \cos k\,(ct - r)$$

Assuming that the pressure in the plane wave $p' = \frac{p_{max}}{r}$,

$$\frac{d^2p}{dr^2} = \frac{p_{max}}{r}\,k^2 \cos k\,(ct - r)$$

$$\frac{d^2p}{dr^2}_{rms} = \frac{p_{max}\,k^2}{r/2}$$

Taking the ratio of the r.m.s. values of the second order of the pressure gradients,

$$\frac{\dfrac{p_{max}}{r^3\sqrt{2}}(4 + k^4r^4)^{\frac{1}{2}}}{\dfrac{p_{max}\,k^2}{r\sqrt{2}}}$$

$$= \frac{1}{k^2r^2}(4 + k^4r^4)^{\frac{1}{2}}$$

$$= \left(1 + \frac{4}{k^4r^4}\right)^{\frac{1}{2}}$$

The ratio is a measure of the discrimination which second-order pressure-gradient microphones exercise against ambient noise (plane wave) and in favour of close speech (spherical wave). It is accurate only for the low frequencies, when the path lengths of the microphones forming the combination and the spacing between the units are small compared with the wavelength.

If the ambient noise is random in incidence and if the directional characteristic of the microphone is proportional to $\cos^2\theta$, then the discrimination is

$$\sqrt{5}\left(1 + \frac{4}{k^4r^4}\right)^{\frac{1}{2}}$$

Appendix 11

Acoustical Phase-shift Networks[3]

THE VECTOR ANALYSIS of the force on the diaphragm of a phase-shift microphone (Section 9.5) showed that if a cardioid directional characteristic is required, the internal path d_i must introduce a phase shift proportional to frequency ($2\pi\frac{f}{c}d_i$ radians) and equal in magnitude to the phase shift provided by the external path d_e. The equivalent analogous circuit of diaphragm and phase-shifting network in generalised terms is shown in Fig. A.11.1.

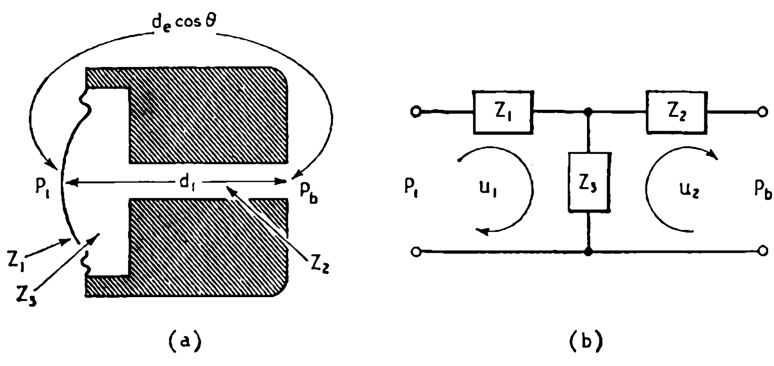


Fig. A.11.1.—(a) Schematic sketch of acoustical phase-shift network; (b) equivalent electrical circuit

Z_1 represents the impedance of the diaphragm;
Z_2 represents the impedance of the narrow slit or rear opening;
Z_3 represents the impedance backing the diaphragm;
p_1 is the pressure on the front surface of the diaphragm;
p_b the pressure at the opening of the narrow slit remote from the diaphragm.

p_1 and p_b are assumed to be of equal magnitude, but they differ in phase because of the shift introduced by the external path $d_e \cos\theta$.

Z_2 and Z_3 in combination provide the phase shift we have previously associated with the internal path (d_i).

Writing down the network equations we have

$$p_1 = u_1 (Z_1 + Z_3) - u_2 Z_3 \qquad (1)$$

$$-p_b = -u_1 Z_3 + u_2 (Z_2 + Z_3) \qquad (2)$$

From Equation 2 $\quad u_2 = \dfrac{u_1 Z_3 - p_b}{Z_2 + Z_3}$

Substituting this value for u_2 in Equation 1, we have

$$p_1 = u_1 (Z_1 + Z_3) - \left(\frac{u_1 Z_3 - p_b}{Z_2 + Z_3}\right) Z_3$$

$$p_1 (Z_2 + Z_3) = u_1 (Z_1 + Z_3)(Z_2 + Z_3) - u_1 Z_3^2 + p_b Z_3$$

$$p_1 (Z_2 + Z_3) = u_1 (Z_1 Z_2 + Z_1 Z_3 + Z_2 Z_3 + Z_3^2) - - u_1 Z_3^2 + p_b Z_3$$

$$u_1 = \frac{p_1 (Z_2 + Z_3) - p_b Z_3}{Z_1 Z_2 + Z_1 Z_3 + Z_2 Z_3} \qquad (3)$$

p_b is of the same amplitude as p_1, but because of the phase shift introduced by the external path, it lags on p_1 by kd radians $= \omega d_e/c$. The effective length of the external path d_e depends on the angle of incidence and at any angle θ is $d_e \cos\theta$. So the phase of p_b relative to p_1 is $\dfrac{\omega}{c} d_e \cos\theta$.

But $\qquad p_b = p_1 e^{j(-\phi)}$

$$= p_1 [\cos(-\phi) + j \sin(-\phi)]$$

where ϕ is $\omega d_e/c \cos\theta$.

If d_e is small compared with the wavelength, $\omega d_e/c$ is a small angle; $\cos\phi \simeq 1$ and $\sin\phi \simeq \phi$.

