Design Parameters of a Dual Woofer Loudspeaker System*

EDWARD M. LONG

Consumer Equipment Division, Ampex Corporation, Elk Grove Village, Illinois

During the initial design stage of a new loudspeaker system, the inability of many compact bookshelf loudspeaker systems to reproduce, with realism, the "impact" sounds of percussion instruments was related in part to the acoustic response shape. An acoustic method of increasing the output in the 200 to 600 Hz range was found in the mutual coupling of two woofers. Woofer and tweeter design and the effects of mounting on tweeter response are discussed. Crossover technique and the effects upon acoustic output due to driver phasing are also discussed.

INTRODUCTION The design of any loudspeaker system includes many interrelated and sometimes conflicting factors. After specifications have been determined, the next important step is the selection of the most suitable design approach or philosophy. Since there are a number of possible approaches available this is not as easy as it might first appear. After selecting a general design philosophy the actual design of the various component parts of the total system may begin. Each component should be engineered to provide maximum performance from the total system. For the purpose of example the design parameters of a specific loudspeaker system (see Table I) will be discussed, with the intention of presenting data that will be useful in the design of other loudspeaker systems. Comparisons will also be made with a loudspeaker system intended for similar general application and of similar cost. This will allow conclusions to be drawn as to the suitability of the selected design philosophy and specific design parameters.

DESIGN CONSIDERATIONS

There are in the literature many excellent discussions of certain parameters of the small sealed enclosure. Arguments can be found both for and against its use. This type of enclosure was chosen as optimum considering the particular application and cost factors. One of the main disadvantages of the small sealed enclosure is the difficulty of achieving adequate bass output while maintaining good efficiency through the mid and upper range of the bandwidth desired. Usually, only the output of the bass transducer is affected by the internal volume of the enclosure of a loudspeaker system which utilizes separate transducers for the bass and treble ranges. The difficulty of adjusting parameters of such a system to optimize for both bass response and efficiency is somewhat less, but the problem remains. This is because the acoustic output in the piston band or mass-controlled region of the bass transducer is still interrelated with the acoustic output in the resonance range. Optimizing the design parameters for maximum conversion efficiency in the piston-band region usually causes a reduction of output in the range of bass resonance. In most cases, if the parameters are adjusted for uniform response from

Table I. Predesign specifications for the loudspeaker system.

1. Application	Stereo home music listening
2. Environment	Room volumes of 700 ft³ to 3000 ft³ Acoustics: Medium dead to fairly live
3. Approximate size and finish	Bookshelf type. Walnut finish. 10 in. maximum depth
4. Target cost	(Specified)
5. Acoustical output level	103 dB minimum (0 dB = .0002 dynes/cm²) at 18 in. on system axis
6. Available input power	8 W rms (may be used with amplifiers of 6 to 60 W)
7. Bandwidth of response	50 Hz to 20 kHz ± 6 dB
8. Distortion	Less than 10% harmonic distortion and noise at system resonance. Less than 1% above 100 Hz
9. Impedance	Nominal 8 ohm (not less than 5 ohm)
10. Controls	None. Preset balance of acoustic output for environments listed above
11. Features	(Specified)
12. Competitive models	(Specified)

* Presented April 28, 1969 at the 36th Convention of the Audio Engineering Society, Los Angeles.

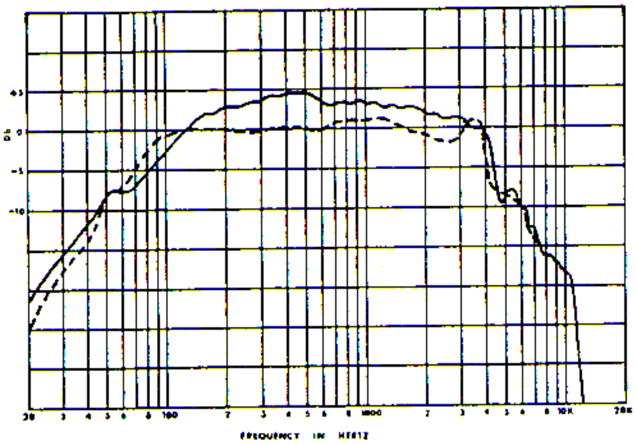


Fig. 1. Effect of flux density upon acoustic output. **Solid curve**: Acoustic output of a 6 in. woofer with a flux density of 5430 Gauss and a loaded Q of .42. **Dashed curve**: Acoustic output of an identical 6 in. woofer with a flux density of 4070 and a loaded Q of 1.02. Flux density measured over a .450 in. search distance microphone 18 in. on speaker axis. Equal power input at 400 Hz.

the bass transducer upper cutoff frequency down to the bass resonance point, efficiency will suffer.

Sound pressure is given [1] by

$$|p| \cong 10 f \rho_0 |U_c|/2r \qquad (1)$$

where $|p|$ = acoustic pressure in dynes/cm^2, r = distance in meters from the source in a free field, f = frequency in Hz, ρ_0 = density of air in kg/m^3, and $|U_c| = |u_c|/A_d$ = magnitude of the rms volume velocity of the active diaphragm area in m^3/sec.

Equation (1) is valid in the lower frequency range where the woofer diaphragm circumference is still less than $\lambda/2$ and the radiation is relatively nondirectional. It can be seen that the acoustic pressure is approximately proportional to the volume velocity. Once the effective piston area A_d has been chosen, the radiated acoustic pressure will become a function of $|u_c|$, the voice coil velocity:

$$u_c \cong e_g BL/(R_g+R_{vc})(R_m+jX_m) \qquad (2)$$

where u_c = voice coil velocity, e_g = *generator or* amplifier voltage, B = flux density in the gap, L = length of voice coil conductor, R_g = internal generator resistance, R_{vc} = voice coil resistance, R_m = total mechanical resistance of loudspeaker system, and X_m = total mechanical reactance of loudspeaker system.

Equation (2) shows that the voice coil velocity is approximately proportional to the flux density and inversely proportional to the mechanical resistance and reactance. The mechanical reactance in the lower frequency piston band under consideration is due mainly to the function

$$X_m \cong \omega M_{md} \qquad (3)$$

where $\omega = 2\pi F$ and M_{md} = mass of the diaphragm.

Since X_m is in the denominator of Eq. (2), the voice coil velocity will be inversely proportional to the mass of the diaphragm. Thus, it can be shown that an increase in flux density or a reduction in mass reactance will yield an increase in efficiency in the mass controlled range.

