INSTRUCTION IN THE USE OF

Microphones for Sound Reinforcement Systems

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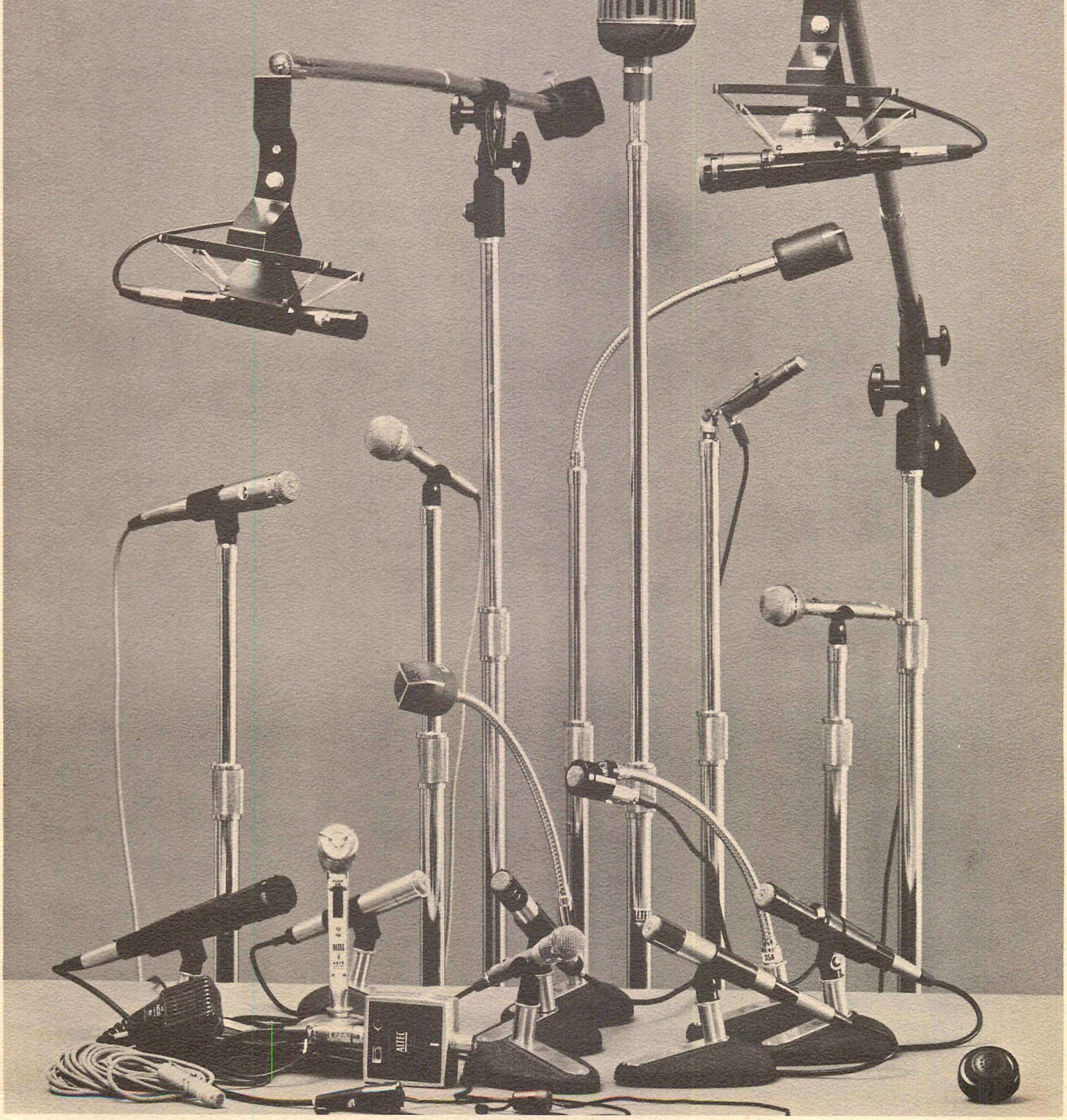
ARTHUR C. DAVIS & DON DAVIS



ALTEC LANSING®
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INSTRUCTION IN THE USE OF

Microphones for Sound Reinforcement Systems

ARTHUR C. DAVIS & DON DAVIS

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of
ALTEC LANSING®
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Part 1

Acoustical requirements of large-audience listening areas and the role played by microphones in meeting system criteria are examined

The design, installation and successful adjustment of a high-quality sound reinforcement system is a complex engineering problem. Modern practices have tended toward increased use of precision acoustical measuring equipment at all three stages prior to customer use of the system. A well-engineered sound reinforcement system today will meet all the following criteria:

1. Provide an acoustic gain between 10 and 30 dB. (See Figs. 1 and 2 for an explanation of acoustic gain and its measurement.)
2. Provide even distribution of the reinforced sound throughout the audience area — typically ± 2 dB from front-to-back or side-to-side for the one-octave band centered on 4000 Hz. (See Fig. 3.) Total variation from the worst to the best seat equals ± 4 dB.
3. Provide uniform frequency response throughout the audience area—typically ± 3 dB as measured with ⅓-octave bands of "pink noise" at positions across the main audience area.
4. Provide correct "time" relationship between the loudspeaker and the listener's ear as compared to the time interval from the talker to the listener's ear. (See Fig. 4.)
5. Provide adequate dynamic range at an acoustic distortion sufficiently low to insure minimum listening fatigue. The reinforcement system should be capable of providing 90- to 100-dB sound pressure level (SPL) to any seat in the audience area at an acoustic distortion below 5% total harmonic distortion (THD).

All of this must be accomplished with components that are rugged, reliable, easy to service, and because of their price, they must have a long life expectancy to allow for a low amortization figure.

This article is intended as a detailed discussion of the microphone's role in such a system and the parameters it must contribute if the system's criteria are to be met.

It can be realized quickly that carbon, crystal, ceramic, and hot-wire microphones are not the type required. The basic types we can consider are:

1. Moving coil
2. Ribbon
3. Condenser

Further, we can divide the use of these basic mechanisms for transducing acoustic energy into electric energy as follows:

1. Omnidirectional response
2. Bidirectional response
3. Unidirectional response

All three types of microphones in each of the directional response patterns will have to have the following questions answered correctly before being connected into the electronics of your reinforcement system:

1. For a SPL of 94 dB at the microphone's diaphragm, what is the effective output in dBm? (10 dynes/cm^2 = 94 dB SPL.)
2. What is the rated impedance of the microphone's output?
3. Does it require an external power source?

Mechanically, the following questions must be answered.

1. Is a shock mounting provided?
2. Is a wind "pop" screen provided?
3. Is any special mounting required?

(This must be ascertained at the design stage since such solutions may not always come off the manufacturer's shelf.)

Finally, you must either receive unimpeachable documentation from the manufacturer (actual anechoic chamber measurements) or else conduct careful tests yourself regarding:

1. Uniformity of frequency response on axis.
2. Uniformity of frequency response on the so-called "dead" side.
3. The separation between the front and the back of directional microphones expressed in dB. (Both in the horizontal and the vertical plane.)
4. The "self noise" level and the "overload" level.

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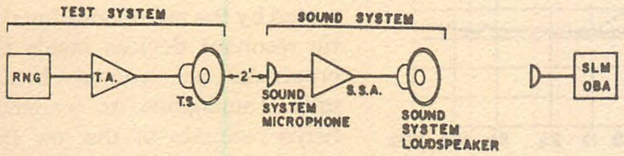
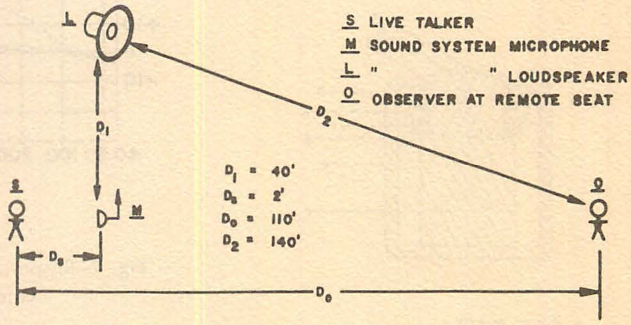


1. Adjust random-noise generator (RNG) and test amplifier (TA) to produce 90 dB SPL at a distance of 2′ in front of the test. Locate loudspeaker (TS) at sound-system microphone position.
2. Carry the sound-level meter (SLM) to the most remote listening position used by a regular audience and, with the octave band analyzer (OBA) adjusted to the 2000-Hz octave band, take a SPL reading in dB. (Sound system is shut OFF during these measurements.)
3. Place sound-system microphone 2′ in front of test loudspeaker (TS). Adjust sound system's gain to a point just below acoustic feedback. Switch on test loudspeaker. Set at same level as step (1): 90 dB SPL at 2′.
4. Again take SPL reading in dB with SLM at most remote listening position and adjust to 2000-Hz octave band. SLM reading in step (2) subtracted from SLM reading in step (4) equals the acoustic gain in dB SPL.

Fig. 1—A step-by-step description of acoustic gain measurement is shown here. Steps 1 and 2 are effected before the sound system microphone is placed two feet in front of the test loudspeaker.



$$SG = (20 + \triangle)\,\mathrm{Log}_{10}\left[\frac{D_1}{D_S} \cdot \frac{D_0}{D_2}\right]$$

EXAMPLE: Large church, 5.5 sec. R.T. at 512 Hz

Assume $\triangle = 0$ for initial calculation

$D_1 = 40'$
$D_S = 2'$
$D_0 = 110'$
$D_2 = 140'$

$$(20 + \triangle)\,\mathrm{Log}_{10}\left[\frac{40}{2} \cdot \frac{110}{140}\right] =$$

$$(20 + \triangle)\,\mathrm{Log}_{10}\,15.7 = 24\ \mathrm{dB}\triangle$$

$\triangle = 0$ is a realistic goal where detailed narrow band equalization is employed. If broad-band equalization alone is used, ½ the gain achieved when $\triangle = 0$ is a conservative estimate.

