

Microphones

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Introduction

A microphone is an *electroacoustic transducer*. When located in a sound field, its output is an electrical signal that reproduces the sound pressure variations that it senses. There are two fundamental types of microphones, *passive* and *active*. The electrical power output of a passive transducer is derived solely from the acoustic power it absorbs, while an active transducer controls an external source of power.

The first application of a microphone was as a telephone transmitter. Bell first tried a passive transducer, simply his receiver used in reverse, but its output was far too feeble for practical use. Then he used a liquid transmitter, with a fine point immersed in a conducting liquid. This was an active transducer, and provided the necessary power by controlling the current from an external battery. The carbon microphone, perfected by Edison, soon became the preferred transmitter, and was used until very recently. It is an active transducer, supplying about a thousand times more electrical power than the acoustical power it absorbs.

The word "transmitter" was originally used for the telephone transducer, and is still so used in telephone technology. The many active transducers proposed as telephone transmitters that used microscopic contacts were dubbed "microphones" by Hughes. In radio, the complete transmitting apparatus is called the "transmitter," so "microphone" was adopted for the transducer in radio to avoid confusion. This usage has become so general that "microphone" is now the word for a general electroacoustic transducer of any type, except with telephones.

A passive transducer is strictly limited by the conservation of energy. It would seem that by using stronger magnets in the Bell receiver used as a transmitter, more energy could be extracted, but magnets are not a source of energy, so all attempts to do this were utterly defeated. There is always some reaction that keeps the energy accounts straight. Most passive transducers have the property that they are reversible: if a transducer converts acoustic energy into electrical energy, the same transducer will convert electrical energy into acoustical energy. This is illustrated by intercom systems that use a moving-coil loudspeaker as a microphone as well. Active transducers are not reversible.

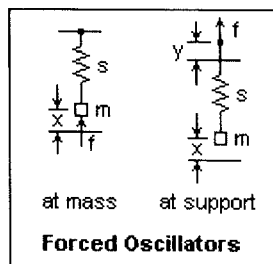
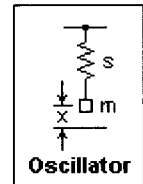
We must deal with three interconnected elements to understand microphones. First is the acoustic input, sound waves in air. These interact with a mechanical system, usually a diaphragm, to excite motion in solid bodies. Finally, the mechanical system interacts electrically to create an electrical generator. Therefore, we shall take up these elements one by one in what follows. There is a large amount of interesting physics and engineering in microphone design.

Vibrations and Oscillators

The diaphragm of a microphone is a mechanical system that vibrates under the influence of the sound waves that reach it. The operation of a microphone is very greatly affected by the motion of the diaphragm, sometimes influenced by air volumes and passages behind it. Therefore, we begin by a thorough examination of the vibrations of mechanical systems.

Small vibrations in gases, liquids and solids are described approximately by linear equations, so that the principle of superposition holds, usually to a very high degree. This means that we can build up any vibration from a superposition of harmonic, or sinusoidal, vibrations. This is a very powerful method in electric circuits, with which the reader is probably quite familiar. The vibrations of even complex systems can be analyzed in terms of *normal modes*, each representing a harmonic vibration of a definite frequency. We shall generally specify frequency as the ordinary frequency f in Hz, or as the angular frequency $\omega = 2\pi f$ in s^{-1} , calling them both frequency, but identifying which is meant by the units.

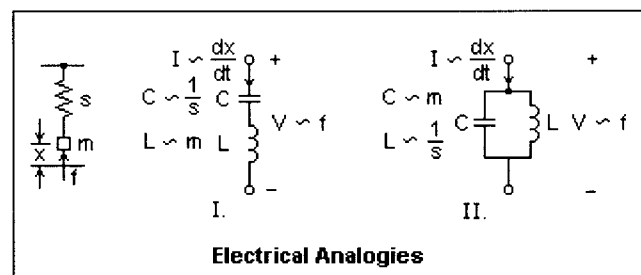
Consider an *oscillator* consisting of a mass m grams and a spring of stiffness s dyne/cm. Let x be the departure of the position of the mass from its equilibrium position in a certain fixed direction. If the mass m hangs from the spring, there will be a certain average displacement $x_0 = mg/s$. We will neglect this displacement in what follows, since it does not affect any of our results, and is only mentioned to avoid confusion in mental pictures. If a positive x compresses the spring, then the force on m will be $-sx$ and the equation of motion will be $m(d^2x/dt^2) = -sx$. If primes stand for the time derivatives, then the equation of motion is $x'' + (s/m)x = 0$, which is linear. It is also very familiar and easily solved in exponentials, so that $x = A \sin \omega_0 t + B \cos \omega_0 t$, where A and B are arbitrary real constants and $\omega_0^2 = s/m$. We can write this instead as $x = Ae^{j\omega t}$, where A is now a complex constant, or a *phasor*, and we agree to take the real part as our solution. Either way we have the necessary two arbitrary constants for a general solution that will match any boundary conditions (x, x' at $t = 0$, say).



Such an oscillator can be driven, or *forced*, by an external force $f(t)$ applied to the mass, so that the force acting on it in the direction of x is $f - sx$. Forced oscillators are shown in the figure. The external force can be applied either to the mass, or to the point of support. The equation of motion is now $mx'' + sx = f$. Let us assume that x and f depend on time through $e^{j\omega t}$, where ω is an arbitrary frequency. Then, we find $-m\omega^2 + sx = f$, where x and f are now phasors, so that $x = f/(-m\omega^2 + s)$. The velocity is $v = j\omega x$, so $v = f/j(m\omega - s/\omega)$, or $f = v[j(m\omega - s/\omega)]$. The quantity in square brackets is called the *mechanical*

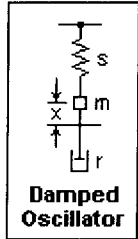
impedance Z' , by analogy to electric circuits, where f corresponds to the voltage V , and v to the current I , and $V = IZ'$. In this analogy, m corresponds to the inductance L and s to the inverse capacitance $1/C$. Then, $\omega_0 = 1/\sqrt{LC}$. The oscillator can be likened to a series circuit driven by a voltage V across it, with the velocity corresponding to the resulting current. This can be a valuable analogy for studying vibrations, or by simulating a vibrating system by an electrical circuit. An oscillator driven at the mass acts like a series resonant circuit, with the velocity of the mass a maximum at resonance.

This is not the only possible analogy. If we assume that I corresponds to the force f , and V to the velocity v , then we have $I = VY$, where Y is the admittance. If m corresponds to C and s to $1/L$, then we have $I = V[j(\omega C - 1/\omega L)]$. Now the equivalent circuit is the parallel combination of L and C across which there is the voltage V , giving rise to a current I . This is just as valid as the previous analogy, but we shall stick mainly to the previous analogy.



Another way to force the oscillator is not to apply a force to the mass, but to move the point of support by an amount y , as shown in the figure above. Now the force on the mass will be $f = s(y - x)$, and the equation of motion will be $mx'' + sx = sy$, or $x''/s + x/m = y/m$. Now, substituting the time dependence $e^{j\omega t}$ and solving for the phasor x , we have $x = sy/(s - m\omega^2)$. If we use this to find $f = s(y - x)$, we find that $f = -m\omega^2 x$. We can equally well consider f as a force applied to the point of support to drive the oscillator. Using the

expression for f in terms of x to replace x in the solution of the equation, we have $-(1/m\omega - \omega/s)f = \omega y$, or $j\omega y = [j(\omega/s - 1/m\omega)] = j\omega y = v$, where v is now the velocity of the point of support, not the mass m . Applying the electrical analogy, with $I = j\omega y$ and $V = f$, we have $I = VY$, where $Y = j(\omega C - 1/\omega L)$, the admittance of the parallel combination of C and L . That is, when driven at the point of support by a force f , an oscillator looks like a parallel resonant circuit, so that at resonance, the velocity of the point of support is a minimum (zero in the ideal case). If we use the second analogy instead, we find a series circuit in this case, as we might expect.