However, since

$$R_m = [B^2L^2/(R_g+R_{vc})]+R_{ms}+R_{mr} \qquad (4)$$

where R_m = total mechanical resistance of the loudspeaker in mechanical ohms, R_{ms} = mechanical resistance of the loudspeaker suspension in mechanical ohms, and R_{mr} = radiation resistance in mechanical ohms, it can be seen that R_m will increase with the square of the flux density, while from Eq. (3) the voice coil velocity will increase only to the first power of the flux density. When $R_m^2 >> X_m^2$, the acoustic pressure will decrease with decreasing frequency [2]. This rate will approach 6 dB/octave. Figure 1 shows graphically the effect of flux density upon the acoustic output level and the effect of Q upon the amplitude response *vs* bandwidth shape. The loudspeakers are identical in all respects with the exception of their magnet structures. The flux density of Loudspeaker B represents an increase of 33% over that of Loudspeaker A. The difference in acoustic output in the piston band can be predicted [3] by:

$$\text{dB} = 20\log_{10}(Bg_1/Bg_2) \qquad (5)$$

and should be approximately 2.4 dB. The increase in output is 2.4 dB at 200 Hz but increases above that value to about 3.5 to 4.0 dB at 500 Hz. The increase in output at 500 Hz over the predicted amount is due to increased radial-mode excitation of the cone. The output begins to decrease at a 6 dB/octave rate below 150 Hz. The slight rise at 50 Hz, which is apparent in both curves, is due to a standing wave mode in the anechoic chamber.

From the foregoing discussion it can be concluded that maximum conversion efficiency and flat response down to bass resonance frequency seem to be incompatible.

Two approaches may be taken in order to solve the problem. The first, which will yield maximum efficiency in the upper piston band, is to increase the flux density to a maximum and reduce the mass to a minimum. When the designer of the loudspeaker system can exercise control over the associated amplifier design and it is known that the loudspeaker system will be used only with a particular amplifier, two methods may be used to compensate for this first approach where the acoustic output will be decreasing at approximately 6 dB/octave with decreasing frequency. The frequency response of an amplifier with a low internal impedance (high damping factor) may be equalized electronically to give the inverse of the loudspeaker response. There is another problem which must be considered in this connection. Referring again to Fig. 1, it will be noted that the output of Loudspeaker B rises over that of Loudspeaker A below 50 Hz. This is because Loudspeaker B is accepting proportionately more power below resonance, which is the same for both systems, than is Loudspeaker A. Part of this increased output consists of increased distortion products caused by nonlinearity in the suspension due to the greater excursions. It may be surprising to discover that increased efficiency means less power handling capability, but this is nonetheless true. This is a particular disadvantage at low frequencies where a slope of 12 dB/octave would be desirable to limit cone excursions and thereby reduce distortion. Electronic boost down to system resonance followed by an abrupt rolloff is possible, but would seem to be a complication which might be avoided. Another method is to adjust the damping factor of the amplifier to a low value. This technique is not as acceptable because of the possible adverse effect upon transient response due to the higher internal resistance of the amplifier.

The second approach consists of adjusting the flux density and the mass of the loudspeaker system to yield flat response through the mass-controlled piston band down to the bass resonant frequency. The value of Q for such a response is approximately unity. Since the transient response is a function of the Q of the system, it should be determined if a Q of 1 will have a deleterious effect upon the transient response. A criterion has been suggested based upon psychological studies which indicate that for satisfactory transient performance [4]

$$R_m/2M_m > 92 \text{ sec}^{-1} \qquad (6)$$

where R_m = mechanical resistance and M_m = the mechanical mass.

Since for the acoustical system, $R_m/M_m = R_a/M_a$, the Q_T or Q for satisfactory transient performance would be [5]

$$Q_T = \omega_0 M_a/R_a < \omega_0/184 \qquad (7)$$

where M_a = acoustical mass and R_a = acoustical resistance.

Equation (7) shows that the value of Q_T is a function of frequency. If this value of Q is used as a guide, a graph can be drawn showing the maximum values for good transient response. Figure 2 shows the values of Q_T as a function of frequency. For frequencies above 30 Hz the value of Q_T is greater than unity. If the loudspeaker system resonance is 70 Hz, as called for in the specifications, the value of Q_T should be less than 2.4. Therefore, Q of one should be satisfactory. A loudspeaker system designed for flat acoustic output down to resonance has the advantage of being able to be used with any modern high-damping-factor amplifier.

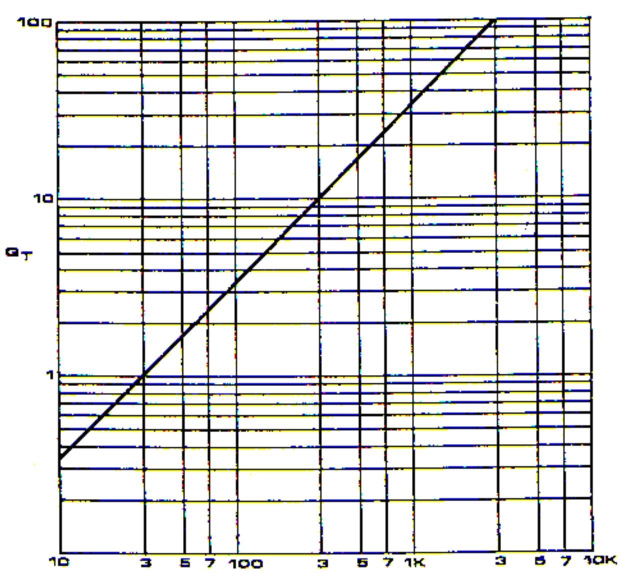
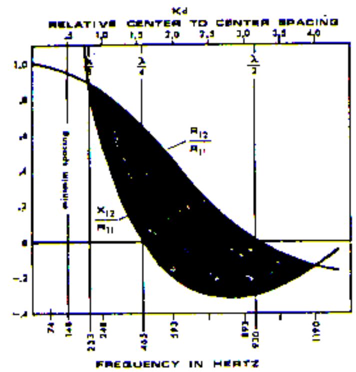


Fig. 2. Value of Q_T as a function of frequency. (After Beranek)



$Kd = \frac{2\pi d}{\lambda}$ $\quad f = \frac{13500 \text{ in/sec}}{\lambda \text{ in inches}}$

CENTER TO CENTER SPACING $d = 7.25''$

Kd	λ/d	f (Hertz)	λ (inches)
4.0		1190	11.5
3.2	2	930	14.5
3.0		890	16.5
2.0		593	23.0
1.6	4	465	29.0
1.0		295	46.0
.8	8	233	58.0
.5		148	92.0
.25		74	184.0

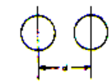


Fig. 3. Relationship between radiation resistance and reactance as a function of piston spacing. Values of frequency and λ for two 6 in. woofers. d = 7.25 in.

At this point, a discussion of a fault which most small bookshelf loudspeakers seem to exhibit is appropriate as an introduction to the next section. This fault is in the apparent inability of such loudspeaker systems to reproduce, with realism, the "impact" sounds of percussion instruments. This seems to be due primarily to a lack of proper balance in the acoustic output. If the output in the range from about 200 Hz to 600 Hz is raised slightly with respect to the output at system resonance, these impact sounds are more realistically reproduced. This can be verified through the use of a graphic equalizer while auditioning this type of loudspeaker system. The shape of the bandwidth-response characteristic obtained by such means is different from that obtained by simply increasing the flux density of a woofer and thereby reducing the system Q at resonance. What is needed is an acoustical method of increasing output equally in the range from 200 Hz to 600 Hz while maintaining the same output at system resonance. For this purpose the effects of mutual coupling were investigated.