Fig. 2—Calculation of gain before feedback.

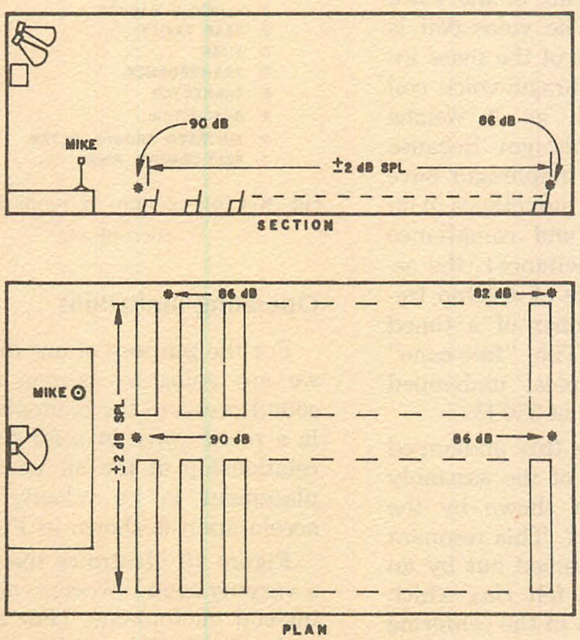
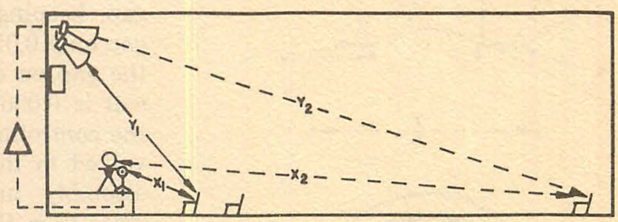
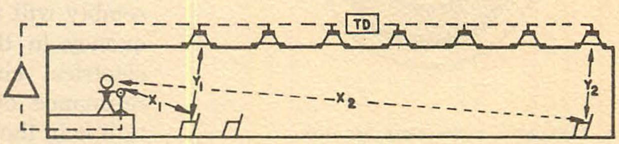


Fig. 3—A criterion of a well-engineered sound reinforcement system is to provide even distribution of sound throughout the listening area.

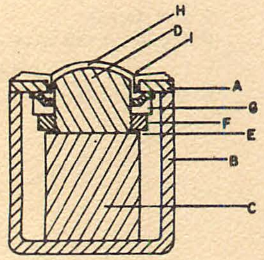


1. If "live talker" is closer to listener than distance from loudspeaker to listener then:
   (a) $y_1 - x_1$ must be $\leq 40'$ (45.6 milliseconds)



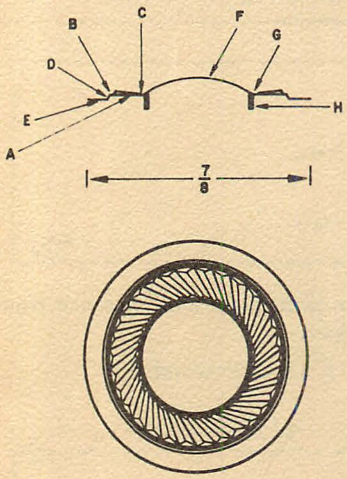
2. If loudspeaker is closer to listener than distance from "live talker" to listener:
   (a) $x_1 - y_1$ must be $\leq 40'$ (45.6 milliseconds) UNLESS SPL from loudspeaker at listener's ear exceeds SPL from "live talker" by at least 15 dB SPL.
   (b) If loudspeaker is not 15 dB SPL higher at the listener's ear than the "live talker" and $x_2 - y_2 > 40'$ (45.6 milliseconds) then the use of a time-delay mechanism should be given serious consideration.

Fig. 4—Time-distance relationships of a sound system are illustrated.



A POLE PLATE
B MAGNETIC RETURN
C ALNICO V MAGNET
D POLE PIECE
E THREADED PORTION
F THREADED ADJUSTING RING
G ACOUSTIC RESISTOR (FELT)
H CAPACITANCE UNDER DIAPHRAGM DOME
I VOICE COIL

Fig. 5—Cross-section of a dynamic pressure unit.



A TANGENTIAL COMPLIANCE SECTION
B HINGE POINT
C " "
D SPACER
E CEMENTING FLAT
F DOME
G COIL SEAT
H VOICE COIL

Fig. 6—Omni-directional diaphragm and voice-coil assembly.

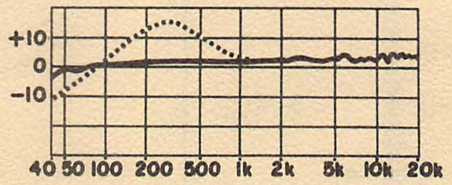


Fig. 7—Response curve of diaphragm and voice-coil assembly.

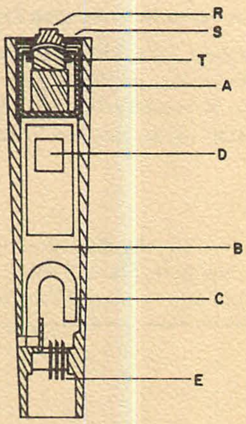
The moving-coil microphone

A basic discussion of a pressure-operated omnidirectional moving-coil microphone is in order before exploring the subject further. In a dynamic pressure unit (See Fig. 5) the magnet and its associated parts (magnetic return, pole piece, and pole plate) produce a concentrated magnetic flux of approximately 10,000 gauss in a small gap. Placed into this small gap is the voice coil mounted on the microphone diaphragm. (See Fig. 6.)

The diaphragm positions and supports the voice coil in the magnetic gap, keeping it in the center of a gap only 0.020-in. wide. (Typically, the gap on either side of the voice coil is 0.006-in.) The voice coil is the controlling part of the mass involved in the diaphragm-voice coil assembly inasmuch as it weighs more than the diaphragm. Because the voice coil and diaphragm have mass (analogous to inductance in an electrical circuit) and compliance (analogous to capacitance), the assembly will resonate at a given frequency in the manner of a tuned electrical circuit. The "free-cone" resonance of a typical undamped unit is in the region of 350 Hz.

If it were left in this undamped state, the response of the assembly would be like that shown by the dotted line in Fig. 7. This resonant characteristic is damped out by an acoustic resistor, a felt ring which covers the openings in the centering ring behind the diaphragm. This is analogous to electrical resistance in a tuned circuit, and damps the resonant point down to a flat response. Even with the unit damped, there is a drooping in the lower frequency range from about 200 Hz down (dotted line, Fig. 7). This is corrected by the use of additional acoustic resonant devices inside the microphone case. A cavity behind the unit (analogous to capacitance) helps resonate at the low frequencies with the mass (inductance) of the diaphragm-and-voice-coil assembly.

Still another tuned resonant circuit is added in the form of a tube (Fig. 8) which couples the inside cavity of the microphone housing to the outside. This tube has an acoustic inductance which is tuned to a low frequency, in this case 50 Hz, so that a flat response extending down to 35 Hz may be obtained.



A ALNICO V MAGNET
B CASE CAVITY
C TUBE
D TRANSFORMER
E CONNECTOR
R RESONATOR
S SINTERED BRONZE FILTER
T FELT DAMPING RING

Fig. 8—Cross-section of omni-directional microphone.

Operating limitations

For the purpose of our discussion we are going to assume that the sound pressure the microphone sees is a plane wave in a far field. The relationship of the air particle displacement to its velocity and its acceleration is shown in Fig. 9.

Figure 10 illustrates the effect of a varying sound pressure on a moving-coil microphone. (For this brief and, admittedly, simplified explanation, assume that a massless diaphragm voice-coil assembly is used.) The acoustical waveform (I), is one cycle of an acoustic waveform, where (A) indicates atmospheric pressure, (AT); and (B) represents atmospheric pressure plus a slight over-

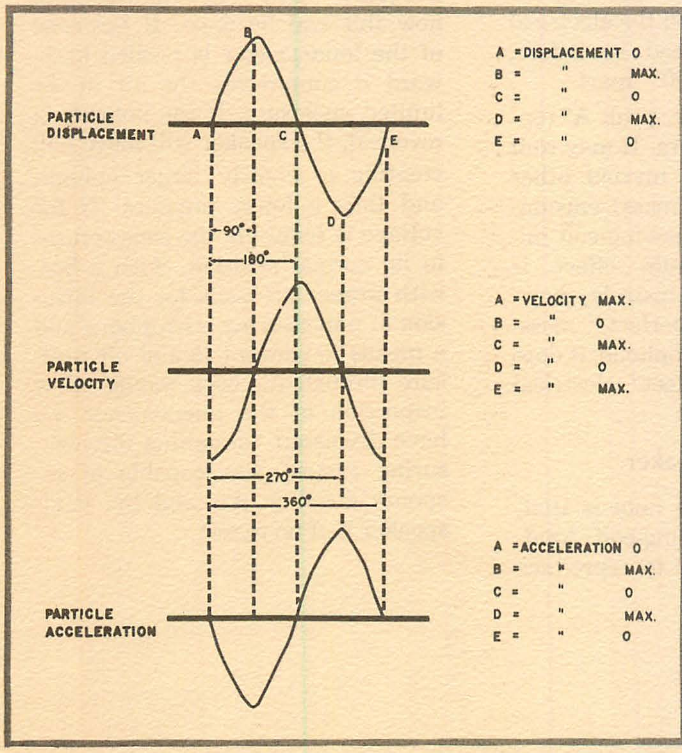


Fig. 9—Air particle motion in a sound field, showing relationship to velocity and acceleration.

