

A Small Unidirectional Dynamic Probe Microphone*

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This paper describes a unidirectional dynamic microphone of small size utilizing the "uniphase" system.¹ Under idealized conditions the elements of the phase shift network employed can be shown to be functions of the independent variables: (a) the inductance of the back entry, (b) its resistance and (c) the distance of the two sound-entries. Under real conditions certain modifications of the phase shift network are necessary; these are described, as is a partially laminated Mylar diaphragm used.

THE requirements for a pressure-gradient unidirectional microphone for use in public address systems can be summarized as being: small size, in order to be unobtrusive; a suitable frequency response and output level available from low and high impedance output connectors; a random energy response equal or less than 1/3, therefore indicating a cardioid, super-cardioid² or hyper-cardioid³ polar pattern as desirable; a uniform discrimination over the audio spectrum transmitted, to avoid feedback at frequencies otherwise favored by the microphone; a polar pattern of rotational symmetry having the major axis of the microphone as axis of rotation, so that positioning of the microphone or of the loudspeakers becomes less critical; a stable acoustic phase shift network within the microphone to insure proper acoustic performance during its life time—an objective not easily attained with network components (particularly mechanical) required to be in delicate balance for a long period of time; and, finally, low mechanical noise sensitivity as well as high blast or "pop" resistivity.

The requirements of both small size and uniform discrimination over the audio-frequency range are somewhat contradictory, because the polar pattern of a pressure-gradient microphone as well as the characteristics of its phase shift network collapse at certain frequencies. In order to maintain proper performance at these frequencies, use is made of the phenomenon of diffraction of sound by the microphone case. Generally, the smaller the case dimensions, the higher the frequencies of incident sound must be to cause desirable diffraction effects, demanding, therefore, an increased frequency range of the phase shift network.

A solution of these problems has been attempted with the design of a first order pressure-gradient microphone based on a refined version of the "uniphase" system.⁴ This system utilizes a single transducer together with an acoustic phase shift network so arranged as to react to the phase shift $2\pi f(d/c)\cos\theta$ which occurs between two points in space immersed in a sound field of plane progressive waves,

where d is the distance between the two points, c the velocity and f the frequency of sound, and θ the angle between the direction of incident sound and the line connecting both sound entries (Fig. 1).

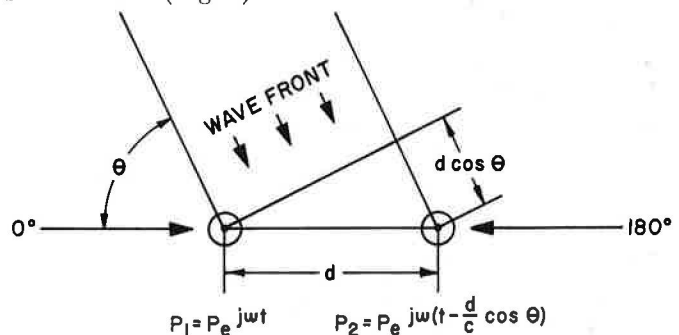


FIG. 1. Two point array.

The application of this system to a moving coil microphone is depicted in Fig. 2, which shows a simplified sectional view of the acoustically interesting part of the microphone as well as its acoustical network. L_3 , R_3 and C_3 represent the inductance, acoustical resistance and acoustical compliance of the diaphragm and voice coil assembly; C_a represents the acoustical capacitance of the cavity between diaphragm and poles of the magnetic structure; L_b and R_b represent the inductance and resistance of the slits formed by voice coil and poles and of the screen A placed adjacent to the air gap; C_b represents the acoustical capacitance of the cavity within the magnetic structure; L_1 and R_1 represent the inductance and the acoustical resistance of the annular slit which surrounds the magnetic structure and which contains the screen B; P_1 is the pressure at the diaphragm while P_2 is the pressure at the back opening of the annular slit; d is the "effective" acoustical distance between the center of the diaphragm and the back opening (d is usually taken as the shortest unobstructed distance between these two points and is assumed to be independent of frequency in the following derivations); $u = Av$ is the volume current in the branch Z_3 , where v is the velocity of the diaphragm and A is its effective area (A is also assumed to be independent of frequency).

With the help of the acoustical network shown in Fig. 2,

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¹ B. B. Bauer, *J. Acoust. Soc. Am.* 13, 41 (1941).

² B. B. Bauer, *Electronics* 15, 31 (1942).

³ R. P. Glover, *J. Acoust. Soc. Am.* 11, 296 (1940).