MUTUAL COUPLING

Theoretical Considerations

Mutual coupling is the effect one piston transducer has upon another when they are closely spaced with respect to the wavelength they are radiating. There are excellent articles dealing with the theoretical aspects of the interaction, or mutual acoustic impedance, between pistons mounted in an infinite plane [6,7]. Mutual coupling is also dealt with to a lesser extent by other authors [8,9]. Figure 3 is a graphic presentation of the effects of the mutual interaction of two 6 in. loudspeakers. The mutual impedance is separated into its resistive and reactive components, R_{12}/R_{11} and X_{12}/R_{11} respectively. The data was derived from Ref. 6. The zero reference line represents the value of radiation impedance that would be seen by a single loudspeaker of the same size. The radiation resistance, which is the real or useful component of the radiation impedance, increases below $\lambda/2$ until it reaches a value which is twice that for a single loudspeaker. The radiation reactance remains below that of a single loudspeaker until the point where the spacing between the two loudspeakers is ¼ of the wavelength

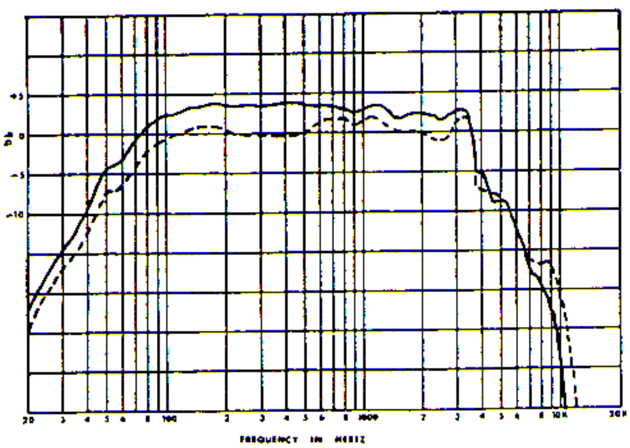


Fig. 4. Increase in acoustic output due to mutual coupling between two 6 in. woofers. **Solid curve**: Two 6 in. woofers in separate identical 650 in.^{3} enclosures; center-to-center spacing, = 9.5 in. **Dashed curve**: Single 6 in. woofer in 650 in.^{3} enclosure. Microphone 18 in. on axis of single woofer or between the two woofers. Equal power input at 400 Hz.

being radiated. At $\lambda/8$ the radiation reactance increases rapidly. The shaded area between the curves of the two functions indicates the wavelengths for which the acoustic efficiency of two loudspeakers increases over that of a single loudspeaker. The acoustic radiation through the range of maximum intercoupling between the two loudspeakers is twice that of a single loudspeaker, as is to be expected since the output will double with each doubling of diaphragm area. However, for wavelengths slightly above $\lambda/2$ the output decreases to that of a single loudspeaker, and below $\lambda/8$ the output actually decreases below that of a single loudspeaker. This latter effect is due to the fact that, as can be seen in Fig. 3, the radiation reactance increases rapidly below $\lambda/8$ and consequently the mass loading decreases the efficiency. Of course, this mass loading can be useful in limiting the excursion of the loudspeakers and therefore reducing distortion in the low-frequency range below system resonance.

Practical Considerations

The values of frequency and wavelength shown in Fig. 3 are for two 6 in. loudspeakers with an effective piston radius of 2.25 in. and a center-to-center spacing of 7.25 in. The graph indicates that the maximum increase in acoustic output over a single loudspeaker will occur at about 465 Hz, with a general increase occurring from a little over 200 Hz up to about 1000 Hz. Above and below these frequencies the output will decrease to about that of a single loudspeaker. Figure 4 shows the frequency response curves of two 6 in. woofers and a single 6 in. woofer. Each woofer was mounted in a 650 in.^{3} enclosure. The center-to-center loudspeaker spacing for these curves was 9.5 in. The function

$$K_d = 2\pi d/\lambda \qquad (8)$$

where K_d = relative separation and d = center to center spacing, indicates that the frequencies shown in Figure 3 for a spacing d = 7.25 in. would all be shifted to the right slightly. The early experiments during the initial design were performed by using a small enclosure for each woofer. This was done in order to control the experiments more closely. The final loudspeaker system uses a center-to-center spacing for the woofers of 7.25 in. The spacing for the experimental work and the final system are still close enough to draw meaningful conclusions about the final system from the experimental data of Fig. 4. The range of maximum output does occur from about 200 Hz to about 1000 Hz where it drops to a value close to that of a single 6 in. woofer. Above about 600 Hz the 6 in. woofer radiation becmes directional with increasing frequency. Below about 100 Hz, the relationship between the longest dimension of the test box and the wavelength of the radiated sound is $\lambda/8$, and neither the radiation reactance or resistance behaves in an easily predictable manner [10]. The frequency of maximum gain does occur, according to Fig. 4, at approximately 450 Hz as predicted by the data of Fig. 3. Figure 5 shows the same two frequency response curves of Fig. 4 after visual normalization intended to indicate more clearly the range of maximum gain due to mutual coupling. The amplifier power input to both the single woofer and the dual woofers was carefully monitored to avoid any possible errors due to different input levels. Both the theoretical and experimental data show then that the acoustic method of increasing the output in the range of 200 Hz to 600 Hz has been found in the mutual coupling of two woofers mounted in close proximity. This increase in output is also relatively uniform across this range, as opposed to the positive slope increase which would be obtained by increasing the flux density. Another advantage is that the increase in output has been achieved without affecting the Q at resonance.

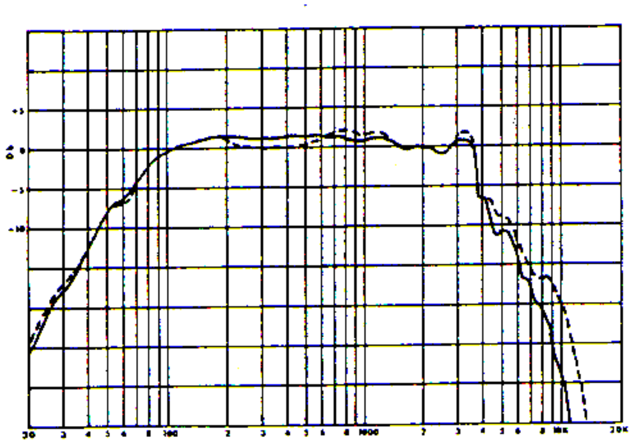


Fig. 5. Acoustic output curves of Fig. 4 with visual normalization to show the frequency range of maximum acoustic gain due to mutual coupling.