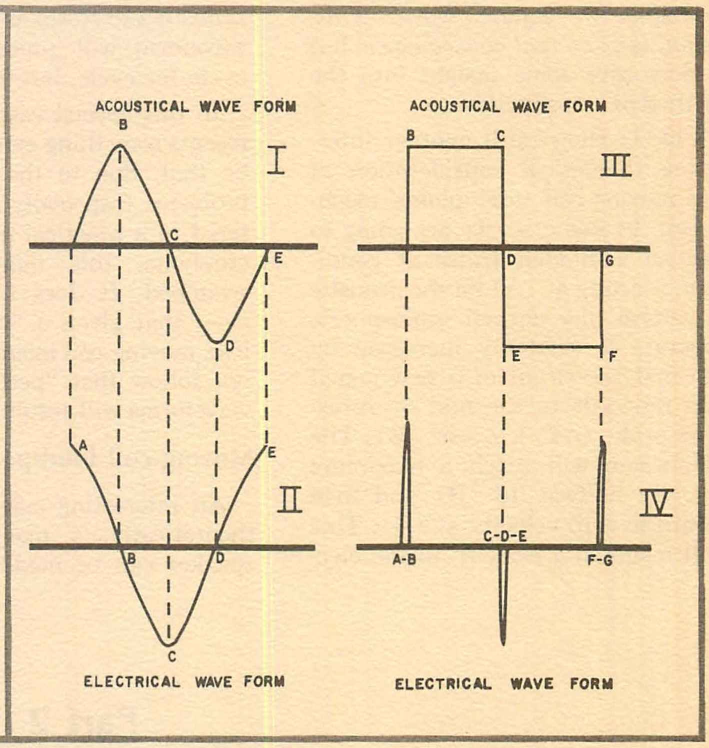


Fig. 10—Effect of a varying sound pressure on a moving-coil microphone. See text.

pressure increment, (Δ) or (AT + Δ).

Looking at (II) on Fig. 10, we see that the electrical waveform output from the moving-coil microphone does not follow the phase of the acoustic waveform. This is due to the fact that at maximum pressure, (AT + Δ) or (B), the diaphragm is at rest (no velocity). Further, the diaphragm and its attached coil reach maximum velocity, hence maximum electrical amplitude, at point (C) on the acoustical waveform. This is of no consequence unless you are using another microphone, along with the moving-coil microphone, in a stereo system where the other microphone does not see the same 90° displacement. Due to this phase displacement, condenser microphones should not be mixed with moving-coil or ribbon microphones. (Sound pressure can be proportional to velocity in many practical cases. See *Handbook of Noise Measurements* by Arnold P. G. Peterson and Erwin E. Gross, Jr., General Radio Co., Page 33.)

Looking at (III) in Fig. 10, let us assume for the moment that we can create a steady overpressure and hold it (generate an acoustic square wave), and that we still have a massless-diaphragm voice-coil microphone as well as a massless loudspeaker. The result would be as shown in (IV). As the acoustic pressure rises from (A) to (B), it represents a velocity; and voltage output from the microphone appears. Then, as the diaphragm reaches its maximum displacement and stays there during the time interval represented by the distance between (B) and (C), no voice-coil velocity exists; hence the electrical output voltage ceases. The same situation would repeat itself from (C) to (E), and from (E) to (F) on the acoustic waveform. It can be readily seen

Fig. 11—Effect of a transient condition on a moving-coil microphone.

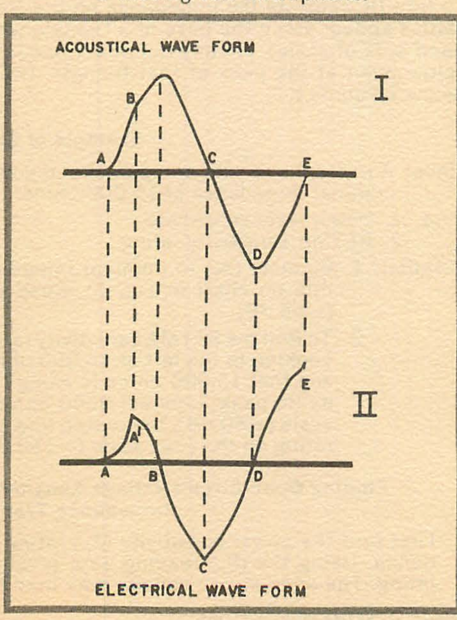
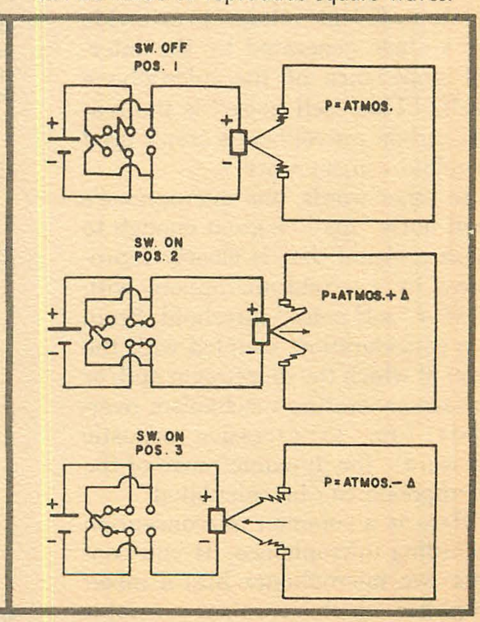


Fig. 12—How a moving-coil loudspeaker can be made to reproduce square waves.



that no moving-coil microphone can reproduce a square wave. *This, again, is of no real consequence,* but it does give some insight into the method of transduction.

Fig. 11 shows still another interesting theoretical consideration of the moving-coil microphone mechanism. In this case, we are going to assume a sudden transient condition. Starting at (A) on the acoustic waveform, the normal atmospheric pressure is suddenly increased by the first wavefront of a new signal and proceeds to the first overpressure peak, $(AT + \Delta)$ or (B). The diaphragm will reach a maximum velocity halfway to (B) and then return to zero velocity at (B). This will result in a peak A′ in the electrical waveform. From (B) on, the acoustic waveform and the electrical waveform will proceed as before, cycle for cycle, but 90° apart.

In this special case, peak A′ represents something extra. It may well be that, due to the myriad other problems (especially mass) encountered in a practical moving-coil microphone, this minute effect is swamped. It does illustrate, however, that given a "perfect," massless, moving-coil microphone, it does not follow that "perfect" electrical waveforms will result.

Moving-coil loudspeaker

An interesting side note is that, theoretically, a moving-coil loudspeaker can be made to reproduce a square wave. Fig. 12 illustrates how this can be done. If the cone of the loudspeaker is pushed backward it compresses the air in the limited enclosure. If the polarity is reversed, the speaker will move out, creating a slightly larger volume, and thus a lower pressure. If the voltage is removed, the cone returns to its normal position. Such a box, with proper air seals for the intrusion of a measuring microphone and a means to prevent loss of air pressure through the cone, would allow inspection of the phenomenon we have discussed (assuming the measuring system was capable of response down to d.c. and the loudspeaker had no mass).

Part 2

Microphone sensitivity, do-it-yourself sensitivity measurement, and directionality are discussed in this installment

Microphone sensitivity

Microphone sensitivity, both actual and apparent, can be discussed at this point. Moving-coil microphones, properly designed, can exhibit excellent sensitivity.

Actual sensitivity can be best visualized by recognizing the following: Some sounds can have such a low SPL at the frequency at which they occur that the pressure exerted on the microphone diaphragm does not create sufficient voltage to override the voltage generated by the internal impedance of the microphone itself. (This "self noise" is the reason carbon microphones aren't used in reinforcement work.)

In other words, one microphone's "self noise" may be great enough to mask a signal that is clearly reproduced by a second microphone with a lower "self noise" threshold. From this consideration, coupled with the level at which the diaphragm and/or voltage-generating mechanism overloads (due to excessive acoustic pressure), the dynamic range of the microphone can be calculated.

Here is a common misconception regarding microphones: If you connect two microphones into a mixer amplifier, whichever mixer control is set lowest for the same SPL at the microphones indicates the most sen-

USING THE MICROPHONE SENSITIVITY CONVERSION CHART

Microphone sensitivity can be specified by several different methods. To properly compare the rated sensitivity of two different microphones, their sensitivity ratings must be converted to the same method. The nomograph in Fig. 2 gives the relationship between:

1. The open circuit voltage (S_v). This is normally specified as dB above or below 1 volt if a sound pressure level of 1 microbar drives the microphone diaphragm.
2. The open circuit power (S_p). This is normally specified as dB above or below 1 milliwatt if a sound pressure level of 10 microbars drives the microphone diaphragm.
3. The RETMA sensitivity rating (G_m). This is normally specified as dB below 1 milliwatt if a sound pressure level of 0.0002 microbars drives the microphone diaphragm.

Finding the RETMA Impedance Rating (R_{mr})

To find (R_{mr}) locate the impedance on the Nominal Impedance in ohms scale. These values appear along the base of a triangle which has for a peak the (R_{mr}) values (left hand side of scale). For any nominal impedance along the base of a triangle use the value given at the peak of that triangle. (RETMA impedance ratings do not extend below 19 ohms.)

Example of Chart Use

Have: A microphone with an open-circuit voltage rating of 1 millivolt (—60 dB) and a nominal impedance of 15,000 ohms.