⁴ B. B. Bauer, *J. Acoust. Soc. Am.*, *op. cit.*

A SMALL UNIDIRECTIONAL DYNAMIC PROBE MICROPHONE

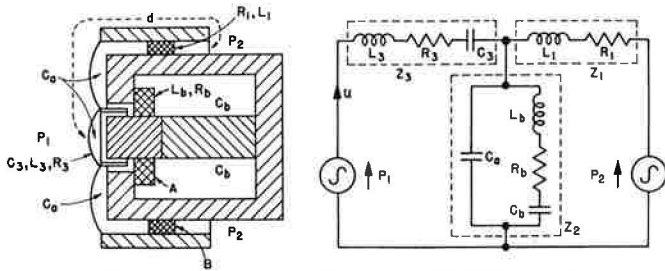


FIG. 2. Cross-sectional view and acoustical network.

the volume current through impedance Z_3 (diaphragm) can be expressed as

$$u = \frac{[(1 + Z_1/Z_2) P_1 - P_2]}{[Z_3 (1 + Z_1/Z_2) + Z_1]} \quad (1)$$

where $Z_3 = pL_3 + R_3 + 1/pC_3$, $Z_2 = (pL_b + R_b + 1/pC_b) || 1/pC_a$ and $Z_1 = pL_1 + R_1$, with $p = j\omega = j2\pi f$. Assuming a field of plane progressive waves, then $P_1 = P \exp j\omega t$, $P_2 = P \exp j\omega (t - K \cos \theta)$, with $K = d/c$, and the amplitude of the volume current is

$$U = P \frac{[(1 + Z_1/Z_2) - e^{-pK \cos \theta}]}{[Z_3 (1 + Z_1/Z_2) + Z_1]} \quad (2)$$

and, since $U = 0$ for $\theta = 180^\circ$ is desired—in order to obtain a cardioid pattern—the condition

$$1 + Z_1/Z_2 = e^{pK} \quad (3)$$

must be fulfilled. It is to be noted that condition (3) does not include requirements for Z_3 , i.e., for the diaphragm parameters.

The network of Fig. 2 does not permit an analytical solution of (3); the approximate values of the network parameters have been ascertained by substitution, trying to obtain in the relation

$$1 + Z_1/Z_2 = ae^{\beta pK} \quad (4)$$

the values $a = a(\omega) \approx 1$ and $\beta = \beta(\omega) \approx 1$ for as wide a frequency range as possible, where a and β are real functions of ω .

For low frequencies, however, it is possible to obtain

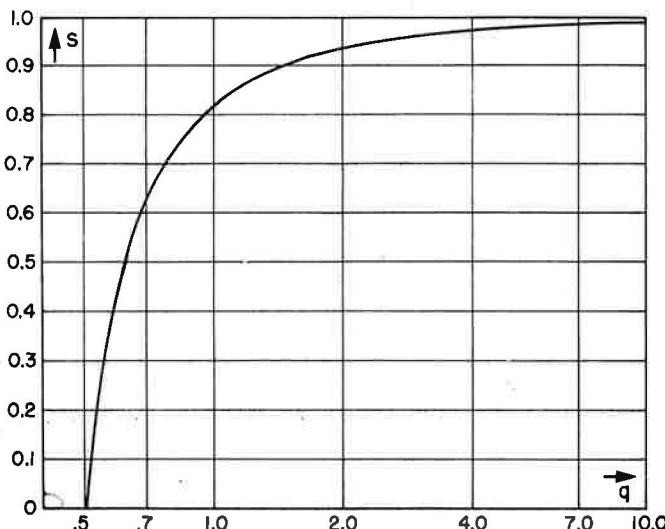


FIG. 3. Auxiliary variable s as function of q .

values for L_b , R_b , C_b and C_a as functions of two independent variables, R_1 , L_1 and of one design parameter, d , best chosen in the form: $q = L_1/KR_1$. These three are particularly useful for estimating the frequency response (i.e., response for $\theta = 0^\circ$) and level of the system at low frequencies.

If $R_b > 0$, the expression on the left of Eq. (3) can be developed by means of McLaurin's expansion and the coefficients of p^n , $n = 1, 2, 3, 4 \dots$ compared with those of the series expansion of $\exp pK$, terminating the series after $n = 4$. The following group of equations can then be obtained:

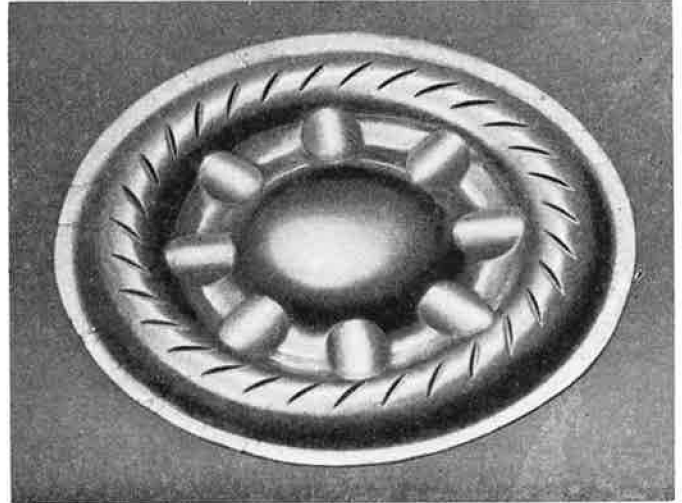


FIG. 4. Photograph of diaphragm.

$$\begin{aligned} K &= R_1 (C_a + C_b) & (5) \\ K^2/2 &= L_1 (C_a + C_b) - R_1 R_b C_b^2 \\ K^3/6 &= C_b^2 \{R_1 C_b [R_b^2 - (L_b/C_b)] - L_1 R_b\} \\ K^4/24 &= C_b^3 \{L_1 [R_b^2 - (L_b/C_b)] + \\ &\quad C_b R_1 R_b [2 (L_b/C_b)] - R_b^2\}. \end{aligned}$$

With

$$q = L_1/KR_1 \quad (q > 1/2)$$

the equations can be rearranged to yield

$$C_b = (K/R_1) s = (L_1/R_1^2) (s/q) \quad (6)$$

$$C_a = (K/R_1) (1-s) = (L_1/R_1^2) [(1-s)/q] \quad (7)$$

$$R_b = R_1 [(2q-1)/(2s^2)] \quad (8)$$

$$\begin{aligned} L_b &= (KR_1/12s^3) \{3(1-2q)^2 - \\ &\quad 2s[1-3q(1-2q)]\} = \\ &\quad (L_1/q^3 12) \{3(1-2q)^2 - \\ &\quad 2s[1-3q(1-2q)]\}. \end{aligned} \quad (9)$$

The auxiliary variable $s = s(q)$ is determined by a quadratic equation, the parameters of which are themselves functions of q :

$$s^2 \{1 - 4q [1 - 3q (1 - 2q)]\} - 4s (1 - 2q) [1 - 3q (1 - 2q)] + 3 (1 - 2q)^3 = 0. \quad (10)$$

Of the two roots, only the larger one will yield positive circuit parameters, as will $q > 1/2$. The function $s = s(q)$ is shown in Fig. 3. Eqs. (5) to (9) show the rather simple

relationship between the network parameters and L_1 and R_1 , while the dependency on q is more complicated. (The term $1 + Z_1/Z_2$ can be shown to depend only on K and q). In order to ascertain the frequency at which a given deviation from the desired values $a = 1, \beta = 1$ occurs, a and β could be calculated using Eqs. (4) to (10). However, it is simpler and sufficiently accurate to use an electrical analogue of Z_1 and Z_2 and measure the transfer characteristic of the L section, as shown in Fig. 2, with Z_3 removed.

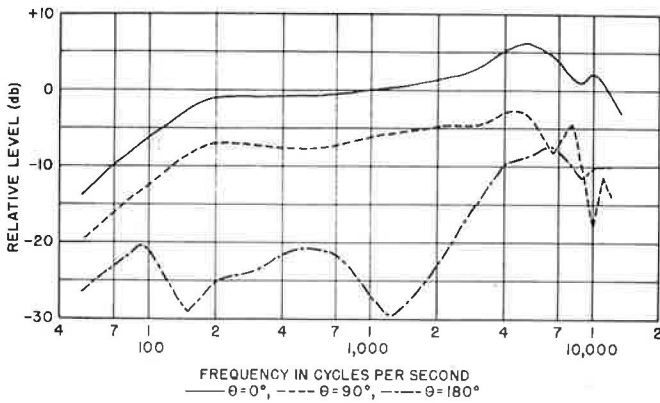


FIG. 5. Frequency response for $\theta = 0^\circ, 90^\circ$ and 180° .