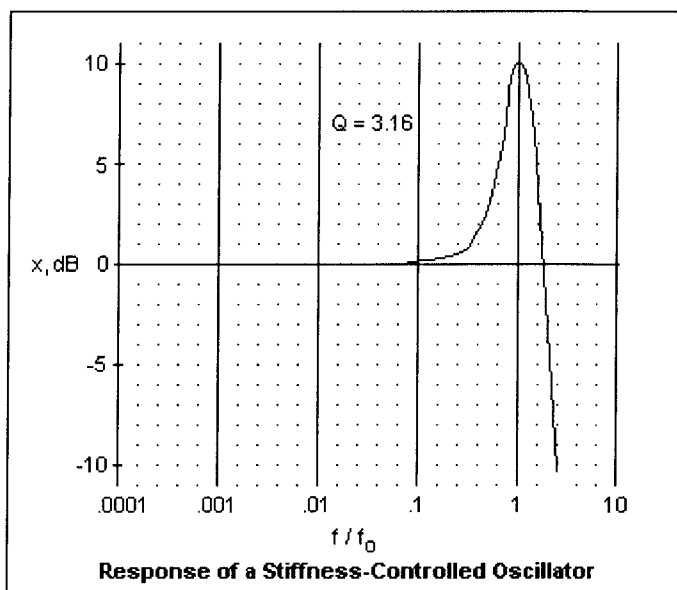


So far we have neglected frictional dissipation in the oscillator. If we add a force proportional to velocity, $-rx'$, the equation of motion becomes $mx'' + rx' + sx = f$. The frictional force is represented in the diagram by a dashpot below the mass. When exponential time dependence is introduced, we find $(-m\omega^2 + j\omega r + s)x = f$, or $v = j\omega x = f/[r + j(\omega m - s/\omega)]$. A real component r has been added to the mechanical impedance. Mechanical impedance Z' relates force f and velocity v by $f = Z'v$, so the dimensions of Z' are dyne-s/cm or g/s. In the MKS system, this is kg/s, of course. r has the same dimensions, g/s while stiffness s has dimensions g/s^2 . At resonance, we see that $v = f/r$, so that v and f are in phase and proportional. A little frictional resistance removes the infinities at resonance, so the quantities vary smoothly through this region.

The Q , or quality factor, of the oscillator is the dimensionless combination $Q = \omega_0 m / r = \sqrt{sm}/r$, containing all three parameters. When Q is larger than 1, the resonance curve of v vs. ω is more or less sharply peaked. When Q is smaller than 1, the resonance is not pronounced. When Q is large, it is very closely $\omega_0/\Delta\omega$, where $\Delta\omega$ is the frequency difference between the half-amplitude points of the resonance curve. It is also the ratio of the energy stored in the oscillator to the energy dissipation per cycle.

We found that the natural vibration of an oscillator with $r = 0$ was a linear combination of $e^{j\omega t}$ and $e^{-j\omega t}$, where ω was the natural frequency $\sqrt{s/m}$. When r is greater than zero, if we try a solution of the form e^α , we find that α must satisfy the equation $m\alpha^2 + r\alpha + s = 0$. Solution by the quadratic formula gives $\alpha = -(r/2m)[1 \pm j\sqrt{4Q^2 - 1}]$ after a little algebra to show how Q enters in this expression. If $Q > 1/2$, we find two exponentially-damped solutions of a frequency slightly different from ω_0 . If $Q < 1/2$, we find a linear combination of two exponentially decreasing functions, with no oscillation at all. If $Q = 1/2$, we have a special case called *critical damping* in which $r = 2\sqrt{sm}$. Not only do we have one solution $e^{-\omega t}$, but $te^{-\omega t}$ is also a solution. The general solution is then $(A + Bt)e^{-\omega t}$. Although this exact form occurs only on the boundary, it is a good approximation in the neighborhood of $Q = 1/2$. The general solution for the forced oscillator is the sum of a solution of the unforced oscillator at the natural frequency, and the special forced response we have previously found at the forcing frequency. The first term is called the *transient solution* and decays with time until finally only the forced oscillation remains. All of these familiar phenomena occur with microphones and loudspeakers.

We generally want the sensitivity of a microphone to be independent of frequency, or "flat." If the microphone is sensitive to pressure, then a force f drives the diaphragm, and we want the displacement $x = v/j\omega$ to be independent of frequency, or v to be proportional to frequency. Since $v = f/[r + j(\omega m - s/\omega)]$ this will happen, at least approximately, if r and $m\omega$ are much less than s/ω . In that case, $v = -j\omega f/s$, which is what we desire. An oscillator that behaves in this way is called *stiffness controlled*. The response of a stiffness-controlled oscillator is shown at the right. The response as $20 \log [x/(f/s)]$ is plotted against $u = f/f_0$, where f_0 is the resonant frequency of the oscillator. The expression for the magnitude of the response



x is $x = (f/s)[(1 - u^2)^2 + (u/Q)^2]^{-1/2}$, which is easily derived from the equations above. This oscillator has $Q = 3.16$. It is not hard to see that any oscillator is stiffness controlled at frequencies much less than its natural frequency. The response is essentially level for frequencies less than a tenth of the resonant frequency. This is the reason that the resonant frequency of the diaphragm of a microphone is made greater than the maximum frequency at which the microphone will be used. The diaphragm of a telephone transmitter to be used from 300 Hz to 3000 Hz may have a resonant frequency of 10,000 Hz for this reason. The resonant frequency is raised by making the diaphragm stiff and light. It should be noted that increased stiffness s means less response.

A microphone whose output depends on the velocity of the response, such as a moving-coil or dynamic microphone, or a ribbon microphone, is made flat by operating near the resonant frequency with a low Q . In this case, $v = f/r$ is independent of frequency. Here, too, we have a trade-off, since increased r means decreased output. An oscillator operated in this range is called *resistance controlled*.

An oscillator driven well above its resonant frequency is called *mass-controlled*, since in this case $v = f/\omega m$. In this region, the displacement drops off very rapidly with frequency, so it is not useful for a pressure microphone. The acceleration $a = \omega v$ is the constant quantity in this region. This may sometimes be desired, but not in a microphone.

Before leaving vibrations and oscillators, let's look at modelling an oscillator using the electrical analogy. Most texts mention that this can be done, but give no examples of how. Assume we have an oscillator with $m = 100$ g and $s = 10^6$ dyne/cm (about 1 kg per cm). We will neglect the damping in this case, but it can easily be added. The natural frequency of this oscillator is 100 s⁻¹, relatively low for electrical simulation, but possible. First, we choose the values of L and C to be used in the circuit. Let us try $L = 1$ H. Since the resonant frequency must be the same as for the mechanical oscillator, $C = 100$ μ F. It is possible to scale the frequency, but we will not get into the complications involved and stick with a direct analogy. Since velocity will be the analogue of current, let us choose 1 mA to represent 10 cm/s. The force exerted on the 100 g by an acceleration of 10 cm/s² will be 10^3 dynes. The voltage induced in L for a current change of 1 mA/s will be 10^{-3} V. Therefore, 1 mV corresponds to 10^3 dynes, or 1 V to 10^6 dynes.

To see that this works, suppose we apply a driving force with an amplitude of 10^6 dynes and a frequency of 50 s⁻¹ to the oscillator. The displacement x will be $10^6/[-(2500)(100) + 10^6] = 1.33$ cm. [2500 is $\omega^2 = (50)^2$.] The velocity amplitude will be $j\omega x = j66.7$ cm/s. Now we consider the electrical circuit. We connect 1 H and 100 μ F in series, and apply 1 V peak at 50 s⁻¹ to them. $Z = j(50 - 1/50 \times 10^{-4}) = -j150$ Ω . The current I will be $1/-j150 = j6.67$ mA. Since 1 mA corresponds to 10 cm/s, the analogous velocity of the mechanical oscillator will be $j66.7$ cm/s, which we have already determined. We can use the circuit to predict the behavior of the mechanical oscillator for any magnitude and frequency of applied force.