Find: 1. Power level sensitivity
2. RETMA sensitivity rating

Solution: 1. Connect (S_v) to nominal impedance in ohms with a straight line (solid line on chart in Fig. 2). Read power level sensitivity from (S_p) scale (—58 dB)
2. To find the RETMA sensitivity rating, locate the RETMA impedance rating. Looking to the left of 15,000 ohms on the nominal impedance scale, we see that 15,000 ohms is along the base of the triangle with 9600 ohms as its peak. Connect 9600 ohms to the open-circuit voltage on the (S_v) scale (—60 dB), the dotted line on chart, and read the RETMA sensitivity rating on the (G_m) scale (—150 dB)

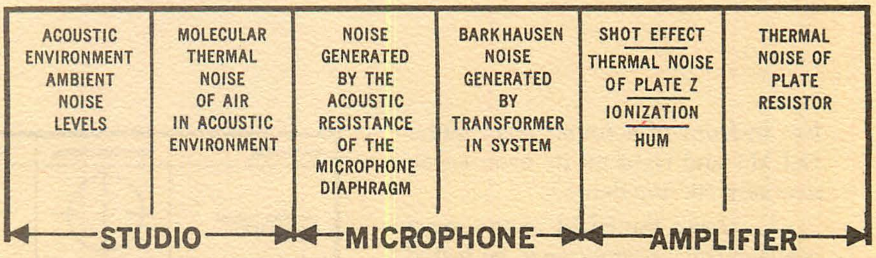
Finding Open-Circuit Voltage Sensitivity of Microphones Undergoing Impedance Transformation

1. First find the power sensitivity (S_p) rating of the microphone at its original impedance. Using the (S_p) reading as a pivot point, realign with the new impedance rating. The open-circuit voltage may then be read on the (S_v) scale.

References: RETMA Standard SE-105
"Microphones for Sound Equipment"—ASA Standard 224.1
"Acoustical Terminology"

sitive of the microphones. This is not entirely true. While it is an indication of the effective output of the microphone, and as such aids in designing input amplifiers, among other circuits, it does not tell the operator which microphone will pick up and reproduce weak sounds with the least noise.

Factors to be considered in evaluating "self noise" in microphone systems is shown in Fig. 1. When extremely quiet studios are used, all of these factors become almost equal in value if all design parameters are maximized. (Barkhausen noise is an exception to this. It is usually much lower in level than the other noise sources.)



NOTE #1 VOICE-COIL IMPEDANCE GENERATES SUFFICIENT THERMAL NOISE TO BE MEASUREABLE.

NOTE #2 IN THE CASE OF MOVING-COIL MICROPHONES, HUM PICKUP CAN BE PREDOMINANT NOISE.

Fig. 1—Sources of self-noise in microphone systems.

Apparent Sensitivity. Apparent sensitivity is another matter. Conversion charts in Fig. 2 allow quick conversion of a manufacturer's microphone ratings. The number of conversions that the chart and data sheet supplies indicates the complexity, confusion and consternation that microphone users face when approaching a strange manufacturer's new offering for connection to an existing sound system. The illustrations also allow a quick calculation of the point where microphone output drops below the sound system's electronic "equivalent input noise" (EIN) figure.

In using microphones in professional sound system work, every effort should be taken to obtain accurate, reliable threshold figures from the microphone manufacturer being specified.

Do-It-Yourself Measurement. In designing a control console for a sound reinforcement system in which a microphone you are not completely familiar with (and in many cases, those you think you are familiar with) is to be used, the following test is invaluable:

(1) Set up a test speaker in the manner shown last month (first article in the series) for measuring acoustic gain.

(2) Place this test speaker two

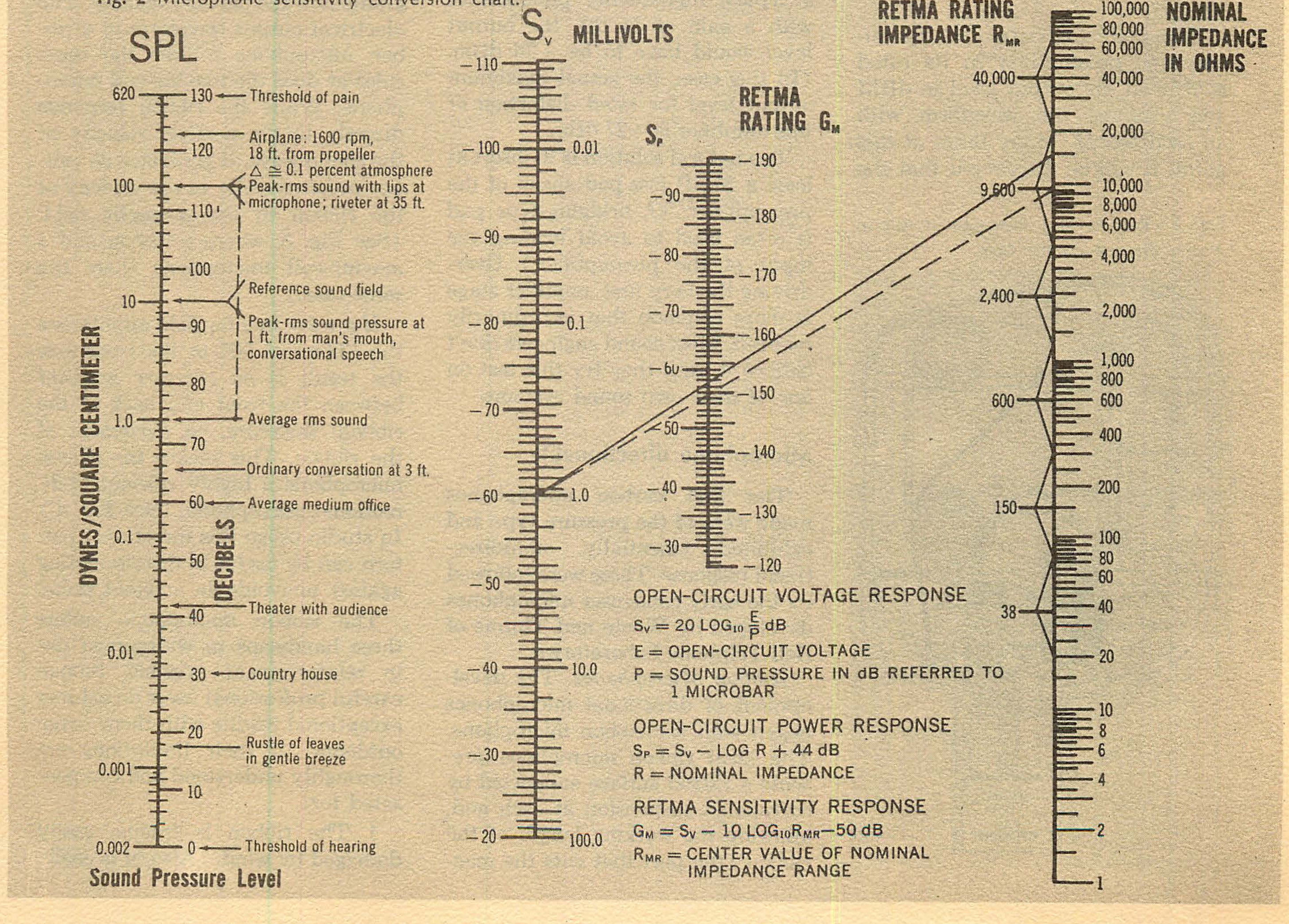


Fig. 2—Microphone sensitivity conversion chart.

feet in front of a sound level meter (SLM) and right on its most sensitive axis (0° incidence).

(3) Using the SLM in the exact position the microphone to be tested will occupy, set the test loudspeaker (using random noise for a source) to a reading on the SLM of 100 dB SPL. (This is the typical SPL generated if the lips are pressed against the microphone while talking.)

(4) Now substitute the microphone to be tested in place of the SLM. The output of this microphone will be led to the input of the microphone preamplifier used in the console; output impedance of the microphone should match the nominal source impedance of the microphone preamplifier. (Since microphones are not normally terminated at the input of the microphone preamplifier, and the actual input impedance of the preamplifier is usually many times that of the nominal source impedance rating; measurements of the microphone output alone does not tell the whole story.)

Terminate the output of the microphone preamplifier in its rated load, read the output level in dBM and examine the waveform with an oscilloscope. Many times at this point, it will be discovered that the

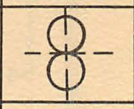
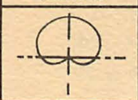
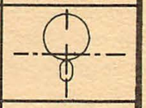
MICROPHONE	OMNIDIRECTIONAL	BIDIRECTIONAL	DIRECTIONAL	'SUPER-CARDIOID'	'HYPER-CARDIOID'
DIRECTIONAL RESPONSE CHARACTERISTIC					
RANDOM ENERGY EFFICIENCY (%)	100	33	33	27	25
FRONT RESPONSE / BACK RESPONSE	1	1	∞	3.8	2
FRONT RANDOM RESPONSE / TOTAL RANDOM RESPONSE	0.5	0.5	0.67	0.93	0.87
FRONT RANDOM RESPONSE / BACK RANDOM RESPONSE	1	1	7	14	7
EQUIVALENT DISTANCE	1	1.7	1.7	1.9	2
PICK-UP ANGLE (2θ) FOR 3 dB ATTENUATION		90°	130°	116°	100°
PICK-UP ANGLE (2θ) FOR 6 dB ATTENUATION		120°	180°	156°	140°

Fig. 4—Directional response characteristics of microphones.

microphone preamplifier is being grossly overdriven. Therefore, an input loss pad ahead of the preamplifier would be required (or else a less sensitive microphone).