DESIGN PARAMETERS

Woofer Design

The nominal size of the loudspeakers chosen as woofers for the loudspeaker system under consideration is 6 in. diameter. The effective piston radius is 2.25 in. and the actual radiating area of the cone is 15.7 in^2. Since there is less difficulty in maintaining good acoustical output in the upper frequency range than if a large cone were used, a cone pulp can be chosen which has relatively high internal dissipation. This results in a much lower Q for the normal resonant modes of the cone

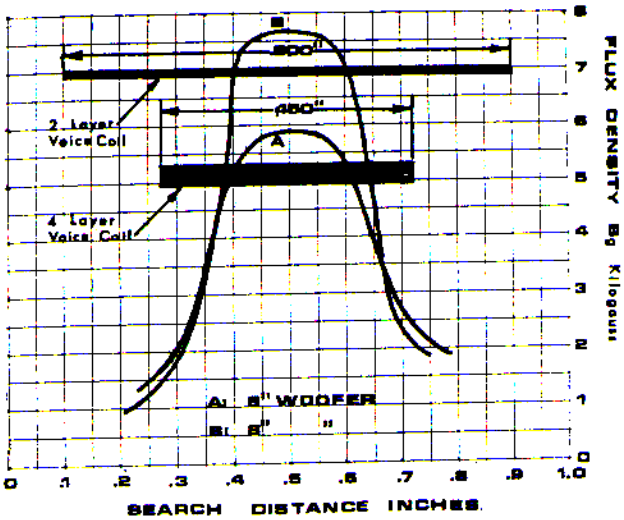


Fig. 6. Magnet flux density. **Curve A:** 6 in. woofer. **Curve B:** 8 in. woofer.

[11], and consequently, a smooth frequency response and good reproduction of transient sounds [12]. The cone pulp contains alpha cellulose fibers which have been hydropulped. Kapok, which consists of very small, light, hollow fibers, is mixed into the pulp and it is then made into a strong combed paper. This type of cone material combines good radiation properties and good dissipation of the modes which tend to build up on any loudspeaker cone. The geometry of the cross-section of the cone shows a gentle curvature from the apex up to the rim. This curvature also helps to dissipate internal energy [13]. The cone is relatively shallow; this tends to obviate the effects of radial modes of vibration, which can be quite severe in deep straight-sided cones [14]. The edge of the cone body has a gentle roll-over which terminates in the annulus.

The annulus is a reverse half-roll configuration. The width is ½ in. This represents 43% of the total active loudspeaker area, the effective piston making up the balance, or 57%. This ratio of annulus to piston is a disadvantage [15], but it is necessary to allow for the low-frequency cone excursions. A large annulus area can cause a large cancellation of acoustic output in the frequency range where it goes into self-resonance and is 180° out of phase with radiation from the cone [16].

The reverse half-roll, which is treated with a viscous fluid, has a very low Q due to internal friction losses and therefore the cancellation effects are minimized.

The magnet design and voice coil play a major roll in the performance of the loudspeaker. A "high-efficiency" type of magnet circuit was chosen because it allows for the long voice coil travel required [17]. This type of magnet structure allows operation of the magnet at about 65% efficiency, which is quite good. As shown by Eq. (2), the flux density has a direct effect upon the acoustic output. The length of the conductor also plays an important part. The force factor in gauss-centimeters times the current through the voice coil in amperes yields the force in dynes per square centimeters. This is the force which acts to drive the cone and results in acoustic radiation. The goal of a good design should be to achieve the maximum efficiency consistent with low distortion in the range of maximum excursion. The annulus and spider elements of the suspension should be the limiting factor in any good high compliance woofer design intended for music reproduction. For the 6 in. woofer, the excursion limit has been set at .200 in. peak-to-peak [18]. The suspension will remain relatively linear through this excursion range. This means that with a top plate thickness of .2391 in., which for cold-rolled low-carbon steel is No. 3 gauge, a voice coil a little over .450 in. long will be sufficient. Figure 6 shows the flux density for the final magnet structure. For reference, the voice coil winding length is shown. As a comparison, Fig. 6 also shows the flux density for an 8 in. woofer with a long voice coil. The shorter four-layer voice coil will produce relatively more force because proportionately more of its winding length is in a strong magnet field. Adjusting the voice coil length to be no longer than necessary for the maximum excursion will allow for economy of the magnet and magnet structure design. Figure 7 shows the frequency response of the 6 in. and 8 in. woofers with voice coils and flux densities shown in Fig. 6. The acoustic output in the range of resonance is within .5dB. The cost of the 8 in. woofer is approximately twice that of the single 6 in. woofer. This indicates that it will be possible, from an economic standpoint, to use two 6 in. woofers in a mutual coupling arrangement. The loaded Q for the 8 in. woofer is .89 and for the 6 in. woofer is 1.02 [19]. (The dynamic mass for the 8 in. woofer is twice that of the 6 in. woofer). It would appear, then, that using a four-layer voice coil of sufficient length for the excursion desired and adjusting the flux density to be no more than necessary is the most economical approach.

Another parameter of importance is the permeance coefficient of the magnet structure. If the gap length is opened up to allow for a four-layer voice coil, the length of the magnet must be sufficient to allow for proper operation. Curve A of Fig. 8 shows graphically the minimum permeance coefficient for optimum operation of an Alnico V magnet of the type used in the 6 in. and 8 in. loudspeakers. The formula for the permeance coefficient p is

$$p = L_m A_g \sigma / L_g A_m r_j = B/H \qquad (9)$$

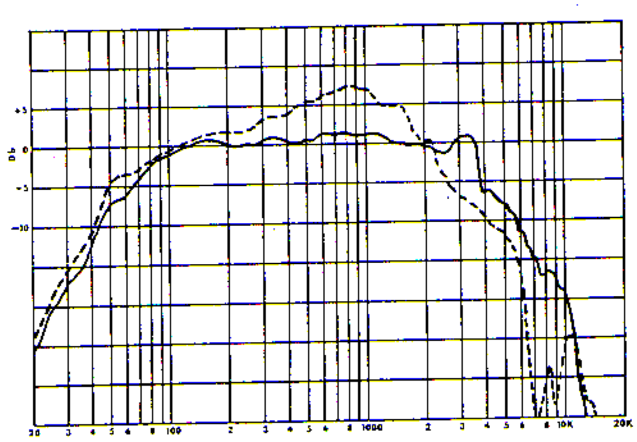


Fig. 7. Acoustic output of the woofers of Fig. 6. **Dashed curve:** 8 in. woofer. **Solid curve:** 6 in. woofer. Microphone 18 in. on-axis. Equal power input at 400 Hz.