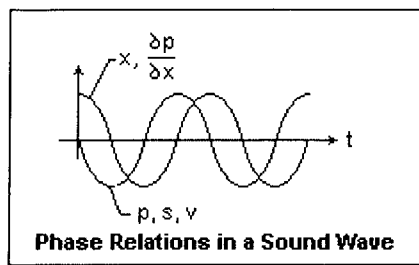
If we take $L = 10$ mH and $C = 1$ μ F instead, the resonant frequency will be $10,000$ s⁻¹. If we apply 1 V at 5000 s⁻¹, Z will still be $-j150\Omega$, and I will still be 6.67 mA, which gives the proper result, 66.7 cm/s. This illustrates how the frequency may be scaled to put the simulation into a more convenient frequency range. 5000 s⁻¹ is 796 Hz, much more convenient than 7.96 Hz.

Sound Waves

Sound waves are longitudinal scalar waves in air. The important quantities are the displacement x , the velocity v , and the overpressure p . The air is treated as a continuum, and x is its displacement and v is its velocity, often called the *particle velocity* to distinguish it from the phase velocity of acoustic waves. The direction of x and v is normal to the wavefront, in the direction of propagation. All of these quantities are exceedingly small for sound of normal intensities, allowing the equations of motion to be linearized to high accuracy. Sound waves obey the principle of superposition, and harmonic (sinusoidal) waves are the basis of analysis. Sound waves are treated in detail in the article [Sound Waves](#).

The density of air at 0°C and 1 atm pressure (STP) is 1.2926 g/cm³. 1 atm is 1.01325×10^6 dyne/cm². Dry air obeys the ideal gas law with a molecular weight $M = 28.97$, and the ratio of the specific heats is 1.402. The phase velocity of sound under these conditions is $c = \sqrt{(\gamma p/\rho)} = \sqrt{(\gamma RT/M)} = 331.5$ m/s. At 20°C, $c =$

343.4 m/s. Since the speed of sound is independent of frequency, propagation is nondispersive, and the group and energy velocities are the same as the phase velocity.



The relations between the quantities in a harmonic plane wave of displacement $x = Ae^{j(\omega t - kz)}$, where ω is the angular frequency, k is the wave vector $2\pi/\lambda$, and $\omega/k = c$. A is an arbitrary complex amplitude, are easily expressed. The particle velocity $v = j\omega x$, and the condensation $s = \Delta\rho/\rho = jkx$. The overpressure $p = j\gamma p_0 s = (\gamma p_0/c) v = rv$. The quantity r connecting overpressure and velocity is the *acoustic impedance* of air. For air at STP, $r = 42.6$ dyne-s/cm³ or g/cm-s. The power in a sound wave is expressed in

terms of the overpressure p by $P = p^2/2r$. The phase relations between these quantities as a function of time at a fixed point are shown in the diagram. $\partial p/\partial x$ is shown for k positive; for k negative, it is multiplied by -1. The other quantities are independent of the sign of k (direction of wave).

An overpressure of 10 μ bar (a μ bar is just a dyne/cm²) or 1 Pa (N/m²) makes a rather strong sound wave. However, p/p_0 is still only about 10^{-5} . The corresponding condensation, or fractional change in density is $s = p/j\gamma p_0$, and even smaller. The particle velocity is $p/r = 10/42.6 = 0.235$ cm/s, much less than $c = 33150$ cm/s. The particle velocity is in phase with the overpressure, but the condensation is in quadrature. Finally, the displacement $x = v/j\omega$ is in phase with the condensation, but in quadrature with the pressure. At 1000 Hz, $\omega = 6283$ s⁻¹, so the magnitude of x will be 3.74×10^{-5} cm, only about 0.4 μ m! The energy flow in the wave is 1.17 erg/s/cm², or 0.116 μ W/cm². Sound is a very small disturbance of air, and it is a marvel that it can be detected by ears and microphones at all.

The threshold of hearing at 3000 Hz is an overpressure of 2×10^{-4} dyne/cm². Hearing is less sensitive at lower and higher frequencies. Since we are dealing with many orders of magnitude, sound intensity is expressed logarithmically, in decibels. The *sound pressure level* (SPL) is dB re 2×10^{-4} dyne/cm², or $\text{SPL} = 20 \log (p/2 \times 10^{-4})$. Normal conversation is carried out at SPL 50 to 60 dB. An SPL of 60 dB corresponds to $p = 0.2$ dyne/cm². An SPL of 120 dB causes discomfort; it corresponds to $p = 200$ dyne/cm². Even at this intensity, the acoustic energy flow is only 0.469 mW/cm². The apparent loudness of a sound increases logarithmically with energy intensity, giving the aural sense a large dynamic range. This is characteristic of all the senses, and is known as Fechner's Law.

The wavelength at 1000 Hz is 33.1 cm, or a little over a foot, something worth remembering. Most microphones, especially the ones popular today, are rather small, and do not disturb the sound field greatly. When the wavelength approaches the size of the microphone, diffraction effects occur that change the distribution of the pressure at the surface of the microphone. For a spherical microphone, diffraction about doubles the overpressure at the point facing the incoming wave. This effect may be relied upon to lift the response of the microphone at high frequencies, when it would otherwise begin to droop. Diffraction effects are important only at high frequencies. In the telephone bandwidth of 300-3000 Hz, they can be neglected.

The acoustic impedance mismatch at the interface between air and water or a solid is very great. The result is that sound is almost perfectly reflected or diffused from a liquid or solid surface. The same occurs for sound generated within water when it reaches the surface. The air and the water are almost perfectly separated acoustically. Sound waves exhibit all the diffraction and interference phenomena that light waves do, and usually more obviously.

We shall usually assume that the pressure at the microphone diaphragm is the pressure in the undisturbed sound wave. This is a good approximation for low frequencies and small microphones, where the microphone disturbs the sound wave only minimally. When the dimensions of the microphone approach the acoustic wavelength, the pressure is affected by diffraction and reflection. If the wave is reflected at the diaphragm, a pressure node is created and the pressure is twice that in the undisturbed wave.

Microphone Types

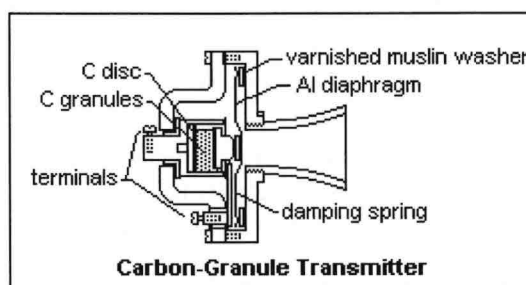
Bell's invention of the electromagnetic telephone in 1875 set off a vigorous search for a good transmitter, which was lacking to make the telephone a practical system. Passive transmitters could not provide sufficient power for general use in view of the lack of electronic amplification. Therefore, active transmitters of greater output that varied their resistance in time with the acoustic signal seemed the only answer. Many such devices rapidly came to light, using a wide variety of interesting techniques. Some used liquids, such as the Bell liquid transmitter itself, and others that employed jets of electrolytes. Others used uncertain mechanical contacts, or flames whose conductance varied with height, which in turn varied with acoustical pressure. Glow discharges in air were sensitive to acoustic waves as well. These devices are all interesting and curious, but none were satisfactory, and it is difficult now to experiment with them, though their principles may make good demonstrations.