The influence of a and β can be best seen by substituting Eq. (4) into Eq. (2), thereby obtaining for the amplitude of the volume current

$$U = P \left\{ [1 + a^2 - 2a \cos(\beta + \cos \theta) K\omega]^{1/2} / [Z_3 [1 + (Z_1/Z_2)] + Z_1] \right\} \quad (11)$$

which will yield, for $a = 1, \beta = 1$

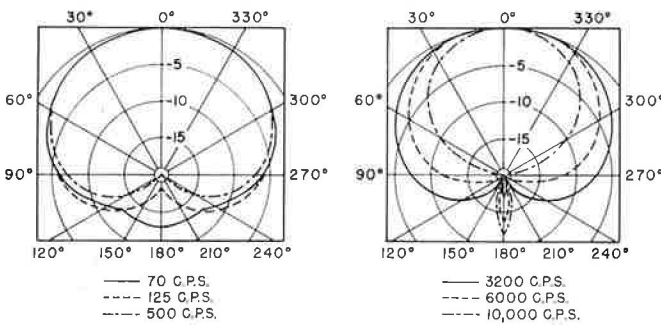


FIG. 6. Polar response for several frequencies.

$$U = 2P \left\{ [\sin(K\omega/2)] (1 + \cos \theta) / [Z_3 [1 + (Z_1/Z_2)] + Z_1] \right\} \quad (12)$$

Assuming small $K\omega$, such that $\sin(K\omega/2) \approx K\omega/2$, then

$$U = P \left\{ K\omega (1 + \cos \theta) / [Z_3 [1 + (Z_1/Z_2)] + Z_1] \right\} \quad (13)$$

$U = U(\theta)$ describes a cardioid. With increasing $K\omega$, the cardioid pattern will deteriorate, and finally for $K\omega > \pi/2$ or $d > \lambda/4$ the cardioid pattern collapses.

For values of a and β other than 1, numerous polar patterns may be obtained. $a = 0$, for instance, represents an

omnidirectional pattern; $a = 1, \beta = 0$ a cosine; $a = 1, \beta = 1/3$ a hyper-cardioid;⁵ $a = 1, \beta = 37/63$ a super-cardioid polar pattern.⁶ $\beta < 1$ establishes a minimum at $\cos \theta = -\beta$ of the value

$$U = P \{ (1 - a) / [Z_3 [1 + (Z_1/Z_2)] + Z_1] \}.$$

Therefore, the interpretation of the polar pattern of the microphone immersed in a field of plane progressive waves provides a very useful tool for the evaluation of the phase shift network employed.

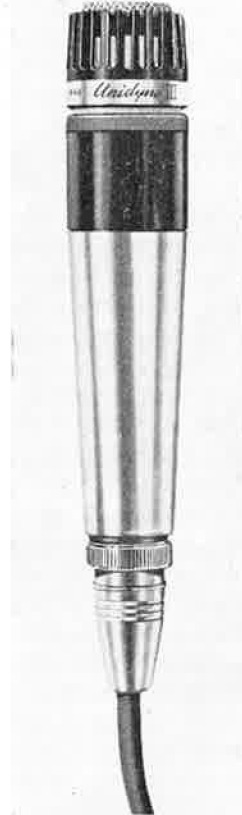


FIG. 7. Photograph of microphone.

The choice of q, L_1 and R_1 is dictated by both the desired front response performance and the desired output level of the microphone. Assuming front response ($\theta = 0$), low frequencies (where $a \approx 1, \beta \approx 1, \cos \omega K \approx 1, \sin \omega K \approx \omega K$), and $R_3 \ll R_1$ and $L_3 \approx L_1$ Eq. (11) may be simplified to

$$U \approx 2PK\omega / \left(R_1^2 + \omega^2 \{ L_1^2 + [2(q-1)/q] L_1 L_3 [1 - (f_0^2/f^2)] + L_3^2 [1 - (f_0^2/f^2)]^2 \} \right)^{1/2} \quad (14)$$

where $f_0 = 1/2\pi (L_3 C_3)^{1/2}$ is the resonance frequency of the diaphragm in vacuo. For q large, so that $(q-1)/q \approx 1$, Eq. (14) may be simplified to

$$U \approx 2PK\omega / \left\{ R_1^2 + \omega^2 (L_1 + L_3)^2 [1 - (f_1^2/f^2)]^2 \right\}^{1/2} \quad (15)$$

with $f_1 = 1/2\pi [(L_1 + L_3) C_3]^{1/2}$ representing the resonance frequency of the back entry inductance L_1 plus diaphragm

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⁵ R. P. Glover, *op. cit.*

⁶ B. B. Bäuer, *Electronics, op. cit.*

inertance L_3 in conjunction with the diaphragm compliance C_3 . Equation (15) can be represented by a circuit in which the pressure $2PK\omega$ acts upon a series-impedance consisting of R_1, L_1, L_3 and C_3 . At frequencies well below f_1 the system is stiffness-controlled and $U \approx 2PKC_3\omega^2$ shows a 12 db per octave rise, while for frequencies well above f_1 the system is mass controlled and $U \approx 2PK/(L_1 + L_3)$ is independent of frequency. The intermediate region can be controlled, i.e., the diaphragm can be suitably damped, by the acoustic resistance R_1 . The resonance frequency f_1 can be decreased by increasing the acoustic inertance L_1 of the back-entry path without altering the mechanical resonance frequency f_0 of the diaphragm. This is a useful method for decreasing the mechanical shock sensitivity of the microphone without changing its frequency response at the lower frequencies. However, the increase in L_1 also decreases the level of the microphone $U \approx 2PK/(L_1 + L_3)$, since L_3 is mainly a voltage-generating mass represented by the voice coil.