It is not unheard of to encounter output levels from microphones that exceed −15 dBm in actual service (for example, a trumpet blown into it at 130 dB SPL). In the case of a typical high-quality preamplifier with a gain of 51 dB, the output level would have to be +36 dBm. (In this case, the actual maximum output level for rated distortion of this amplifier is +27 dBm.)

The natural solution is to insert at least a 20 dB loss pad ahead of the preamplifier. (A bridging-type pad is often used to avoid loading the input of the preamplifier.) Overdriving the very first amplifier stage is more common that is ordinarily realized. Most sound engineers don't believe it until they try this test on one of their own sound systems.

Microphone directionality

The first carbon microphones made were of the pressure type and exhibited essentially omnidirectional response. These were followed by the early condenser microphones developed by Wente and Thuras of Bell Telphone Laboratories.

Ribbon Microphones. The development of directional microphones began with the ribbon microphone. The ideal ribbon microphone presents a ridged surface supported by corrugated suspension at each end, allowing the working length of the ribbon (the part that cuts the magnetic field of force) to move as an unflexed plane through the magnetic lines of force.

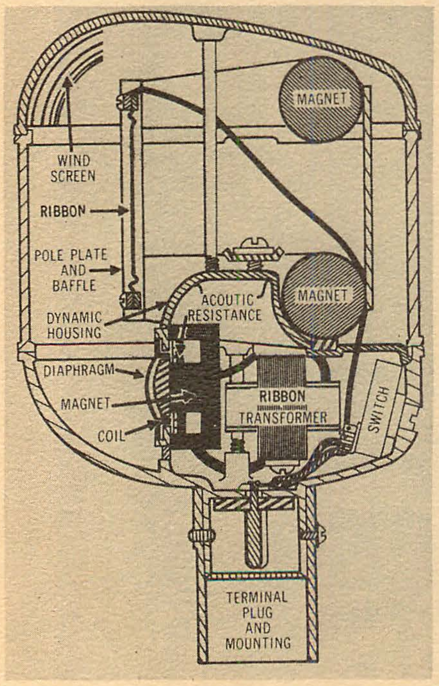
This metallic ribbon is suspended in the magnetic field and is freely accessible to air vibrations from both sides. The ribbon responds to the difference in pressure between the front and the back surface. In other words, it responds to the pressure gradient or the particle velocity. The electrical output waveform of a ribbon microphone follows the same rules as does the moving-coil microphone. While the ribbon responds directly to the particle velocity of the sound wave, it generates its voltage proportionally to the velocity of the ribbon in the magnetic field. (See Fig. 3, where a ribbon and a moving-coil microphone share the same case.)

If a wave in a far field approaches the ribbon at 90° to the front axis the result is *no pressure gradient* between the front and back of the ribbon; therefore, no movement of the ribbon. This makes the ribbon microphone a highly efficient bi-directional microphone. (See Fig. 4.) In studio usage this directional pattern can be useful in discriminating against unavoidable ambient noise.

The ribbon microphone suffers three handicaps to widespread use in reinforcement systems, though careful professional users do achieve exceptional results with them (even outdoors) if their limitations are thoroughly understood and compensated for:

1. The ribbon is fragile, easily damaged by wind. (Also, it is easily

Fig. 3 – Ribbon and moving-coil microphone share the same case.



damaged by being knocked down or dropped.) Thus performers should not (as they often do) test the microphone by blowing into it. When it can be shielded from the wind (such as in a wind-free theater in a protected ravine), or mounted where the performer cannot get closer than 6 feet from it (overhead or the far side of the footlights), damage can be minimized. To meet a psychological need, the performer can be given an empty case to carry in his hand while the pickup is made from overhead.

2. The ribbon microphone should not be used where it must be mounted near a large surface. Ribbon microphones placed against a wall lose sensitivity, for example. (The sound wave reflects off the

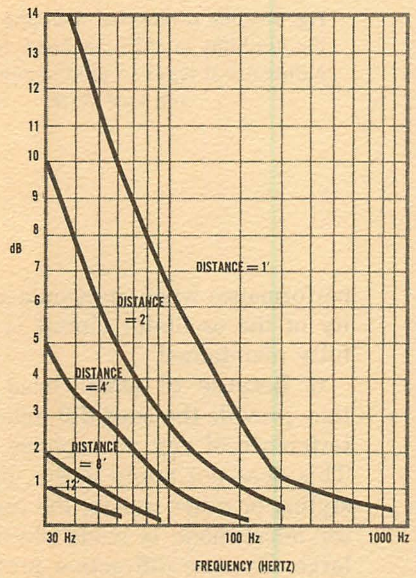


Fig. 5—The low-frequency response of a velocity microphone changes as it is approached due to a proximity effect.

wall to the rear of the ribbon, destroying the pressure gradient. The maximum response of a ribbon microphone will occur $2N - 1\lambda/4$* from the wall. In the case of the moving-coil pressure-type microphone, greatest output will occur if it is placed next to the wall since its maximum response occurs at the surface of the wall and at intervals of $2N\ \lambda/4$ from the wall.

3. As a ribbon microphone is approached, its low frequency *changes*. This proximity effect is charted in Fig. 5. There is a real necessity to keep performers away from this microphone for best results. The ribbon microphone makes an excellent narration microphone in studios but is usually misused by being placed on the table in front of the narrator rather than on a boom over his head. It should be mentioned that some performers have used the proximity effect to generate special tonal characteristics. In cases of this type, the microphone is used as a generating rather than as a reproducing instrument, and fidelity is dispensed with.

**N = an integer, λ = wavelength*

Cardioid Microphones. Cardioid means heart-shaped, and refers to the plane view of the polar pattern. Once the moving-coil pressure microphone and the ribbon velocity microphone were on hand, mathematically-minded people quickly saw an advantageous summation of their voltage outputs. Fig. 6 illustrates the results of such a combination—the cardioid microphone.

It is necessary to remember, in both the case of the cardioid and the velocity microphone, that the polar patterns, while illustrated in two dimensions, exist in three dimensions. That is, an omnidirectional microphone polar pattern is not a circle, but a sphere. The polar response of a bi-directional microphone resembles two spheres, and a cardioid's response resembles a pumpkin, with the microphone located at the stem position.

In well-designed microphones, the vertical polar response will closely approximate the horizontal response. (Many mysterious problems have been solved by discovering that, in "bargain microphones," the horizontal and vertical polar characteristics were not the same.

The ability to vary the rear lobe on a directional microphone is the best known method of picking up both a weak voice and a strong voice in the same area. Placing the strong voice on the rear side of the microphone

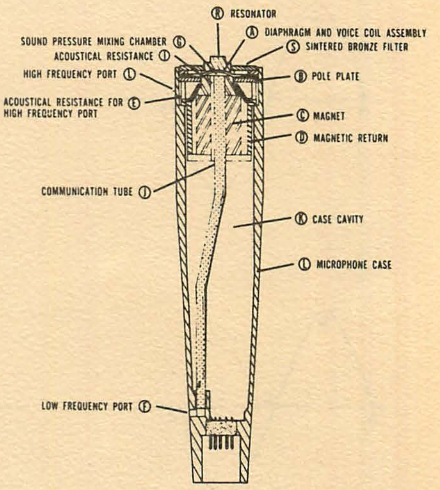


Fig. 6—Cross-section of a typical cardioid microphone.

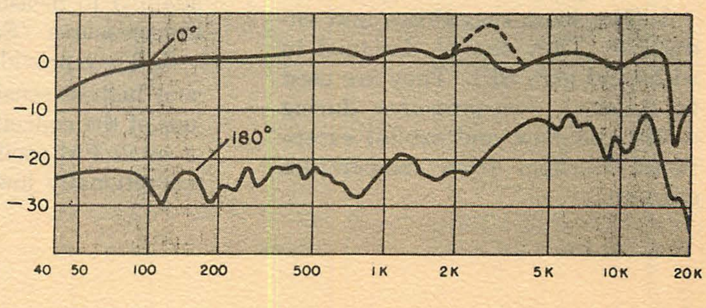
allows you to attenuate it (relative to the weak voice). The cardioid pattern can also be obtained from the moving-coil microphone alone if access to the rear of the diaphragm by the sound wave is carefully planned in relation to phase.

Figure 6 shows the internal construction of such a microphone, with its rear ports for high and low frequencies and the necessary communication tubes.

Fig. 7 details the frequency response taken first on the 0° axis, then on the 180° axis. Sound reinforcement engineers must be careful to avoid microphones whose 180° response introduces detrimental amplitude or phase changes, thereby triggering premature feedback problems. Recording and broadcast engineers, however, can safely ignore such details. (Where a Radio-TV show includes sound reinforcement to a live audience, an acoustical engineer is a worthwhile consultant to have available.)

Condenser-type Microphones. The condenser-type microphone comes closer to the perfect transducer than any other practical microphone in

Fig. 7 — Frequency response on 0° axis and 180° axis.



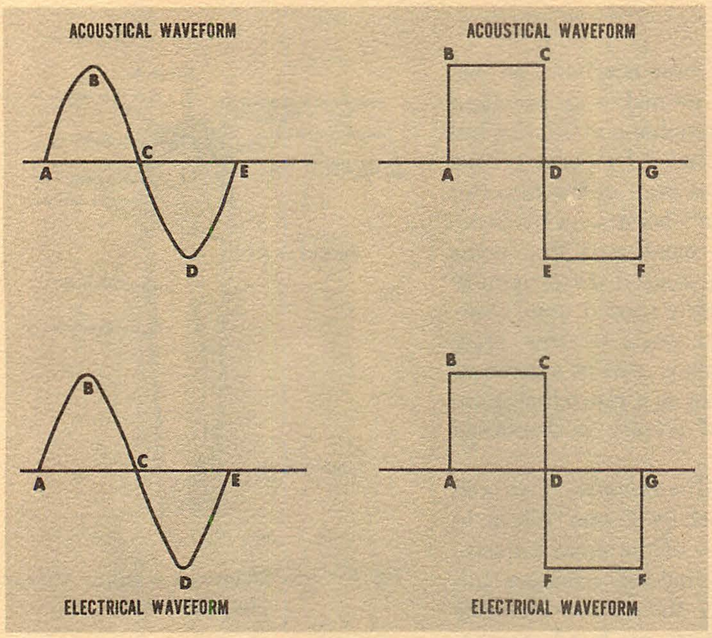


Fig. 8—Condenser microphones generate output signals in step with acoustic waves.