Émile Berliner invented a variable-resistance solid transmitter using the contact between a metal diaphragm and a metal ball 1877. Edison followed with a similar solid-contact transmitter in 1878. In that year, Hughes suggested that the contact between small grains might be superior to that of larger bodies, and named transmitters using this principle *microphones* to distinguish them. Hunnings devised a microphone using carbon granules from coke, which was the starting point for Edison's search for a better transmitter. Edison perfected a transmitter using anthracite coal granules that became the standard for telephones after 1881. It was small, simple and easy to use, and above all had a large output. The carbon transmitter was a standby from this time until solid state electronics made it possible to put amplification in the telephone set, allowing other kinds of transmitters to be used. It is still the only transmitter that can be used in practice without electronic amplification. It has now been superseded by the electret condenser microphone, which is superior in characteristics and very cheap. Still, it is disappointing that carbon microphones are not still available for experimentation and non-critical use.

The Carbon Granule Microphone

Kohle-mic

A single-button carbon microphone as a telephone transmitter is shown in the figure. The mouthpiece acts as a horn to increase the acoustic pressure on the diaphragm. The displacement of the diaphragm is transmitted directly to the carbon button, which contains carbon granules between two carbon discs. The front and rear contacts are insulated and brought out to terminals. An external battery drives current through the button, which has a resistance of 30 to 100Ω. The resistance varies slightly when the diaphragm is displaced, causing a change in the current and a consequent change in voltage, which is the output of the microphone. The analysis in this section applies to any device in which the resistance is made to vary by the displacement of a diaphragm, not just to the carbon microphone.

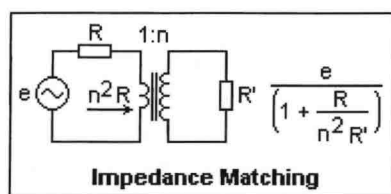


Suppose that the resistance of the button is $r = R - ax + bx^2$ to a sufficient degree of approximation. The constants a and b can be assumed positive, with b much less than a . The resulting current when a constant voltage E is applied will be $i = (E/R)(1 - ax + bx^2)^{-1}$. This can be expanded in powers of x to get $i = (E/R)[1 + ax + (a - b)x^2 + \dots]$. The quadratic term is usually negligible. If x varies sinusoidally, then the alternating current variation is $i' = (Ea/R)x$. This produces an alternating voltage of $e = Eax$, that can be considered as the Thévenin equivalent voltage source, in series with internal resistance R .

The overpressure $p \cos \omega t$ in the sound wave will produce a displacement $x = (pA/s)\cos \omega t$ in a stiffness-controlled diaphragm of stiffness s and area A . Of course, this expression may be generalized as required, but this approximation is good enough for the present purposes. This gives a generator voltage of amplitude $e = EapA/s$. The constants a and s are usually not well-known, but this at least shows the effects of the most important parameters. In particular, e is proportional to the DC voltage across or the current through the button.

The sensitivity of the microphone is expressed as $S = e/p = EaA/s$ or $IaRA/s$, normally in decibels: $\text{dB} = 20 \log S$. Telephone transmitters with a current of roughly 25-50 mA have S about -15 dB or -20 dB, or 0.18 - 0.10 V/μbar. This can be raised by a factor of 10 using a transformer, which gives 1.0-1.8 V at an

impedance level of 5 kΩ, a very creditable output. A transformer with a turns ratio of 1:10, or $n = 10$, raises the voltage by a factor n , to ne , and the impedance by a factor n^2 , to n^2R .



The power output into a load resistance R' coupled through a transformer of turns ratio n is $P = [e/(R + n^2R')]^2(n^2R')$. If we let $u = n^2R'/R$, then differentiate $P(u)$ with respect to u and set the result equal to zero, we find that $u = 1$, or $n^2R' = R$, the very familiar result for maximum power. On the other hand, if we wish the maximum voltage (as for the input to a high input impedance amplifier), then the

load resistance should be made as high as possible. If we use a transformer, then the DC current is not affected by such arrangements, and can still be adjusted to any level required. Impedance matching is illustrated in the diagram at the left.

Carbon transmitters of 1920 had sensitivity above -30 dB from about 600 Hz to 1900 Hz, strongly peaked at 1000 Hz. This range contains most of the voice power, but gave the telephone sound a somewhat unnatural quality. By 1934, the -30 dB bandwidth had been extended to 275-3100 Hz and was much less peaked, giving about -15 dB sensitivity from 75-2500 Hz. Further improvements were rather less dramatic, but the 1000 Hz peak was completely removed. This was a result of diaphragm and backing plate design, not changes in the carbon button.

A double-button carbon microphone has a push-pull action that cancels second harmonics. A good microphone with level response from 60 Hz to nearly 10 kHz, except for some wiggles of a few dB near the upper limit, was created. The sensitivity, however, was only about -47 dB, much less than that of a telephone receiver, in microphones optimized for good fidelity. By this time electronic amplification was available, so low output was not a drawback. Such microphones were used in recording and broadcasting in the early days, and gave very good service.

The outstanding disadvantage of the carbon microphone is noise, the so-called "carbon hiss," that could not be eliminated, though it could be reduced by careful preparation of the granules. This noise is inherent in the source of variable resistance, which was the surface properties of the carbon granules. Carbon by itself, even in bulk, exhibits $1/f$ (pink) noise, and this was exacerbated in the granules. The noise can be represented as a random voltage or current generator in series with the signal generator, in the usual way for noise analysis. The carbon granules could be damaged, and even fused together, by unusually high currents, such as those produced by inductive kicks. If the hermetic seal of the button was damaged, moisture could cause the granules to pack. The resistance of the microphone would decrease in that case, and it would become much less sensitive.

The Piezoelectric Microphone

Crystal - Mic

Tourmaline

It was long known that certain crystals, notably tourmaline, would attract light objects when strongly heated. This was the *pyroelectric* effect, the production of electrical polarization upon heating. While studying this effect, the brothers Pierre Curie (1859-1906) and P.-J. Curie (1855-1941) discovered the direct *piezoelectric* effect, or the production of electrical polarization when a crystal was strained, in 1880. In 1881 they announced the converse effect, the production of strain when an electric field was applied to a crystal. Much of the pyroelectricity previously observed was simply the piezoelectric effect due to strains caused by thermal expansion, but there is also a primary pyroelectric effect.

The application of an electrostatic field to any substance may cause mechanical strains by the phenomenon of *electrostriction*. These strains are proportional to the square of the applied field, and do not change if the field direction is reversed. Piezoelectricity is quite distinct; piezoelectric strains are proportional to the electric field, and reverse if the field is reversed. Piezoelectricity, where it exists, is usually much larger than electrostriction.

The description of the piezoelectric effect is made complicated by the many directional quantities and the crystal symmetries that enter. Strain is deformation per unit length, and has six components, three axial and three shear. Stress, force per unit area, also has six components. Stress and strain are related by a symmetrical matrix with, in general, 21 independent components. Electrical polarization, dipole moment

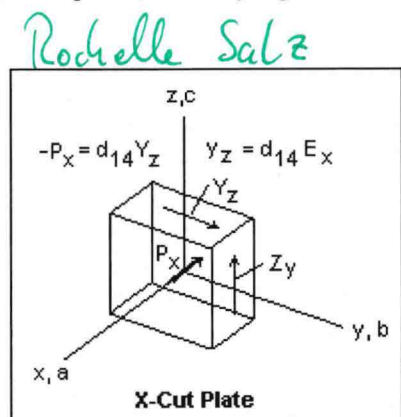
per unit volume, has three components, as does the electric field. Therefore, we have 18 quantities, all depending on each other and on the orientation of the crystal. At least we can assume that the dependence is linear, and described by a certain number of coefficients.