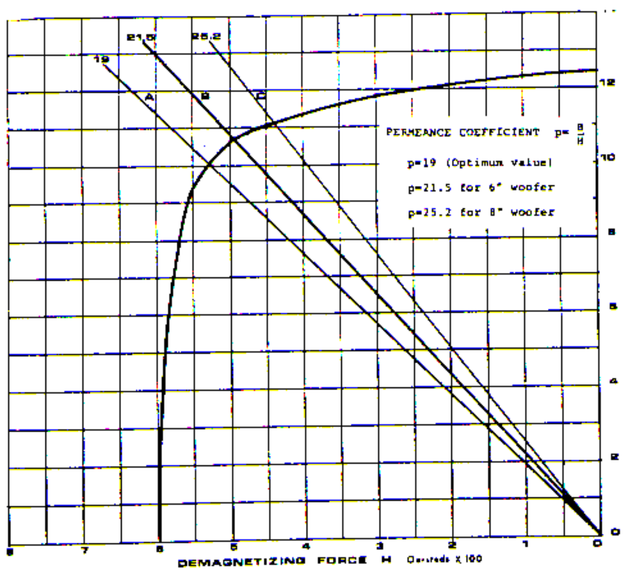


Fig. 8. Permeance coefficients. **Curve A:** Minimum value for optimum operation. **Curve B:** 6 in. woofer. **Curve C:** 8 in. woofer.

and represents the ratio of the total external permeance to the permeance of the space occupied by the magnet [20]. This value of permeance coefficient is optimum because this point on the curve also represents the peak energy product of the magnet ($B_d H_d$) maximum [21]. Curve B of Fig. 8 indicates that the value of the permeance coefficient could be reduced slightly. However, it has not been reduced because, first, some tolerance should be allowed for the manufacture of production magnets, and second and more importantly, the magnet must be protected from the demagnetizing force produced by the voice coil when it is energized by program material. The 6 in. woofer of Curve B will accept slightly over 45 W of sinewave power before a permanent change in acoustic level of -1 dB results. Since the final loudspeaker system uses two of these woofers in parallel and they will divide the power, the input power may reach 90 W before the loudspeakers will be demagnetized one dB. The 8 in. loudspeaker of Curve C is operating even further up the slope of the demagnetization curve. The operating point of 25.2 as well as the fact that there are a smaller number of voice coil turns in the gap of the magnet structure indicates overdesign with regard to the demagnetization problem.

Tweeter Design

Figure 9 shows a cutaway drawing of the tweeter. The nominal size of this tweeter is 3½ in. The cone housing is an open-back type. The large plastic cup chamber makes the tweeter a self-contained unit which could be operated without any further baffling. The cup chamber also tunes the natural resonance to approximately 750 Hz. The damping material in the cup absorbs standing-wave energy and smooths the response. The cone is a shallow curvilinear type. Directly behind the cone and in contact with it is a felt damping pad. This pad smooths the response by removing standing-wave energy from the cone. It also restricts the movement of the cone and thereby lowers the Q at resonance. The end result of the design is a drastic reduction in the "nasal" quality of the sound which seems to plague most small tweeters. The low resonance allows the tweeter to be operated with a crossover frequency of 1500 Hz.

The frequency response of the tweeter is shown in Fig. 10. The solid curve is for an unmounted tweeter. The response is ±2.5 dB from slightly below 700 Hz to above 17 kHz. The dashed curve shows the effect of mounting on the frequency response. The surface upon which the tweeter is mounted is 19 in. × 13 in. The center of the tweeter is 3½ in. from the edge of the enclosure. The reinforcements and cancellations due to the mounting are readily apparent.

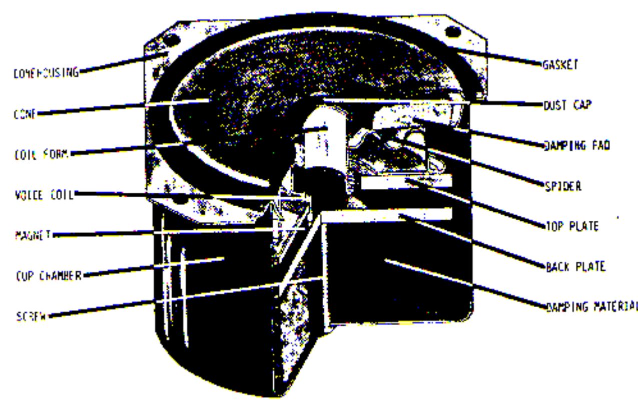


Fig. 9. Cutaway drawing of the tweeter.

Crossover Network

The design of the crossover network proceeded through three stages of development which considered the response of the network into a resistive load, the effect of the varying loudspeaker impedance upon the network, and the effect of the network upon the acoustic output of the loudspeakers. The last item is the most important, since the smooth blending of the acoustic output from the loudspeakers is the main function of the crossover network. Figure 11 shows the schematic di-

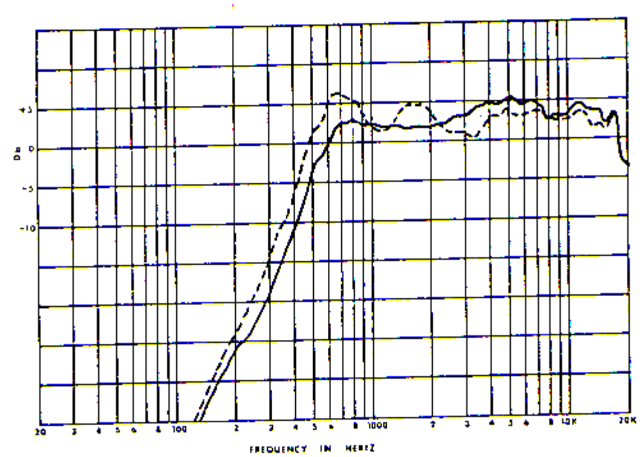
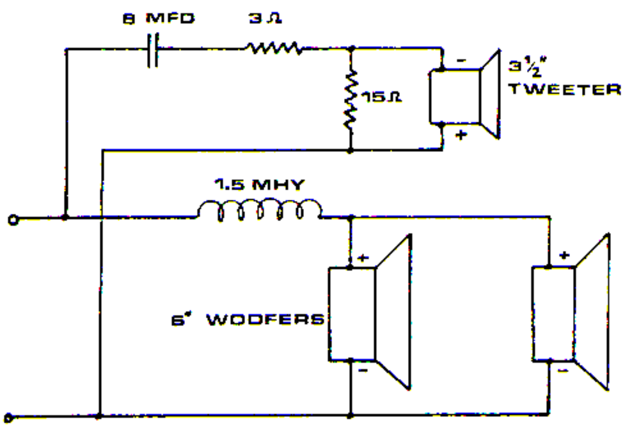
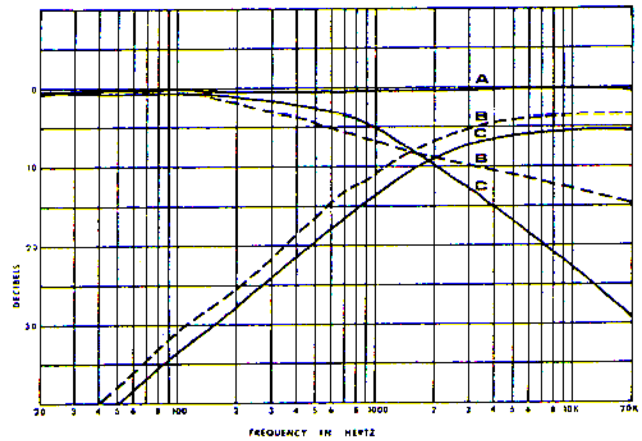


Fig. 10. Effect of tweeter mounting on acoustic output. **Solid curve:** Unmounted tweeter (no baffle). **Dashed curve:** Tweeter mounted in enclosure. Microphone 18 in. on-axis.



a.



b.