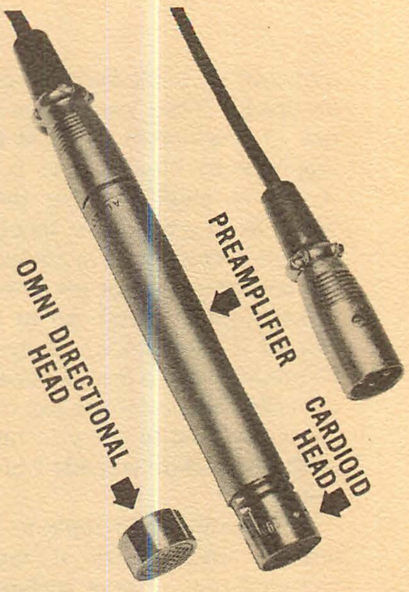


Fig. 9—Some condenser microphones offer the flexibility of changing heads from omnidirectional types to unidirectional types and vice versa.

use today. (This is not to say that poor condenser microphones are not produced, but the potential for superior performance is inherently available in the condenser microphone.) There are drawbacks, too, to be sure, including high cost relative to other types, bulkier size and potential for breakdown due to the need for a power supply.

Here are some of the advantages that the condenser microphone offers over other types:

1. Diaphragms with the least mass and greatest rigidity, yet providing sufficient output.

2. Smoothest, most extended, and most stable frequency response (witness the measuring microphones, such as: the Western Electric 640AA, Altec 21BR series, and the Bruel & Kjaer series, among others).

3. The most ruggedness and endurance. They can be designed to withstand high SPL. They are used to measure over-pressures during rocket launches, and exhibit exceptional freedom from temperature changes.

(In its original form, the condenser microphone was not so rugged. An American recording expert, carefully smuggling home the first post-World War II European condenser microphone system, found that it had a large gold-sputtered diaphragm, against which was a pressure-type electrical contact. The microphone had been brought back on a propeller-type aircraft, it seems, and the steady excitation of the diaphragm by the large amplitude, low frequency engine noises had worn the gold off the diaphragm at the contact point. The diaphragm was re-sputtered, making the unit operational again.)

4. They generate an output electrical waveform in step with the acoustical waveform and can be adapted to measure essentially d.c. overpressures. (See Fig. 8.)

5. They provide the lowest noise and highest sensitivity. The proper use of the capacitor head to control a small FM oscillator can result in exceptionally low-noise microphone performance when electronic stability of the oscillator circuit is carefully maintained.

6. Because of small head size, they provide the least diffraction interference of any microphone type. This does not apply where, due to styling or other causes, the condenser microphone is made physically large. Because diffraction problems appear at that frequency where the wavelength/4 = the diameter of the front of the microphone, it can be calculated why the very best quality microphones have diaphragms that range between ½-in. and 1-in. in diameter.

7. Versatility. It is possible to use one preamplifier base and carry both an omnidirectional head and a unidirectional head in your pocket to meet whatever requirements the situation at hand calls for. (See Fig. 9.)

Part 3

Condenser microphones and practical application notes

THE PRINCIPLE OF A condenser cardioid microphone is illustrated in Fig. 1. The sound field acts on both sides of the stretched diaphragm. To achieve the cardioid characteristic, sound pressure on the back side of the diaphragm is delayed by an acoustic network consisting of an acoustic resistance, R_1, and an acoustic compliance, C_1. Therefore, between the sound entrance on the front and the sound entrance on the rear of the microphone, there will exist a pressure difference proportional to the effective sound path between front and rear (which also will be dependent on frequency).

Figure 1 also shows the equivalent circuit for this cardioid microphone. Since the sound pressure difference, which constitutes the driving force for the diaphragm, increases with increasing frequency, and the output voltage of the microphone is proportional to the amplitude of the diaphragm, the diaphragm's movement has to be resistance controlled if the output of the microphone is independent of the frequency of the sound field. This resistance, "R," is formed by the thin air film between diaphragm and back plate.

In the graph of Fig. 2 (top), the sound pressure is plotted as a function of distance for a certain instant of time. "D" designates the effective sound path from the front to the rear sound entrance of the microphone. As can be seen, the pressure difference is, at first approximation, proportional to the distance, "D." It will increase with increasing frequency.

The sound wave coming from the rear will first hit the rear sound entrance of the microphone, acting on the back side of the diaphragm with a certain delay, "T," caused by the acoustic resistance, R_1, and the acoustic compliance, C_1. (See Fig. 2, bottom.) If this delay is equal to the time it takes the sound pressure to reach the front side of the diaphragm, both pressures will cancel each other, and the microphone will not be sensitive for this sound wave. Sound coming from the front side will first act on the front side of the diaphragm, then, after being delayed by the sound path between front and rear and by the acoustic network, it will act on the back side of the diaphragm. There will be a considerable difference between these two sound pressures. Accordingly, the microphone will be sensitive in this direction.

The arrangement discussed so far works satisfactorily only for micro-

Fig. 1—Principle of a condenser cardioid microphone and its equivalent circuit.

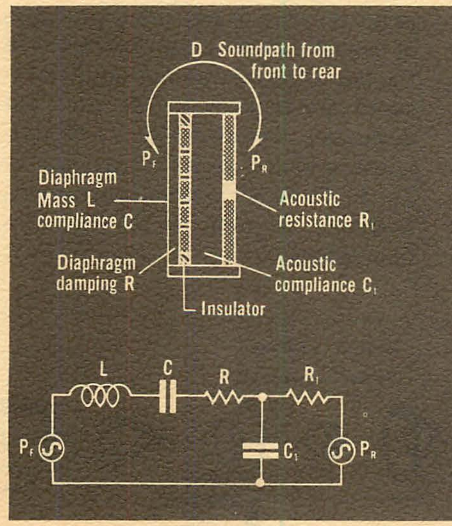


Fig. 2—Sound pressure for the microphone in Fig. 2. See text.

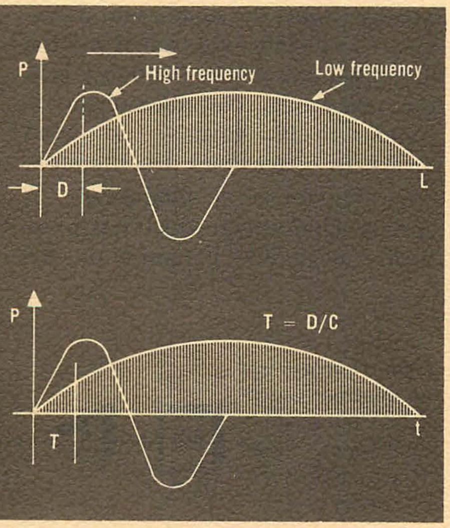
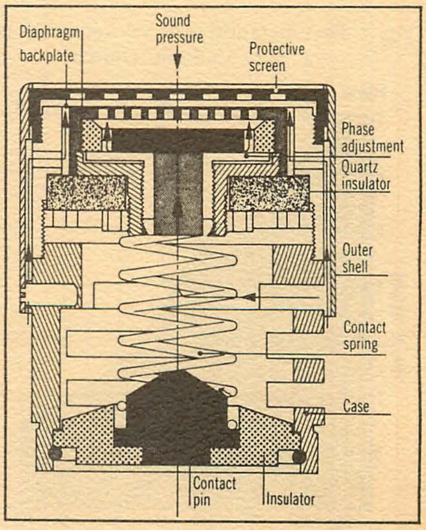


Fig. 3—Schematic cross section of a condenser cardioid microphone.



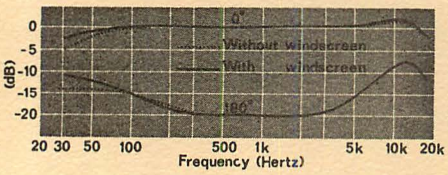


Fig. 4—Frequency response for 0-deg. and 180-deg. sound incidence.

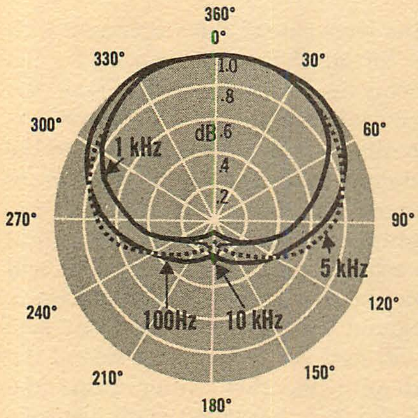


Fig. 5—Polar pattern of microphone used for plotting in Fig. 5.