The symmetry important here is that of the point group of the crystal, those operations leaving one point fixed. There are 32 possible point groups, each the basis of a crystal *class*, divided into six or seven crystal *systems*. Crystals that do not have a centre of symmetry may exhibit piezoelectricity; those with a centre of symmetry cannot, by Neumann's theorem, which states that any property of a crystal must have at least the symmetry of the crystal. Such crystals are called *hemihedral*. Their axes are essentially one-sided, and opposite directions on them are not equivalent. This is required if the piezoelectric strain is to be proportional to the electric field, and reverse with it. Of the 32 crystal classes, 20 may be piezoelectric. There are, in general 18 coefficients connecting the electric field to the strain in the direct effect, or the polarization to the strain in the converse effect.

If X is a stress, in dyne/cm^2 , and x is a strain, dimensionless, then the relation between them is of the form $X = kx$, where k has the dimensions of stress, and is called an elastic modulus. The inverse relation is $x = sX$, where $s = 1/k$ is called a compliance, with dimensions cm^2/dyne . This is really a matrix relation, and the matrix s is the inverse of the matrix k , not a simple reciprocal, though often the actual relations are simple. Analogously, the direct piezoelectric effect can be expressed by $P = ex$, where x is strain, P the polarization in esu/cm^2 , and e is a piezoelectric constant with the dimensions of polarization. The converse effect can be expressed by $x = dE$, where x is the strain, E the electric field in statvolt/cm or erg/esu , and d is a piezoelectric constant, with dimensions of inverse field. The polarization and the electric field are also related by $P = \eta E$, where η is the electric susceptibility. Again, it must be emphasized that these are all tensor relations generally involving many coefficients, and a constant spontaneous polarization P_0 may also be involved. In that case, the P above is the change due to E .

When the Curies made their initial studies, which included discovering the piezoelectricity of quartz, which has been very important, they found that **Rochelle salt, or Sel de Seignette** (a pharmacist in La Rochelle who isolated and discovered the medical properties of the salt in 1672), had an extremely large piezoelectric effect. Rochelle salt was used extensively in microphones, and is of considerable interest besides, so the discussion here will focus on it. However, it is typical of all such materials. **Rochelle salt is sodium-potassium tartrate tetrahydrate, $\text{NaKC}_4\text{H}_4\text{O}_6 \cdot 4\text{H}_2\text{O}$** , which easily forms large orthorhombic crystals. Above 55°C , it begins to form separate Na and K tartrates dissolve in the water of crystallization, and disintegrates irreversibly. To preserve the crystal, it should not be heated above 45°C . **Its large piezoelectric effect occurs only between -18°C and $+24^\circ\text{C}$, called the lower and upper Curie Points** of the substance. Between these temperatures, it is an electrostatic analogue of a ferromagnetic material, called a *ferroelectric*, with a large spontaneous polarization. Like ferromagnetic materials, it is divided into *domains* of constant spontaneous polarization, but the domains are quite large, even centimetres in size. The domains are not obvious to the eye. Its crystals are enantiomorphic, like quartz, but only right-handed crystals occur in most cases.

An X-cut crystal plate of Rochelle salt is shown at the right. The x, y, z axes correspond to the crystallographic axes a, b, c . An X-cut plate has the normal to its broad surface parallel to the x -axis. There are only three piezoelectric coefficients for Rochelle salt, which relate the three shear strains to the three components of the electric field. The shear stress Y_z is a force per unit area in the z -direction on a surface whose normal is parallel to the z -axis. For equilibrium, it must be equal to Z_y . The opposite faces have the same forces acting, but in the reverse direction. This shear stress gives rise to an electric polarization in the direction shown, with d_{14} as the coefficient. The other nonzero coefficients are d_{25} and d_{36} , relating to zx and xy shears, respectively. The three coefficients are different in value, but d_{14} is the largest. For Rochelle salt, d_{14} is about $2.6 \times 10^{-4} \text{ cm/statvolt}$, or $1/d = 3850 \text{ statvolt/cm}$ (the electric field for unit strain). For a strain of 10^{-4} , an electric field of about 115.5 V/cm is required. The exact value of d depends on the temperature and the circumstances of the crystal, but this gives an idea of its magnitude.



A "45° X-cut rod" is an X-cut with the lateral sides making equal angles with the z and y axes. The shear strain y_z is $2\Delta L/L$, if L is the length of the rod, and ΔL is the change in length. The formula at the right in the diagram for the converse effect serves to find the change in length for any applied field. Of course, the strain and stress are related by the elastic constants of Rochelle salt, but we will not go into that here.

Ammonium dihydrogen phosphate or ADP, $\text{NH}_4\text{H}_2\text{PO}_4$, as well as the potassium salt potassium dihydrogen phosphate or KDP, are also strongly piezoelectric, resembling Rochelle salt quite closely. The Curie temperature of ADP is 147.9°C , it has no water of hydration, and is quite stable, so it makes more durable devices. The symmetry is about the same, but a little higher, so that $d_{14} = d_{25}$. Ceramics like barium titanate, BaTiO_3 , which are ferroelectrics (two lattices oppositely polarized spontaneously; the observed polarization is the difference), also are strongly piezoelectric. If a microphone is described as "crystal," it usually contains ADP; if it is called "ceramic," barium titanate is the active element. Any piezoelectric device is reversible; if a voltage is applied to a piezoelectric microphone, it will emit sound. It is also strictly a passive electroacoustic transducer, and the output power cannot exceed the acoustic input power.

Rochelle salt first became more than a curiosity around 1917, when it was applied by Langevin to ultrasonic acoustic transducers, or hydrophones, for the detection of submarines. Not only could strong signals be created in water by the converse piezoelectric effect, but the same crystals could be used to detect the reflected waves. This was, in fact, the origin of the important field of ultrasonics, which used acoustic waves of greater frequency than 20 kHz, which were inaudible but very useful.

The impedance mismatch between air and a diaphragm is much greater than the mismatch between water and a crystal hydrophone, so microphones are much more difficult to devise. The first microphones had a 45° X-cut bar, 1-2 cm long, 0.4-1 cm wide and 0.1 to 0.2 cm thick, cemented between a diaphragm and a backing plate. A much more sensitive arrangement was a "bimorph" of two cemented X-cut plates with one thin electrode between them. They could be arranged to bend or twist, and could be operated from a diaphragm through mechanical leverage. A typical inexpensive modern ceramic microphone responds from 30 Hz to 15 kHz, with a sensitivity of -60 dB ($1 \mu\text{V}/\mu\text{bar}$) and an advertised internal impedance of $8\text{k}\Omega$ at an unspecified frequency (Kobitone LM037). The capacitance measures 786 pF, which gives a capacitive reactance of $20.2\text{k}\Omega$ at 1 kHz. Piezoelectric microphones give low output at a moderate internal impedance, and must always be used with amplification.

Piezoelectric transducers were used as analog phonograph pickups, giving a much higher output than dynamic pickups. As driven elements, piezoelectric devices are used as telephone receivers, acoustic transducers and record cutters. They are used for small loudspeakers, although dynamic loudspeakers give much better results. In 1925, G. W. Pierce invented the acoustic interferometer, which uses an X-cut plate and a parallel reflector to measure the speed of sound with great accuracy. He devised an oscillator that was very sensitive to the reaction of the air on the crystal. W. G. Cady developed the quartz crystal resonator at about the same time, which has had widespread application as a frequency-control device.

The Condenser Microphone

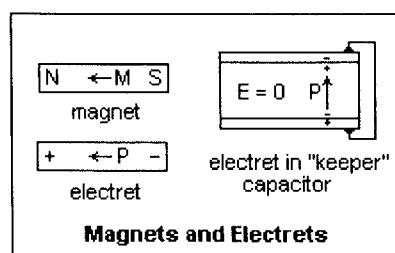
Condensator Mic

The condenser, or capacitor, microphone was perfected by C. H. Wente in 1917 as a much-needed low-noise substitute for the carbon microphone, and as a standard microphone for acoustical measurements. It was then used in broadcasting for a few years, until replaced by the dynamic microphone, which is much easier to use. The capacitor microphone is very simple in principle, and is still used for acoustical measuring instruments. The recent development of the electret capacitor microphone (ECM) has overcome all the inconveniences of the traditional capacitor microphone, and it is now used almost universally in general applications, as in telephones and consumer electronics. An excellent, easy to use microphone can be purchased for a dollar or two, and operated on 5 V at 0.5 mA. The ECM depends on two recent developments, the polymer electret film, and the field-effect transistor. We shall discuss it after looking at the traditional capacitor microphone.