It must be observed that the equations derived above are based on a number of assumptions: neither d , the effective acoustical distance between the diaphragm and the back opening, nor the effective area of the diaphragm, which is used to transform the mechanical impedance of the diaphragm into its acoustical analogue Z_3 , are independent of frequency; in addition, the calculations are based on a sound field consisting of plane progressive waves (in the case of spherical waves, for instance, d is dependent on the distance to the sound source); only at low frequencies, furthermore, is it permissible to represent the network parameters by lumped circuit elements; finally, the effects of the diffraction of sound around the microphone have been neglected, although diffraction contributes significantly to the unidirectional characteristics at high frequencies.

The physical forms of the circuit elements, therefore, have to be given a suitable geometry in order to present distributed impedances which—in combination with the effects of diffraction—will perform properly at high frequencies. Also, in order to balance some of these effects, it may not always be most favorable to aim at the condition $\alpha = 1$, $\beta = 1$ over the entire frequency range of the phase shift network.

Based on the foregoing, a microphone has been designed meeting the requirements previously outlined. Particular attention has been given to the development of a suitable diaphragm, which is shown in Fig. 4. It is molded of Mylar polyester film, a material admirably suited for this purpose because of its excellent physical and chemical characteristics, which are retained over a long period of time. The diaphragm is partially laminated and is contoured to obtain the greatest possible ratio of the fundamental to the frequency of the first overtone, in order to insure a smooth frequency response. The contour, by means of suitably molded passages, also permits relatively unimpeded vibrations of air underneath the surface of the diaphragm, especially in the area where the voice coil connects the diaphragm, thereby reducing an acoustical resistance component otherwise included in R_3 . Another advantage of this contour is the ease with which the diaphragm compliance can be controlled by

only one dimension of the contour, without changing the thickness of the diaphragm material. Of course, larger changes of compliance can be obtained by changing the thickness of this material—production techniques have been developed which easily permit the forming of Mylar as thin as $\frac{1}{4}$ mil.

The frequency response of this system for $\theta = 0^\circ, 90^\circ$, and 180° is shown in Fig. 5, while the polar response for various frequencies at a distance of 24 in. from the sound source is shown in Fig. 6. As can be seen, a very slightly distorted cardioid at 70~ with a 180° -discrimination of 13 db develops into a true cardioid at 125~ with a 6 db 90° -discrimination, therefore indicating a random energy response of $1/3$. The 90° discrimination increases with frequency and reaches a value of 20 db at 10 kc, while at 3.2 kc a back lobe appears, which increases in magnitude to a 180° -discrimination of 12 db at 10 kc, tending towards a hyper-cardioid performance with a random energy response of $1/4$. In order to insure rotational symmetry of the polar pattern, all microphone parts contributing to acoustical performance have rotational symmetry with the major axis of the microphone as the axis of rotation.

Figure 7 is a photograph of the completed microphone showing a microphone cartridge with its magnetic structure, diaphragm and a part of the phase shift network. The microphone case provides room for the acoustic capacitance C_3 , a microphone transformer, and a shock mount designed to isolate the cartridge from mechanical vibrations transmitted through the microphone case. A grille in front of the diaphragm provides suitable blast protection. The case end has a conical configuration and can be inserted into a swivel mount, if desired.

The performance of this microphone in talk tests under severe conditions is excellent, proving the validity of the design principles outlined above.

THE AUTHOR



Charles Ernest Seeler was born on April 8, 1920. He studied at the Institute of Technology in Berlin from 1939 to 1940, the Institute of Technology of Darmstadt from 1940 to 1944, and received his B.S. degree in 1942 and his M.S. degree in physics in 1944.

Before joining Shure Brothers, Inc., he worked at the Ringsdorf Werke in Germany, the University of Notre Dame, and Electro-Voice, Inc. Mr. Seeler joined Shure Brothers, Inc., in 1953, in a research and development capacity, specializing in phonograph reproduction. He is currently engaged in microphone development.