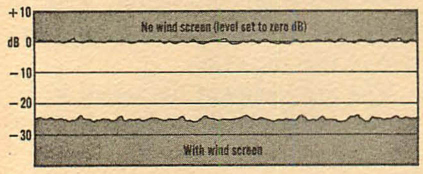
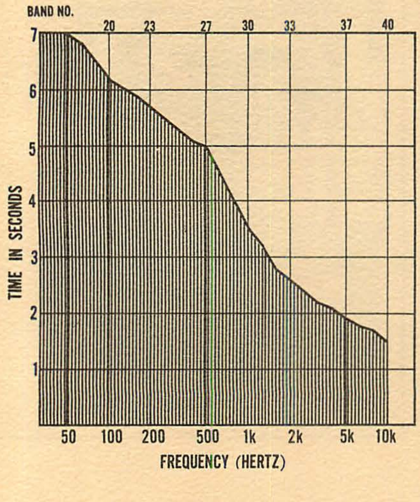


Fig. 6—Wind-noise reduction from a wind screen.

Fig. 7—Reverberation time vs. frequency in a large Catholic church.



phones of large diameter (1½ to 2-in.) which have an inherently long sound passage between front and rear, and which at high frequency provide directivity due to their size. Cardioid microphones of smaller size require much more complicated acoustic networks to achieve flat frequency response and good front-to-back discrimination over a wide frequency range.

Figure 3 shows a schematic cross-section of a cardioid microphone of this type. The maximum diameter is only ¾-in. and the total length, ⅞-in.

In this microphone, the sound pressure, which acts on the back side of the diaphragm, has to pass through several phase-delaying acoustic networks. One network consists of a narrow ring passage formed between the outer shell and the case of the microphone. This narrow passage decreases the propagation velocity, thereby providing the necessary time delay for the sound pressure. Another network consists of a narrow passage between two adjustable discs (phase adjustment and a cavity forming a R-C network similar to that discussed above.) The combination of these two networks can be adjusted so that the microphone has high sensitivity, flat frequency response, and good discrimination.

Figure 4 shows a frequency response for 0° and 180° sound incidence. The excellent low-frequency response and good discrimination at low frequencies are especially noteworthy. Since the diaphragm movement is resistance controlled and the diaphragm resonance frequency is in the mid-range, the microphone is comparatively sensitive to wind blasts. In many cases it is therefore advisable to use a wind-screen. A special wind-screen which has practically negligible influence on the frequency response (and even improves the discrimination at the low end) is available. The dashed curves in Fig. 4 show the response with wind-screen.

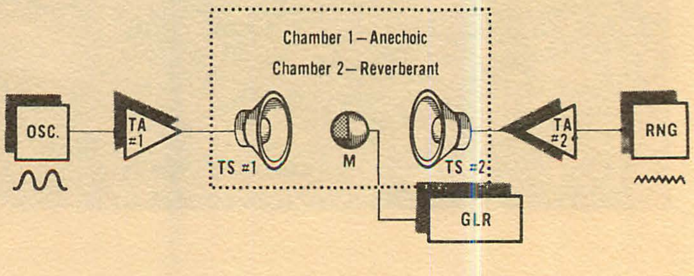
It is interesting to note the polar pattern of such a microphone for various frequencies (See Fig. 5). The pattern is nearly a perfect cardioid over most of the frequency range, and only slight deviations occur at the very low and very high frequencies. Figure 6 indicates the degree of wind-noise reduction available from the wind-screen.

Random notes for using microphones

The major technical points covered by this article are summarized in Chart I. The remainder of this exploration of microphones for sound reinforcement consists of a series of practical notes that have served the authors as a good mental checklist when "all is not going according to the book."

It is always helpful to remember that when a system is in a state of acoustical feedback, you are attempting to solve one of two problems: either the feedback is in-phase with the input and the gain has exceeded unity, or the phase, ϕ, of the system equal $2N\pi$ radians. In other words, you either have a big bump in overall house response or you have some slope in the response that creates a detrimental phase relationship. Acoustic measurements can point the way to pull additional dBs of acoustic gain by the dozens out of the hat if the bumps and slopes measured are equalized. The excellent discussion by Richard V. Waterhouse and the series of articles by Dr. C. P. Boner covers this subject, feedback, in great detail. (See bibliography.)

Fig. 8—Four-way test of effective use of a cardioid-pattern microphone tested as an omnidirectional unit.

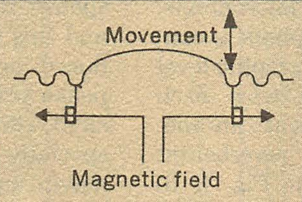
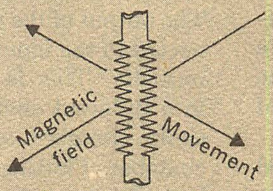
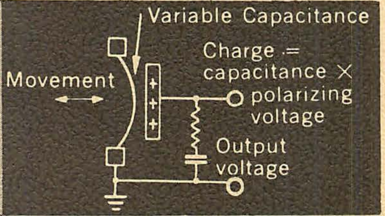
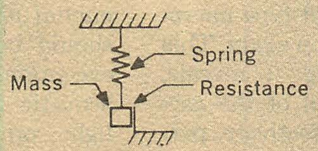
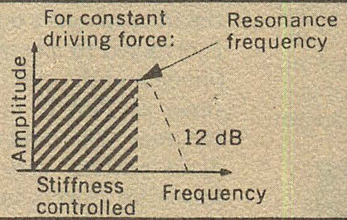
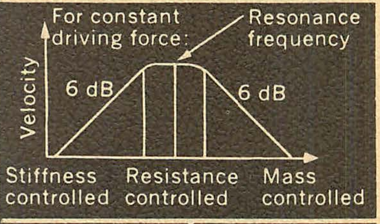
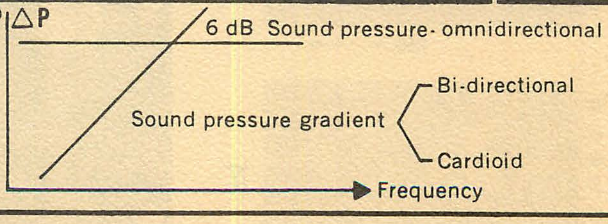


Directional microphones in reverberant rooms. Experience has shown that bi-directional and unidirectional microphones are of far less use in a reverberant space than often claimed.

In examining this, let us assume that the correct loudspeakers, electronics, time relationship, and sound distribution basics have been reasonably satisfied. (Usually, the choice of an inadequately or improperly chosen microphone is a manifestation of faulty equipment elsewhere in the system, as well.) One very common mistake today is to use super-directional line-source microphone types. These are designed for two special cases: (1) very absorbent surroundings (outdoors, heavily-treated rooms); (2) for recording or broadcast use. (Frequency response is sometimes so irregular here that it assures severe feedback problems in a reinforcement system.)

Still another problem is the common occurence of reverberation time rising at lower frequencies in the reverbant space. Figure 7 shows a typical reverberation time vs. frequency plotting for a large Catholic church. This indicates that the absorption of the room is not the same at all frequencies. Some front-to-back discrimination can be expected at high frequencies, but as the frequency is lowered ever-increasing energy is being uniformly diffused in the room, negating the effectiveness

Chart I—Microphone Principles

MICROPHONE TYPES	DYNAMIC MICROPHONES				CONDENSER MICROPHONES	
	MOVING COIL		RIBBON			
OPERATING PRINCIPLES						
OUTPUT VOLTAGE	VOLTAGE PROPORTIONAL TO VELOCITY				VOLTAGE PROPORTIONAL TO DISPLACEMENT	
PRINCIPAL MECHANICAL SYSTEM						
DIRECTIVITY	OMNI-DIRECTIONAL PRESSURE	CARDIOID PRESSURE GRADIENT	BI-DIRECTIONAL P. GRADIENT	CARDIOID P. GRADIENT	OMNI-DIRECTIONAL PRESSURE	CARDIOID P. GRADIENT
DIRECTIONAL EFFICIENCY	0 dB	5 dB	5 dB	5 dB	0 dB	5 dB
ACCESSORIES REQUIRED	None				Power supply	Power supply
IMPEDANCE	Low				Very high Preamp close to microphone	
FREQUENCY RESPONSE	Good	Fair	Good	Good	Very good	Very good
MECHANICAL RUGGEDNESS	Very good	Good	Good	Good	Very good	Good
RESISTANCE TO ENVIRONMENT	Good	Good	Good	Good	Good	Good
FORCES IN A PLANE WAVE SOUND FIELD	For constant sound intensity: Far field plane wave					

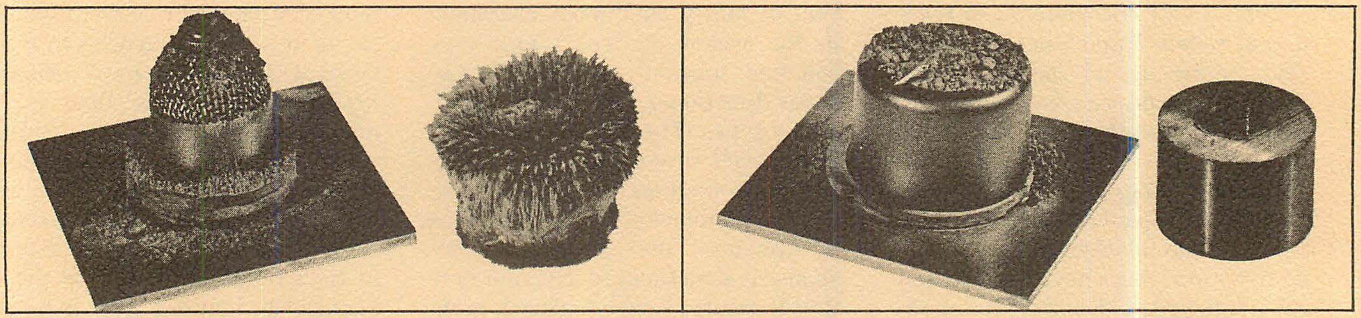


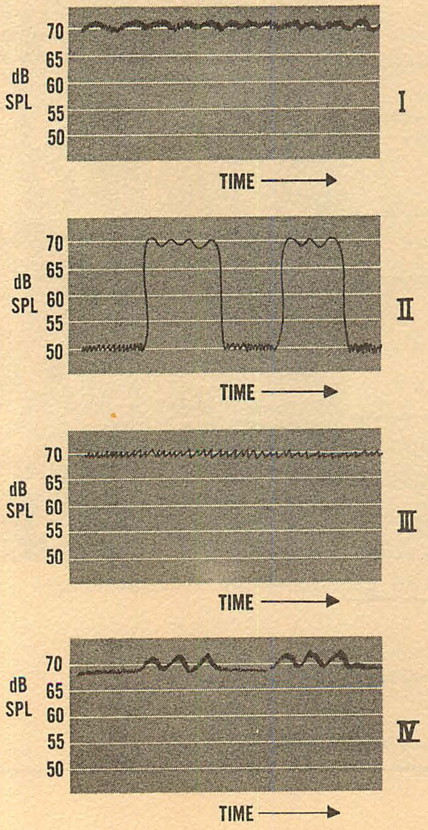
Fig. 10—Microphones should be protected against environmental hazards. Metallic filings at left, for example, can reach the inner mechanism of a microphone with inadequate protection. In contrast, a properly-protected microphone at right does not allow foreign objects to damage delicate mechanism.