A capacitor microphone consists of a metallized diaphragm that forms one plate of a capacitor, a backing plate forming the other. The diaphragm is tightly stretched to have a high resonant frequency, and is placed very close to the backing plate. Grooves are cut in the backing plate to control the mechanical impedance of the diaphragm. The capacitance of a plane-parallel capacitor of plate area A and separation h is $C =$

$4\pi\kappa\epsilon A/h$ F, where A is in m^2 and h is in m. The dielectric constant is κ , and $\epsilon = 8.854 \times 10^{-12}$ F/m, the MKS constant that masquerades as a physical reality. Since the dielectric is air, we can take $\kappa = 1$. The charge on a capacitor charged to a voltage V is $Q = CV$. If h varies, then $dC = -(4\pi\epsilon A/h^2)dh = -(C/h)dh$. This means that the current will be $i = -(CV/h)(dh/dt)$.

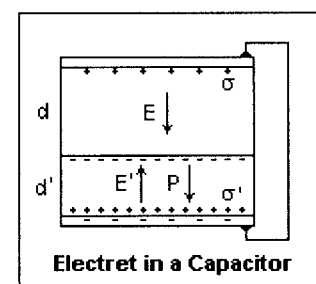
The microphone can be represented as a Norton equivalent circuit with a current generator i in parallel with a capacitance C . This can be transformed to a Thévenin equivalent circuit of a generator $e = i/j\omega C = (V/h)x$, where x is the displacement of the diaphragm, in series with a capacitance C . If the diaphragm is stiffness-controlled, then $x = pA/s$, so $e = pVA/hs$ and the sensitivity $S = e/p = VA/hs$. The sensitivity is proportional to the bias voltage V and the diaphragm area A , and inversely proportional to the separation h and the stiffness s . The stiffness is raised if h is reduced, so they cannot be varied independently. Taking $A = 10 \text{ cm}^2$, $h = 0.01 \text{ cm}$, and $s = 1 \times 10^8 \text{ dyne/cm}$, we find $S = 10^{-5} \text{ V V}/\mu\text{bar}$. If $V = 200 \text{ V}$, then $S = 2 \text{ mV}/\mu\text{bar}$, or -54 dB . This happens to be a relatively typical value for a capacitor microphone. The capacitance of the microphone will be about 88.5 pF , which will give a capacitive reactance of $1.8 \text{ M}\Omega$ at 1 kHz , and $18 \text{ M}\Omega$ at 100 Hz . The microphone cannot be located far from the amplifier. Furthermore, the bias supply must be extremely well regulated and ripple-free, since every variation will be combined with the acoustic signal.



An *electret* is a body with a permanent polarization, analogous to a magnet which has a permanent magnetization. Polarization P is dipole moment per unit volume, and has the dimensions of surface charge density. If P is uniform in the electret shown, a surface charge $+P$ appears on the left-hand face of the electret, and a charge $-P$ on the right. The electric dipole moment of the bar electret is $p = PAL$, where A is the cross-sectional area of the electret, and L its length, just as the magnetic dipole moment of the bar magnet is $m = MAL$.

An electric field exerts a torque on a dipole tending to align the dipole with the field, just as a magnetic field acts on a magnetic dipole. The bar magnet and bar electret shown establish fields in space, and these fields have energy. Some of this field (the "demagnetizing field") acts in the reverse direction to the polarization or magnetization, tending to reduce it. A soft iron bar may be placed over the poles of a magnet as a "keeper" through which most of the field will pass. Magnetic pole strength can be considered to be induced at the ends of the keeper, which will neutralize some of the pole strength of the magnet, reducing the demagnetizing field. Exactly the same thing occurs with an electret when it is placed between the plates of a shorted capacitor. The surface charge due to the polarization is neutralized by the surface charge of opposite sign induced on the capacitor plates. This happens naturally when an electret is exposed to the air, as it collects charged particles floating as ions and dust. A magnet is not neutralized in the same way, because there are no free magnetic charges.

Assume that we have a sheet electret of thickness d' , rigid polarization P and permittivity ϵ' . Let this sheet be placed in a parallel-plate capacitor so that there is a space d between the upper plate and the upper surface of the electret, and the permittivity of this air space is ϵ . This capacitor is carefully sealed away from floating charges so that the electret does not become neutralized. When the plates are shorted to each other (a "short" in this case can be a resistance of many megohms) the voltage difference between them becomes zero. The fields in the air space and in the electret must be opposite in direction, and related by $Ed = E'd'$ so the total voltage is zero. A charge density σ will appear on the upper plate, and an opposite charge will appear on the lower plate. The net charge at the bottom of the electret will be $\sigma' - \sigma$, where $\sigma' = P$. An opposite charge will appear on the top of the electret.



The field E is given by $\epsilon E = \sigma$ and the field E' by $\epsilon' E' = \sigma' - \sigma$. Since $E'd' = Ed$, $\sigma = \sigma'[d'/(d' + \epsilon'd/\epsilon)]$, and so $E = (P/\epsilon)[d'/(d' + \epsilon'd/\epsilon)]$. As d becomes small, this approaches $E = P/\epsilon$. If the upper plate is the diaphragm of a microphone, the electric field E plays the same role as the field produced by the bias voltage in an ordinary capacitor microphone. Elimination of the bias voltage and all its inconveniences makes the electret capacitor microphone a very desirable device.

Since the capacitance is small, even with the dielectric properties of the electret, the ECM still has a very high internal impedance that is almost entirely capacitive. This drawback is eliminated by putting an FET

right at the diaphragm. The gate presents a very high impedance to the diaphragm capacitor plate, and the FET is manufactured to have a small drain current for $V_{GS} = 0$. The output voltage is the voltage across an external drain resistor in the range $2k\Omega$ to $50k\Omega$, which becomes the output impedance of the microphone. The output voltage also increases proportionately to the drain resistance, of course.

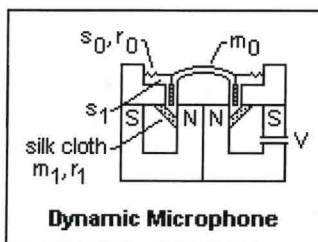
An example of an ECM is the Kobotone LM045. This microphone is remarkably small, only 9 mm in diameter and 6 mm tall. Diffraction effects will be negligible, so this omnidirectional microphone will probe an acoustic field without perturbing it. Its bandwidth is 20 Hz to 12 kHz, and its sensitivity is advertised at -64 dB (without specifying the load resistor, which is unfortunate), or $0.63 \text{ mV}/\mu\text{bar}$. The power supply range is 2 to 10 V, and the current drawn is less than a milliampere. I found that a load resistance of $22 \text{ k}\Omega$ and supply voltage +5 gave excellent results. Perhaps most remarkable is that it costs only \$2.29! The ECM is a worthy successor to the carbon microphone for general uses.

The Dynamic Microphone

Dynamic Mic

The principle of the dynamic microphone was known in 1877 when Bell developed the telephone, but it was impossible to use because of the lack of electronic amplification. In all dynamic transducers, a coil of fine wire is free to move in a strong annular magnetic field. If the coil is moved by a diaphragm, a voltage is induced in the coil. If a current flows through the coil, forces are exerted that cause the coil to move. The equations governing these effects are $F = BLi$ and $e = BLv$. B is the magnetic field in tesla (webers per square metre), L the length in metre, v the speed in metre/s, i the current in A, and e the voltage in V. From these equations, we find that $F/i = e/v$, or $Fv = ei$. Fv is the rate of doing mechanical work, and ei the rate of doing electrical work. The signs are such that when mechanical work is done, an equal electrical work appears, and vice versa. This shows very clearly that we are working with a reversible effect and that the conservation of energy is observed. The word "dynamic" simply refers to the role of motion in the device; it is not a very well-chosen term, but always refers to a moving-coil device.