Fig. 11. a. Schematic diagram of the dual woofer loudspeaker system. b. **Curve A:** Voltage input to crossover network of Fig. 11a. **Curve B:** Voltage measured at woofer and tweeter terminals. **Curve C:** Voltage measured across 7.5 ohm resistive loads replacing speakers.

agram of the crossover network. The values shown for the capacitor and the inductor indicate that this is not strictly a crossover network in the classical sense. It would be more correctly classified as a low-pass high-pass filter. The curves of Fig. 11 show the first two effects mentioned above. The most dramatic effect is the change in the response of the low-pass section between the resistive load and the woofer impedance load.

The acoustical response of the woofers, the tweeter, and the complete system, are shown in Fig. 12. The woofer and tweeter are connected to the crossover network with their terminals 180° out of phase with respect to a dc voltage [22]. The constant-resistance network will cause the voltages appearing across the woofer and tweeter terminals to be 90° out of phase. A 90° phase shift also occurs between the woofer input voltage and its acoustic output in the upper range of its response [23].

Thus, the acoustic output is in phase in the crossover region when the loudspeaker terminals are out of phase to a dc voltage. The effect of phasing upon the acoustic output is shown in Fig. 13. The solid curve shows the acoustic response when the woofer and tweeter are properly phased as described above. The dashed curve shows the acoustic response when the woofer and tweeter are connected so that they are in phase with respect to a dc voltage across their respective terminals, but out of phase acoustically at crossover. These curves were made 18 in. on the axis of the loudspeaker system which is defined, for this system, as a point in the middle of a triangle formed by the two woofers and the tweeter. Curves made at various points off-axis indicate that a smooth blending has also been achieved for different angles off-axis.

Figure 14 shows the effect on the acoustic response of the tweeter of two different methods of reducing the acoustic output level. The solid curve was made by adjusting the value of a resistor in series with the tweeter so that the acoustic output level above 10 kHz was the same as that produced when a proper L-pad was used. The dashed curve is the acoustic response of the tweeter with the L-pad. The improper loading of the network due to the series resistor causes the acoustic output of the tweeter to be greater in its low-frequency range. The damping of the tweeter resonance is also seriously impaired. The use of a proper L-pad not only provides the correct load impedance for the high-pass section of the

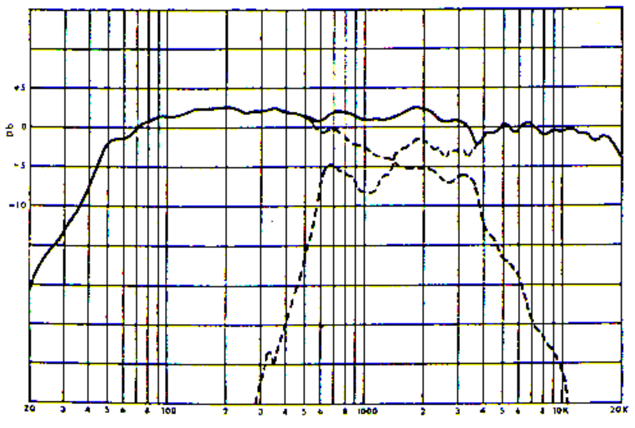


Fig. 12. Acoustic output of woofers, tweeter and complete speaker system. **Solid curve**: Complete system. **Dashed curves**: Response of woofers through low-pass section of crossover network and response of tweeter through high-pass section. Microphone 18 in. on system axis.

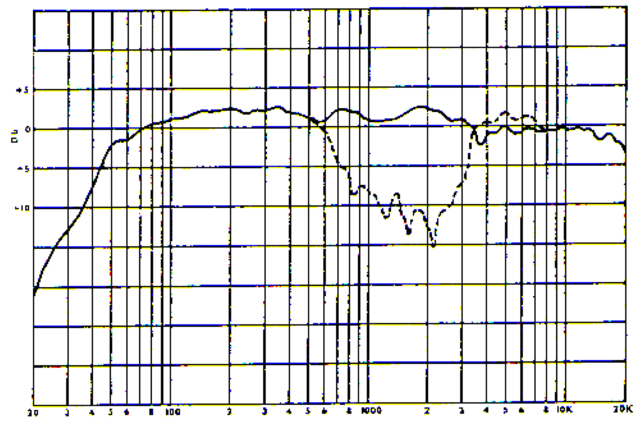


Fig. 13. Effect of woofer-tweeter phasing on acoustic output. **Solid curve**: Acoustically in phase. **Dashed curve**: Acoustically out of phase. Microphone 18 in. on-axis.

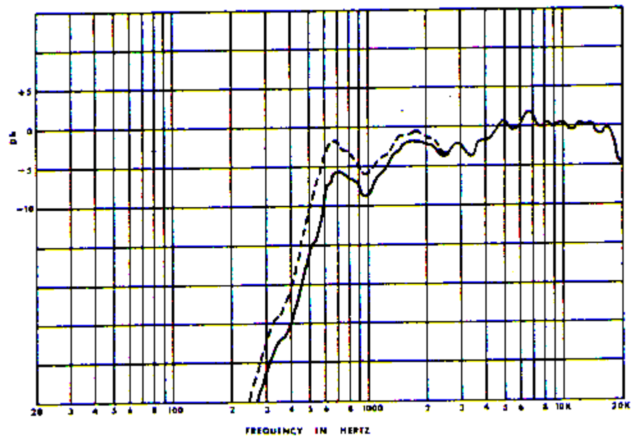


Fig. 14. Acoustic response different methods of reducing acoustic output level. **Solid curve**: Use of L-pad. **Dashed curve**: Use of series resistance. Microphone 18 in. on-axis.

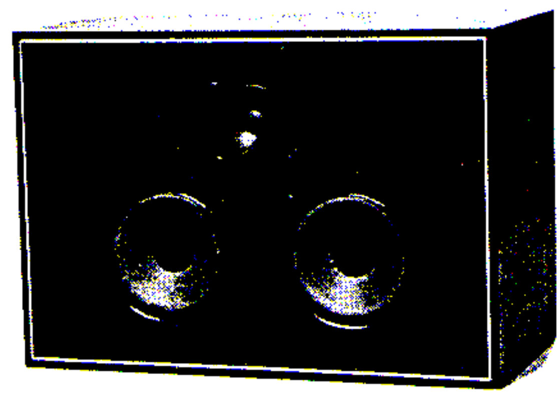


Fig. 15. The complete dual-woofer speaker system, showing relative location of woofers and tweeter.

crossover network but also maintains the damping of the tweeter resonance in the region where the series reactance of the crossover capacitor is increasing.