of the unidirectional response. In practical terms, it means that unidirectional microphone output from a signal in a reverberant space becomes bass accentuated.

Unidirectional microphones work best in anechoic chambers, not in reverberant chambers. Figure 8 details a four-way test of the effective use of a cardioid-pattern microphone (M), tested as an omnidirectional unit.

Two signals adjusted to equal level at (M) are put into the chamber. One is an oscillator sine-wave signal

Fig. 9—Plotting in I and II, using test setup of Fig. 9, are made in an anechoic chamber. Plot I shows test as an omnidirectional unit; plot II as a unidirectional unit. Plots III and IV are done in a reverberant chamber.



fed through a test amplifier to test speaker #1, aimed at the zero axis of the microphone and turned on and off. The other is a random noise generator feeding noise through a test amplifier to test speaker #2, aimed at the 180° axis of the microphone. The output of the microphone is connected to the input of a graphic level recorder. This is done twice, each in a different environment: an anechoic chamber and a reverberant chamber. Plotting (I) in Fig. 9 illustrates the type of response that occurs: The sine wave and the random noise appear equally mixed in the microphone's output. Next, (M) is changed to a cardioid-type microphone and the test is run again. This time, as Plot II in Fig. 9 reveals, the rear discrimination of the cardioid pattern very effectively performs its work when the sine-wave signal is switched on.

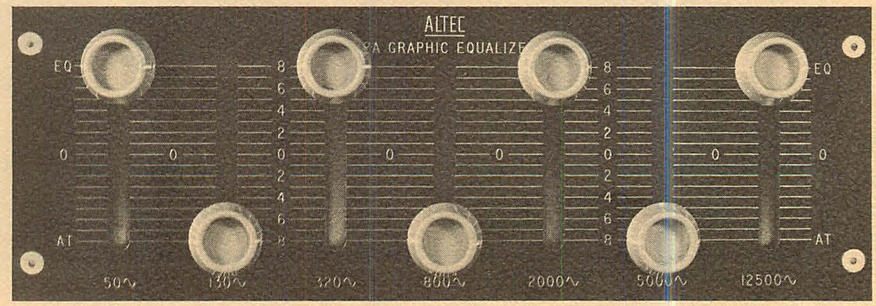
Now let's look at the results from the same test setup *in a reverberant chamber*. Plot (III) in Fig. 9 shows the omnidirectional microphone response of both signals. (The overall level is adjusted to the same level as that of the anechoic chamber. It would appear higher if the settings were left the same due to the useful reflections now also arriving at the diaphragm.) Plot (IV) in Fig. 9 reveals that little, if any benefit, is realized by the cardioid microphone in a reverberant chamber.

At this point it is well to remember that cardioids have, type for type, rougher random frequency response than omnidirectional units, and this roughness will result in reduced acoustical gain due to the peaks present. If a cardioid is desired, either the condenser unit or a specially selected and calibrated moving-coil cardioid should be used. A response chart of the specific microphone accompanies each calibrated unit.

Shock-mounted and environmental protection. All microphones should be shock mounted without exception. Just a few experiences in acoustic measurements shed much light on sympathetic resonances wreaking havoc on rigidly-mounted microphones. Microphones should also be protected against environmental hazards to avoid costly damage. See Fig. 10.

Use of multiple microphones. In reinforcement systems, each additional microphone brought up in gain reduces the maximum gain available from anyone of them by approximately 3 dB. Even worse, if the space is reverberant, it results in the

Fig. 11—A graphic equalizer can be used to smooth frequency response, achieving maximum acoustic gain.



reverberation being picked up and re-amplified by each additional microphone. Provisions for a skilled operator, or at the very least, interlocking relays activated by floor mats near the microphones, are much better answers than trying to turn all of them up at the same time. In some situations, all may need to be on. When this is the case, narrowband equalization can perform wonders.

Broadband equalization. If a quality microphone(s), electronics, and speaker system have been selected and installed in correct relationship to each other, a graphic equalizer, such as one shown in Fig. 11, can be used to "smooth" the overall acoustic response for maximum acoustic gain. This is the very minimum that should be done in the way of equalization. Use of a ⅓-octave filter set with a random noise generator as source and walking the audience area with a sound level meter will allow surprisingly smooth response curves via the graphic equalizer. Once you have heard this simple equalization accomplished *you will never fail to do it for every subsequent sound reinforcement system with which you may be involved.*

Narrowband equalization. "Miracles" are being accomplished today under extremely trying acoustical conditions by following the overall smoothing of the acoustic response with specific narrowband filtering of individual feedback modes. (This subject has been treated extensively elsewhere, and the reader is referred to the bibliography.) Suffice it to say that narrowband equalization is a very specialized business and only the best professional help should be engaged if it is contemplated. An enormous amount of time can be wasted trying to "hunt and peck" at this technique, with much money squandered without any results to show for it. When professionally done, the improvements can range from very good to startling.

In sum, microphones are more complex than the average sound engineer often realizes. We have barely nicked the surface of a vast collection of data. It is hoped that by pointing out, in a non-mathematical way, a few of the broad concepts that hint at underlying complexities, that some of you will be encouraged to dig deeper and enjoy the multi-dimensioned mathematical inter-relationships as well. The bibliography was compiled to allow the reader to trace the thinking presented here, as well as to encourage him to venture further.

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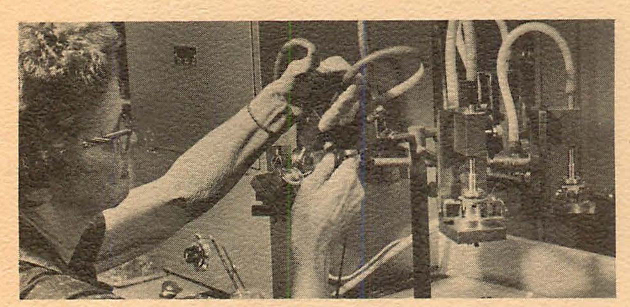
QUALITY CONTROL

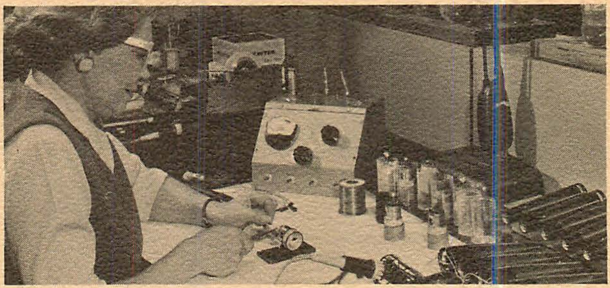
Another big reason behind the success of Altec microphones lies in the reliability and integrity of construction that is built into every microphone unit. This achievement is the product of three important operations: Detailed examination of purchased materials to determine their conformance to design requirements and specifications; assembly of each microphone unit by hand by thoroughly trained and skilled factory technicians; and implementation of a rigid quality control procedure, consisting of eight in-process inspections and tests, a final inspection, and a production acceptance test conducted in Altec's anechoic production test chamber.

An all-important, three-way guarantee to the consumer that actual performance will measure up to the stated specifications.

Production acceptance test run on 683B Microphones in an Altec Anechoic Chamber for precise response measurements.



Cementing voice coil and diaphragm subassemblies onto microphone pressure unit.



Soldering microphone pressure unit assembly into microphone chassis.

Altec microphones are engineered and manufactured to the same high standards of quality that have made "Voice of the Theatre"® speaker systems, Altec audio controls, monitors and other sound equipment the standard of the industry for so many years.

Take our Solid State Condenser Microphone Systems (M49 Series), for example. Extremely wide, smooth frequency response. Front-to-back discrimination of 20 dB. Omnidirectional or cardioid types. Battery or AC operated. Lightweight but rugged, with power supplies to match. Altogether, these fine, precision-made instruments are the most advanced professional mikes on the market today.

The M49 is typical of the complete Altec mike line, which includes selectable pattern types, miniature lavaliers, close-talking models and other solid-state condenser types, plus mounts, wind screens and accessories.

So go ahead and put Altec on. Why not start by asking your Altec Sound Contractor for complete technical data? He's listed in the Yellow Pages under "Sound Systems." Or, if you prefer, write direct to us at 1515 S. Manchester Ave., Anaheim, Calif. 92803.

ALTEC LANSING

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