In somewhat friendlier mixed units, we have $F = BLi/10$, where F is in dyne, B is in gauss, and i is in A. Also, $e = BLv \times 10^{-8}$ in V, where B is in gauss, L is in cm, and v is in cm/s. The generated voltage in a microphone is $e = 10^{-8}BLpA/Z'$, where p is the overpressure in μbar , A the area of the diaphragm in cm^2 , and Z' is the mechanical impedance of the diaphragm. It is clear from this that if the diaphragm is a simple oscillator, the output cannot be independent of frequency, since the magnitude of Z' is least at resonance, and increases rapidly for higher and lower frequencies. L is the total length of wire in the coil, $L = \pi dN$, where d is the diameter of the coil and N the number of turns.



Dynamic microphones became possible when it was realized that by making the diaphragm oscillator include air volumes within the microphone, the response could be flattened very nicely, at the expense of some sensitivity. The structure of a typical dynamic microphone is shown in the figure. The domed diaphragm acts like a rigid piston, and carries the coil of wire, which moves in the annular gap of high magnetic field. The pole pieces of the microphone are soft iron, with permanent magnets providing the field. The mounting rim of the diaphragm contributes stiffness and a little resistance

(s_0, r_0), while the diaphragm itself contributes mass (m_0). It is acted upon by the acoustic overpressure, so that the microphone is a pressure microphone, with omnidirectional characteristics. The air in the small volume beneath the diaphragm acts as another stiffness element (s_1), while the kinetic energy of the air moving in and out from under the diaphragm through the silk cloth contributes mass (m_1), as well as a larger resistance (r_1). The result is two coupled oscillators, whose parameters can be varied to get the best results. The electrical analogy was a help in designing dynamic microphones, since the results of different arrangements could be studied easily without building actual microphones. The port V is to help low-frequency response.

A typical dynamic microphone has a very low internal impedance, seldom as large as 10Ω , and this impedance is approximately resistive over the whole frequency bandwidth. On the other hand, the sensitivity is quite low, no more than -90 dB or -100 dB, so amplification is essential. At least 40 dB can be gained with transformers, bringing the output up to -50 dB when applied to the amplifier, which is not too bad. Dynamic microphones are low-noise, require no bias voltages or other nuisances, and are relatively

rugged. They replaced capacitor microphones almost completely in broadcasting and recording. Most high-quality microphones are still dynamic microphones, and are relatively expensive.

A dynamic microphone in reverse becomes a loudspeaker. Of course, the designs are quite different, because they must be optimized for different things. A small loudspeaker makes a very passable microphone, and can be used for this purpose, as in an intercom system. Such small loudspeakers radiate poorly at low frequencies, so this is compensated by making them resonate at a few hundred Hz. This is easily recognized in the oscilloscope traces when the output of a small loudspeaker is used as a microphone, since it tends to ring at this frequency. A transformer of turns ratio 1:10 can be used to increase the voltage output, and to make the internal impedance about 800Ω for the usual 8Ω speaker. The same amplifier that drives the speaker from a line of this impedance can be turned around to drive the line in turn when a button is pressed.

The Ribbon Microphone

Bänder mic (M160)

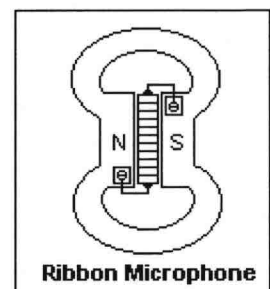
All the microphones so far described are operated by the acoustic overpressure acting on one side of a diaphragm, and so may be called *pressure microphones*. Because pressure is a scalar quantity, these microphones are omnidirectional, except for diffraction effects that depend on frequency. To realize a directional microphone, it is necessary to operate the microphone by some *vector* quantity. One vector quantity in an acoustic wave is the *pressure gradient*, which is parallel to the direction of propagation and in phase with the displacement. A microphone operated by the pressure gradient is called a *pressure-gradient* microphone. Since velocity is also a vector quantity, such microphones are also called *velocity* microphones, but they are not directly operated by particle velocity.

If $\partial p / \partial x$ is the pressure gradient, the force in a direction making an angle θ with the propagation direction, acting on a surface of cross-sectional area A and length L is $f = -(\partial p / \partial x)AL \cos \theta$. For a harmonic travelling wave, $f = jkpAL \cos \theta$. This is the desired angle-dependent force. It is in quadrature to the pressure, and proportional to $k = \omega / c$. If ω is much larger than the resonant frequency of the surface, then the mechanical impedance is $j\omega m$, where m is the mass of the surface. The velocity of the surface is the ratio of the force to the mechanical impedance, or $v = (pAL / mc) \cos \theta$, where c is the speed of sound. This means that v is proportional to p independently of frequency.

This equation holds when $kL \ll 1$, that is, when the size of the microphone is small compared to a wavelength. At high frequencies this is not necessarily true, and the force is smaller, dropping to zero when $f = c/L$. Instead of $kl \cos \theta$, we have $2 \sin[(kl \cos \theta)/2]$ in the expression for the force on the surface. This falloff is compensated by an increase in pressure difference due to diffraction.

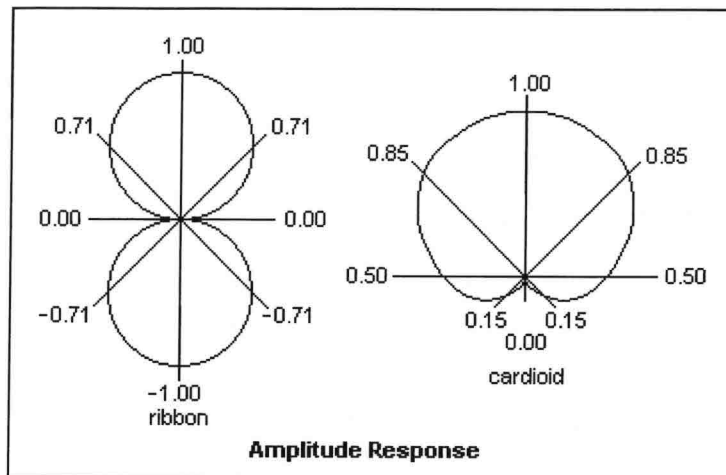
I have spoken of a "surface" to avoid using the word "diaphragm," which gives the wrong idea when used of a pressure-gradient microphone. The most important pressure-gradient microphone is the *ribbon* microphone. The surface in this case is a corrugated aluminium ribbon supported in a strong magnetic field. The emf generated when the ribbon moves is proportional to v , and so to the overpressure p . That this ribbon acts like a thick surface is harder to realize. The ribbon is exposed to pressure equally on front and back, and the distance L is determined by the size of the baffle in which the ribbon is suspended. L turns out to be roughly equal to the radius of a circular baffle. (not $2L$, as might be expected).

The ribbon acts like a coil of only one turn, so the generated emf is very small. On the other hand, not only is the internal impedance very small (less than an ohm), but the velocity can be made higher by reducing the mass m to a minimum value. The sensitivity of a ribbon microphone may be -90 dB or -105 dB, but more than 40 dB can be gained at once with a transformer, so its output of -50 dB is comparable to that of a dynamic microphone. There is usually a transformer at the microphone to match it to a transmission line, and another transformer at the amplifier input.