Why a highly efficient tweeter should have been designed and then its acoustic output level reduced by means of a pad may at first seem obscure. Two factors are responsible for such an approach. The power-handling capability of the tweeter, which uses a small voice coil to achieve good high-frequency response, is much less than that of the woofers. Studies made by George Brettell of the energy distribution of various program material indicate that the high-frequency energy density is much higher than previously thought [24]. The high-frequency response capabilities of program sources seems to be improving also. By using a highly efficient tweeter, less input power is required to produce the acoustic output level necessary to match the level of the woofers. Therefore, the input power to the tweeter may be reduced. This approach allows operation of the loudspeaker system from relatively high-power amplifiers with a reduced danger of destroying the tweeter.

PERFORMANCE

The dual-woofer system is shown in Fig. 15. Figure 16 shows its acoustic response compared to that of a system using the 8 in. woofer referred to previously. The power input of the two systems was adjusted to be equal at 400 Hz. The impedance curves for each system are also shown for reference. The power input requirements of the two systems of Fig. 16 are shown in Fig. 17. The dual-woofer speaker system's impedance causes it to accept more power in the range from about 80 Hz to 400 Hz with a maximum of approximately 1 dB more at 150 Hz. Across the greater portion of the range, the dual-woofer loudspeaker system requires less input power to produce equal or greater acoustic output than the 8 in. woofer loudspeaker system. In the high-frequency range, the tweeter is capable of producing equal acoustic output while requiring approximately 3.8 dB less input power. This affords extra protection for the tweeter of the dual-woofer loudspeaker system.

Another criterion of performance is acoustic output

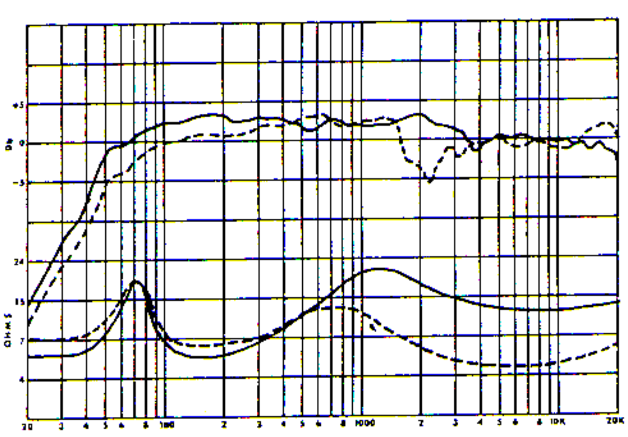


Fig. 16. Comparison of the acoustic response of the dual 6 in. woofer system and a speaker system using a single 8 in. woofer. **Solid curve**: Dual-woofer system. **Dashed curve**: 8 in. woofer speaker system microphone 18 in. on system axis. Equal power input power at 400 Hz.

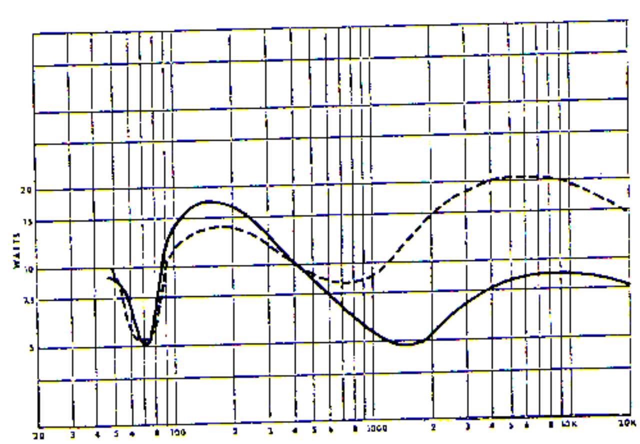


Fig. 17. Power input curves for the speaker systems of Fig. 16 for the acoustic output shown. **Solid curve**: Dual-woofer speaker system. **Dashed curve**: 8 in. woofer speaker system.

level *vs* distortion. It would be of little use to design a loudspeaker that will produce high acoustic levels if the distortion is also high. Distortion measurements were made at various acoustic output levels for both the dual-woofer loudspeaker system and the system which uses the 8 in. woofer. The relative merits of an extra-long two-layer voice coil *vs* a shorter four-layer voice coil were discussed earlier. The criterion was suggested that the suspension should be the limiting element in the design with respect to distortion, and that the voice coil should be no longer than necessary. In order to determine whether this criterion is valid, the distortion components of the dual-woofer loudspeaker system and the system which uses the 8 in. woofer were measured at system resonance which was 70 Hz for both, and at an acoustic output level of 105 dB at 18 in. The results are shown in Table II.

Table II. Input power and distortion at system resonance.

	Input Power	Distortion: Second Harmonic	Distortion: Third Harmonic
Dual-woofer system	5.3 W	13.3%	11.2%
Single 8 in. woofer system	5.6 W	17.8%	9.0%

The above data would seem to validate the criteria since, for the dual-woofer system, the third harmonic distortion, which is due mainly to symmetrical nonlinearity of the voice coil, is almost the same as the second harmonic distortion component, which is mainly produced by the suspension [25]. The extra-long voice coil of the 8 in. system does reduce the third harmonic distortion, but the second harmonic distortion is almost twice as great as the third for this system.

CONCLUSION

It appears that the effects of mutual coupling can be used to advantage in shaping the acoustic output response of a loudspeaker system. The parameters of the individual loudspeakers and the crossover network can be adjusted to take advantage of the effects of mutual coupling. The impedance characteristics of a loudspeaker can be adjusted to advantage with respect to the input power in order to achieve a desired acoustic output.

ACKNOWLEDGEMENT

The author wishes to thank Donald S. Schroeder for assistance in preparation of graphics for this paper and Robert J. Fanella for his encouragement.