Then $p_b \simeq p_1 \left[1 - j\dfrac{\omega}{c} d_e \cos\theta\right]$

Substituting in Equation 3,

$$u_1 = \frac{p_1 (Z_2 + Z_3) - p_1 Z_3 (1 - j\frac{\omega}{c} d_e \cos\theta)}{Z_1 Z_2 + Z_1 Z_3 + Z_2 Z_3}$$

$$= \frac{p_1 \left[Z_2 + Z_3 \left(j\frac{\omega}{c}\, d_e \cos\theta \right) \right]}{Z_1 Z_2 + Z_1 Z_3 + Z_2 Z_3}$$

$$= \frac{p_1}{Z_1 Z_2 + Z_1 Z_3 + Z_2 Z_3} \cdot \left[Z_2 + Z_3 \left(j\frac{\omega}{c}\, d_e \cos\theta \right) \right]$$

Z_1, the impedance of the diaphragm, affects only the frequency response of the microphone; d_e, Z_2 and Z_3 affect both sensitivity and directional characteristic.

The factor

$$Z_2 + Z_3 \left(j\frac{\omega}{c}\, d_e \cos\theta \right)$$

determines the directional pattern. We see that if

$$Z_2 = Z_3\, j\frac{\omega}{c}\, d_e$$

the equation has the form

$$(a + b\cos\theta)$$

and the directional characteristic is a cardioid if

$$\frac{Z_3\, j\, \omega d_e}{c\, Z_2} = 1$$

If Z_3 is a compliance C_{m3}, its reactance is $1/j\,\omega C_{m3}$; then

$$\frac{j\,\omega d_e}{c\, Z_2\, j\,\omega C_{m3}} = 1$$

and

$$Z_2 = \frac{d_e}{c C_{m3}}$$

This indicates that if a cardioid characteristic is desired, Z_2 is a resistance of the value d_e/cC_{m3}. If, however, Z_2 is a mass m_2, then Z_3 is a resistance R_3 of value

$$R_3 = \frac{cm_2}{d_e}$$

Appendix 12

Frequency Range and Power[6,7,8,9]

MICROPHONES used in high-quality sound-reproducing systems should be uniformly sensitive over the frequency range and the range should be wide enough to include all frequency components present in the original sound. A uniform response from 100 c/s to 8,000 c/s is sufficient to reproduce natural speech, but for satisfactory reproduction of music the range would have to be extended below 100 c/s and above 8,000 c/s. For complete realism the frequency range of the microphone and reproducing system must include not only the musical range of the instrument itself but also the frequencies of any noise characteristic of the instrument or any noise associated with its operation. The origin of the noise may be mechanical or acoustical and take the form of key clicks or the hissing noise produced by air streams.

Table A.12.1 lists the frequency range of some musical instruments and certain familiar sounds; it will be seen that the range extends from 40 c/s to 17,000 c/s. If the low notes of the piano or organ are to be reproduced directly, the range must extend to bass frequencies

Table A.12.1.

FREQUENCY RANGES OF SOME MUSICAL INSTRUMENTS AND FAMILIAR SOUNDS

Instrument	*Frequency range*
Bass viol	40–6,000 c/s
Timpani*	50–3,000 c/s
Bass drum*	50–2,000 c/s
Snare drum	80–15,000 c/s
14 in. cymbals	400–14,000 c/s
Footsteps	80–13,000 c/s
Key jingling	700–17,000 c/s
Male speech	100–9,000 c/s
Female speech	250–10,000 c/s

* Noises associated with the bass drum and the timpani extend the range of the instrument to 6,000 c/s and 5,000 c/s respectively.

of the order of 16 c/s. Transducers, both loudspeakers and microphones, to cover such a wide range are extremely difficult to produce.

A.12.1. Restricted Frequency Range

If the sound-reproducing system has a restricted frequency coverage, then the most pleasing reproduction of music is obtained when the range extends by an equal number of octaves above and below a mean frequency of 800 c/s. If the system is intended for speech only, the mean frequency is about an octave above the optimum for music. Articulation tests have shown that frequencies below 500 c/s add little by way of intelligibility to speech; the loss of these low frequencies reduces intelligibility by only 2 %, but restricting the frequency range below or above 1,500 c/s results in a loss of intelligibility of some 35 %. The frequency of 1,500 c/s is therefore selected as the mean frequency for intelligibility in systems intended for speech.

A.12.2. Power in Speech

Power in speech is dependent on the level and varies widely from 0·01 μW for a whisper to 1,000 μW for a shout, a range of 50 dB. For normal speech intensities, the pressure on the diaphragm of a

Table A.12.2.

PEAK POWER OF MUSICAL INSTRUMENTS*

Origin of sound	*Power (watts)*
Orchestra of 75 performers at loudest	75
Bass drum at loudest	25
Pipe organ at loudest	13
Trombone at loudest	6
Piano at loudest	0·4
Trumpet at loudest	0·3
Orchestra of 75 performers at average	0·09
Piccolo at loudest	0·08
Clarinet at loudest	0·05
Violin at softest used in concert	0·0000038

* By courtesy of Bell Telephone Laboratories.

microphone situated at a distance of 30 cm (or 12 in.) from a speaker is of the order of 1 dyn/cm^2 and at normal conversational levels the average speech power varies from 10 μW to 16 μW, depending on the method of measurement.

The peak power of certain musical instruments is large in comparison with the human voice. Amongst the most powerful of the directly-operated instruments in the percussion section is the bass drum, while the trombone is the most powerful in the brass section. Table A.12.2 shows the results of an investigation by the Bell Telephone Laboratories of the peak power of musical instruments. It will be seen that the volume range from a solo violin playing softly to the large orchestra playing double forte is over 70 dB. If the reproducing system were adjusted so that weak sounds could be heard above the general noise level in a living room, then the peak power of the orchestra would be intolerable. For this reason and for reasons explained in the Training Manual for Sound and Television Broadcasting,[10] the volume range in broadcasting is restricted to about 24 dB.

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