Most significantly, we now have a sensitivity that is proportional to $\cos \theta$. The ribbon microphone is equally sensitive to sound coming from front and back (so two people can face one another across it and both be equally heard), and is completely insensitive to sound coming from 90° or 270° . This pattern is not dependent on frequency, as are diffraction effects. The ribbon microphone discriminates by a factor of 3

against isotropic noise. It can also be turned so that noise sources can be put in a zone of low sensitivity. These features made the ribbon microphone the standard for broadcasting, and the lozenge-shaped shiny microphone a familiar sight.



The cosine sensitivity of the ribbon microphone can be combined with the isotropic sensitivity of a pressure microphone to make a microphone with a response proportional to $(1 + \cos \theta)$. This curve is a *cardioid*. Cardioid microphones favor sound coming from 0° , and discriminate against sounds coming from 180° . A cardioid microphone at the front of a stage will pick up sounds originating onstage, and reject those coming from the direction of the audience. Polar plots of the amplitude sensitivity of the ribbon and cardioid microphones are shown at the

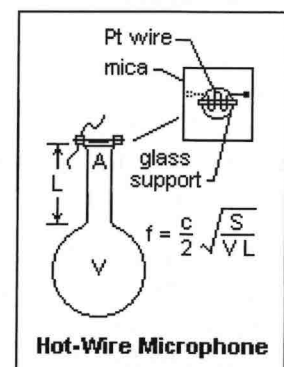
left.

The amplitude response of a pressure gradient microphone changes sign for a reversal of the direction of the wave, while that of a pressure microphone does not. This has a strange effect in a standing wave, where waves are moving in opposite directions. At a point where the pressure microphone gives a maximum signal, the pressure-gradient microphone gives a minimum signal, and vice-versa.

The Hot-Wire Microphone

The hot-wire microphone is not like the other microphones we have studied. It does not reproduce sound pressure variations electrically, but is more of a detector of sound and an indicator of its energy. Since the name does crop up from time to time, we'll describe it here for completeness. It is specifically used for low frequencies and for infrasonic signals. It was developed during the 1914-1918 war as a sound ranging device, for acoustic location of artillery to aid counterbattery fire. After the war, Tucker and Paris perfected the hot-wire microphone for infrasonic detection, publishing their results in 1921.

An example of a hot-wire microphone is shown at the right. It consists of a very fine platinum wire placed over the neck of a Helmholtz resonator and heated by a current passed through it. The wire is supported by a thin glass rod and a disc of mica. The disc is clamped between silver rings that make the contacts. When a sound wave of the resonant frequency arrives, air rushes in and out of the neck of the resonator at that frequency. This air flow cools the wire by forced convection, so its resistance decreases. The resistance decrease is easily detected by a Wheatstone bridge. The hot wire of a typical device is $6 \mu\text{m}$ in diameter, with a resistance of 350Ω and requiring about 30 mA to heat.



A Helmholtz resonator consists of a volume V and a neck of length L and cross-sectional area A . Its resonant frequency is given by the formula in the diagram, where c is the speed of sound. A 125-ml Florence flask makes a good Helmholtz resonator. I measured $L = 5.5 \text{ cm}$ and $A = 1.54 \text{ cm}^2$, which gave $f = 256 \text{ Hz}$ (the physicist's middle C). The actual resonance was an A, or 220 Hz, on the musician's scale, not far off. Without the resonator, the sensitivity of the hot-wire microphone is very low, so practical devices are all resonant. The microphone can be applied to frequencies as high as 512 Hz.

In addition to the DC change in resistance, it is also possible to detect AC variations in the hot-wire resistance. These variations are at twice the sound frequency, since the air blows alternately in and out, and the cooling does not depend on the direction of the air velocity. The hot-wire microphone is, accordingly,

not applicable to speech or music. As its use in acoustic ranging indicates, it has a rather quick response. It is useful in a frequency range where other microphones are unresponsive.

Microphone Specifications

The most fundamental microphone specifications should allow you to estimate how much electrical output you will get for a certain acoustical input. All microphones, except for carbon microphones, require electronic amplification, and you should be able to select a suitable amplifier. The microphone can be represented by a Thévenin equivalent circuit with a generator in series with an internal impedance. At a minimum, the microphone specifications should supply these two parameters.

The generator voltage is typically specified in dB re 1 V per μbar , or $e = 20 \log p$, where p is the acoustic overpressure in dyne/cm^2 or μbar . Sometimes the pascal is used as the reference pressure, $1 \text{ Pa} = 10 \mu\text{bar}$. This adds 10 dB, so manufacturers like to use it to imply that their microphones are more sensitive. The sensitivity is specified at some reference frequency, usually 1000 Hz. A curve of dB vs. frequency, giving the frequency response of the microphone, may be available. A bandwidth of 50 Hz to 5000 Hz between -3 dB points is not bad, but most microphones exceed this, and 30 Hz to 10,000 Hz is often found.

The *calibration* curve of microphone sensitivity in dB vs. frequency must be measured experimentally. This is usually done in one of two ways. If a constant overpressure is applied at each frequency, a *constant-pressure* calibration results. If the microphone is placed a reasonable distance from the microphone in an anechoic chamber, so that it is activated by what is essentially a plane wave, the result is a *free-field* curve. The free-field curve will be more representative of the response of the microphone in actual use, since it includes the effects of diffraction and other influences that change the relation between the acoustic pressure in a wave and the pressure on the diaphragm. On the other hand, the pressure calibration reflects the fundamental properties of the microphone, and is useful in designing measuring instruments.

Equally important is the specification of the internal impedance of the microphone, which varies over an extremely wide range for different microphone types, from less than 1Ω for a ribbon microphone to many megohms for a capacitor microphone. This is more difficult to find, and sometimes you must assume a typical value for the microphone type involved. A microphone with a low internal impedance, say 50Ω or less, is normally used with a step-up transformer that provides 40 dB of gain effortlessly (with a 1:100 transformer) and matches well to the high input impedance of many amplifiers. A microphone with a high capacitive internal impedance cannot be used with a cable of any length, so the amplifier must be very near. In the modern electret condenser microphone, the amplifier is an FET in the microphone cartridge itself.

Sometimes microphones are specified in terms of output power for a given acoustical input, often as dB re 1 mW per pascal. Transformers do not change the power, of course, so they do not affect this specification. With this figure, it is easy to estimate how much amplifier gain will be necessary to bring the electrical signal to the desired level. 0 dB is, as usual, re 1 mW. This is not as useful a specification as that of sensitivity and internal impedance, from which it can easily be derived.

The directionality of a microphone is often of interest in applications. A pressure microphone is fundamentally omnidirectional at lower frequencies (wavelength larger than the physical size of the microphone). A ribbon microphone has maximum sensitivity from front and back, and is insensitive to sound coming from the sides. Its sensitivity has front and back lobes of approximately equal size of $r = a \cos \theta$ shape. A *cardioid* microphone has the very desirable characteristic of a large forward-back asymmetry in its sensitivity. Placed at the front of a stage, it can pick up sound coming from the stage, and reject that coming from the audience. The directional sensitivity can be expressed in a polar plot. The directionality can be expressed in dB by subtracting the average sensitivity over all directions from the maximum sensitivity. For a ribbon microphone, this is 4.8 dB.

References

L. E. Kinsler and A. R. Frey, *Fundamentals of Acoustics*, 2nd ed. (New York: John Wiley & Sons, 1962). Particularly Chapters 1 and 11.

A. L. Albert, *Electrical Communication*, 2nd ed. (New York: John Wiley & Sons, 1940). Chapter V. Note that "bar" here is really the μbar .

W. G. Cady, *Piezoelectricity* (New York: McGraw-Hill, 1946). This venerable classic is still the best explanation of piezoelectricity, correct and complete.

A. Wood, *Acoustics* (New York: Dover, 1966; originally London: Blackie and Son, 1940). Hot-wire microphone, pp. 303-305.

Inexpensive ECM's, small loudspeakers, crystal and ceramic microphones, and dynamic microphones are available from [Mouser](#).

A new type of microphone is described in [Schwartz Engineering](#).

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