NOTES

1. Leo L. Beranek, *Acoustics* (McGraw Hill Book Co., New York, 1954), p. 222, Eq. 8.16. In this equation 'p' has been converted to dynes/cm² by multiplying the numerator by a factor of 10.
2. *Ibid.*, p. 192.
3. Rollin J. Parker and Robert J. Studders, *Permanent Magnets and Their Application* (John Wiley, New York, 1962), p. 255.
4. Beranek, *op. cit.*, p. 206.
5. *Ibid.*, pp. 225-226.
6. R. L. Pritchard, "Mutual Acoustic Impedance Between Radiators in an Infinite Rigid Plane", *J. Acoust. Soc. Am.* **32**, 730 (1960).
7. S. J. Klapman, "Interaction Impedance of a System of Circular Pistons", *J. Acoust. Soc. Am.* **11**, 289 (1939).
8. Daniel J. Plach and Philip B. Williams, "Loudspeaker Enclosures", *Audio Engineering*, 12, (July 1951).
9. Hugh S. Knowles, "Loudspeakers and Room Acoustics", in Keith Henney (ed.), *Radio Engineering Handbook* (McGraw-Hill Book Co., New York, 1959), 5th ed., Ch. 11, pp. 11-14.
10. Beranek, *op. cit.*, p. 216.
11. Murlan S. Corrington, "Transient Testing of Loudspeakers", *Audio Engineering*, 9 (Aug. 1950).
12. R. J. Larson and A. J. Adducci, "Transient Distortion in Loudspeakers", *IRE Trans. on Audio* **AU-9**, 79 (1961).
13. J. Q. Tiedje, "Speaker Design", *Radio Engineering*, 11 (Jan. 1936).
14. Charles L. McShane, "HiFi Loudspeaker Cones", *Electronics World*, 38 (Feb. 1963).
15. Murlan S. Corrington and Marshall C. Kidd, "Amplitude and Phase Measurements on Loudspeaker Cones", *Proc. I.R.E.* **39**, 1021 (1951).
16. Harry F. Olson, *Acoustical Engineering* (D. Van Nostrand Co., New York, 1957), p. 194.
17. Rollin J. Parker, "Permanent Magnets in Audio Devices", *IRE Trans. on Component Parts* **CP-5**, 32 (1958).
18. Frank Massa, *Acoustic Design Charts*, (Blakiston Co., Philadelphia, 1942), p. 129, Chart No. 62. The excursion limit was determined for the desired output (103.5 dB) at the final system resonance of 70 Hz.
19. See Appendix for the method used to determine Q.
20. Indiana General Corp., *Design and Application of Permanent Magnets*, Permanent Magnet Manual 6A, p. 20.
21. *Ibid.*, p. 10.
22. "In phase to dc" means that if the negative terminal of a dc voltage source is connected to the common terminals of the woofer and tweeter and the positive terminal of the source is connected to the other terminal of both the woofer and tweeter, their cones will move in the same direction.
23. Corrington and Kidd, *op. cit.*, p. 1023.
24. John G. McKnight, "The Distribution of Peak Energy in Recorded Music, and Its Relation to Magnetic Recording Systems", *J. Audio Eng. Soc.* **7**, 65 (1959).
25. The distortion produced by the suspension is primarily second harmonic, apparently due to the fact that the half-roll annulus motion is nonsymmetrical.

APPENDIX

Determination of Loudspeaker Q

Figure A1. shows the measuring setup for the determination of loudspeaker Q. The loudspeaker is represented by its simplified equivalent circuit. Unloaded Q is determined first. R_g is made very large with respect to Z_m. In most cases, a 1000 ohm resistor will be sufficient. The audio generator is tuned to the loudspeaker resonant frequency f_0; f_1 and f_2 are the frequencies below and above resonance, respectively, where the voltage is

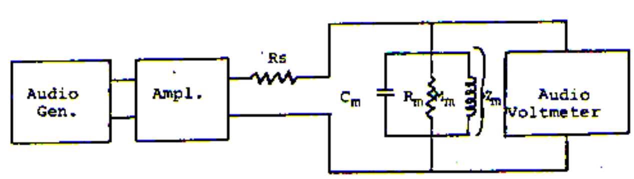


Fig. A1. Setup for measuring loudspeaker Q.

.707% of the voltage measured at f_0. At these two points the voltage will also be 45° out of phase with the voltage at f_0.[1] The unloaded Q of the loudspeaker is[2]

$$Q_1 = f_0/\Delta f \qquad (1)$$

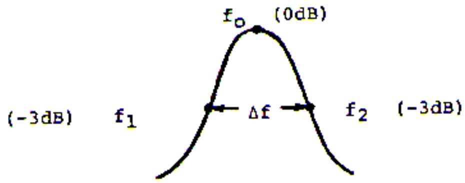


Fig. A2. Values of frequency for the determination of loudspeaker Q.

where $\Delta f = f_2 \text{-} f_1$ (see Fig. A2). The loaded Q or the value of Q when the loudspeaker is connected to an amplifier is given by[3]

$$Q_2 = Q_1[(R_{vc}+R_g)/(Z_m+R_g)] \qquad (2)$$

where R_{vc} = voice coil resistance, R_g = internal generator resistance, and Z_m = total loudspeaker impedance at f_0.

The value of R_g may be determined by the followimg method: With the amplifier output unloaded, set the output voltage to 1 V (or any convenient small voltage). Place a variable load resistance across the output terminals of the amplifier. Reduce the load resistance until the voltage is ½ the original value. The value of the load resistance R_L then equals the internal amplifier resistance R_g.

The damping factor of the amplifier equals the ratio of the load impedance Z_L to the internal amplifier impedance Z_g. At resonance the impedance of the loudspeaker is resistive ($Z_L = R_L$). In a well designed amplifier within the passband there will be little or no phase shift, and there the internal impedance is almost purely resistive ($Z_g \cong R_g$). The damping factor is therefore

$$DF = R_L/R_g. \qquad (3)$$

The regulation in dB can be read directly on most audio voltmeters by noting the reading in dB with no load and the reading in dB with load resistance R_L connected across the output terminals. It may be calculated from

$$dB\ regulation = 20 \log_{10}[(R_g+R_L)/R_L]. \qquad (4)$$

1 Frederick E. Terman, *Radio Engineering*, 3rd Edition: New York: McGraw-Hill Book Co., Inc., 1947, p. 44.
Terman uses $\Delta f = 1/2Q$ indicating that $\Delta f = f_0 - f_1$ or $f_2 - f_0$. This would make the formula for **Q**, $Q = f_0/2\Delta f$. The Radiotron Designer's Handbook, p. 841, also uses this form.

2 Leo L. Beranek, *Acoustics*, 1st Edition; New York: McGraw-Hill Book Co., Inc., 1954. p 229.

3 F. Langford-Smith, *Radiotron Designer's Handbook*, 4th Edition; Harrison, N. J.: Radio Corporation of America, 1952, p. 841.

THE AUTHOR

Edward M. Long was born in 1932 in Canandaigua, New York. During the Korean War, he taught pulse techniques, multiplexing, and FM theory at Fort Monmouth Signal School. He completed his engineering studies at Fisher College in 1957.

Mr. Long became a project engineer at Sylvania Home Electronics, and did design work on loudspeaker systems there and at Audio Dynamics Corporation. He later became senior acoustics engineer at C.T.S. of Paducah, Inc., where he assisted in the manufacture of loudspeaker systems. In 1968, he joined Ampex Corporation as senior acoustics engineer.

Mr. Long is an associate member of the Acoustical Society of America, a member of the Audio Engineering Society, and secretary of the Chicago Acoustical and Audio Group.