HOW TO TEST YOUR SPEAKERS

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Testing the Properties of Loudspeakers

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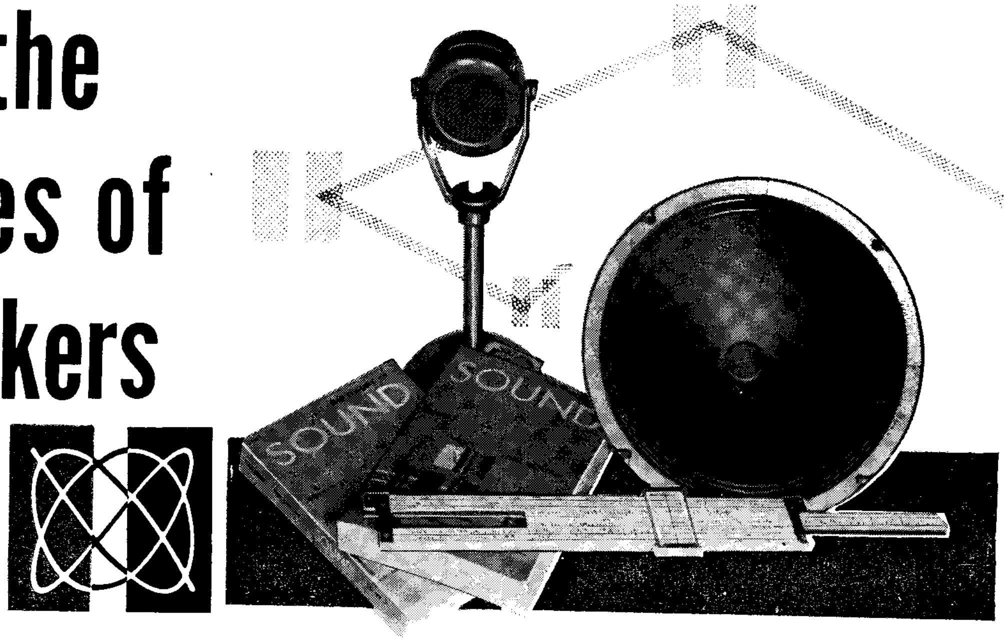
Collins Radio Co.



Simple tests and measurements require little more equipment than a.c. voltmeter and audio oscillator.

WHAT are the properties of loudspeakers which influence their ability to reproduce faithfully and efficiently? Mass, compliance, flux density, acoustic output are terms which enter the conversation when this question is answered. But how can these properties be measured? Precise and accurate measurements on loudspeakers are difficult and expensive, requiring skill and specialized equipment. We can be less rigorous, however, and make simple measurements which will clearly demonstrate many of these interesting properties of speakers. Some of these tests have been collected and are presented here. These tests have been simplified as much as possible and require only equipment which is available to the average technician and experimenter. These tests will be directed toward the acoustical portion of the reproducing system.

In making these tests remember that a loudspeaker is a carefully made unit. Do not tamper with the mechanism of an expensive unit. If any alterations are to be attempted, be sure to try your ideas first on an inexpensive unit. The tests described in this article, however, will not injure your speaker if reasonable care is exercised.

For the purpose of this article the acoustical portion of an audio system will be that apparatus between the secondary of the output transformer and the listener's ear. We will be concerned with the loudspeaker proper, its baffle, and the characteristics of the listening room.

A loudspeaker is an electro-acoustic device, so we cannot entirely divorce ourselves from the electrical portion of the audio system. As a matter of fact, we will make as many tests and measurements as possible with electrical equipment because it is more easily handled.

Speaker impedance vs frequency measurements: The accepted method of measuring impedance is, of course, with an impedance bridge which will indicate the resistance and reactance of the loudspeaker. In the absence of such a bridge, however, the magnitude of the impedance can be determined very simply by the voltage comparison method. The schematic of the test method is shown in Fig. 1. The maximum value of R is not critical as long as it is about ten times the nominal impedance of the loudspeaker. It is convenient to place a knob and scale on R and calibrate the scale in ohms with an ohmmeter. An audio tone of 400 cps from a test record or audio oscillator is then fed into the amplifier and the gain adjusted until a convenient deflection is obtained on the voltmeter (about three-quarters scale) with switch S in position 1. S is then placed in position 2 and R adjusted until the voltmeter reads the same in positions 1 and 2. The value of R will then be the magnitude of the speaker impedance at 400 cps. A typical plot of impedance *vs* frequency of a free speaker is shown in Fig. 8.

If it is desired to determine the resistive and reactive components of the loudspeaker impedance when no bridge is available, the method just described may be employed by first tuning out the reactive component. See Fig. 2 for connections.

Above the resonant frequency, the speaker impedance is composed of resistance and inductance. The effect of the reactance can be cancelled by adding, in series with the speaker, a reactance of opposite sign, *i.e.,* above cone resonance add series capacity. To determine the impedance, adjust R for maximum resistance, then vary C for a minimum reading of the voltmeter with S in position 1. Once this minimum has been obtained, proceed as previously outlined to adjust for equal voltages across the speaker and the resistance. The impedance of the speaker is then the value of resistance indicated by R plus a reactance equal to the reactance of the capacitor but with the sign reversed, that is, it is inductive. For example, if at 100 cps a capacity of 50 μfd. is necessary for the minimum reading with S in position 1 and equal voltages appear across the resistance and loudspeaker with an R of 5 ohms, the impedance of the speaker is found to be:

$$Z_s = R + jX_c = 5 + j32 \text{ ohms}$$

Notice that the sign of the reactance has been reversed. This measuring procedure can be repeated for different frequencies.

If only the relative impedance curve

COIN	WEIGHT—(grams)
Dime	2.50
Penny	3.05
Nickel	5.05
Quarter	6.40
Half-dollar	12.35
Silver dollar	26.70

Table 1. Weights of common U. S. coins.

Table 2. Sound absorption at 512 cps.

MATERIAL	COEFF.
Wood sheathing (varnished)	.03
Concrete	.016
Carpet	.25
Cork floor (waxed)	.05
Draperies	.35
Acoustic Celotex	.7
Upholstered chair	1.6
Adult person	4.2
Glass	.027
Sheetrock	.03
Open window	1.0
Couch	4.8

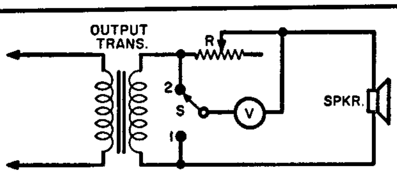


Fig. 1. Determining the impedance.

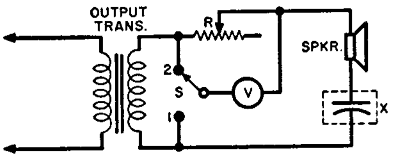
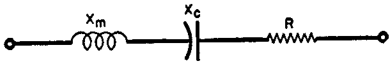
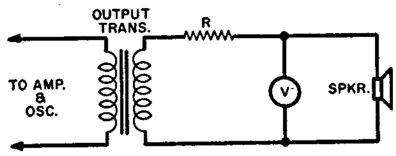


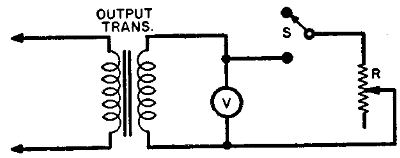
Fig. 2. Finding the resistive component.



← Fig. 3. Simplified equivalent circuit.



(A)



(B)

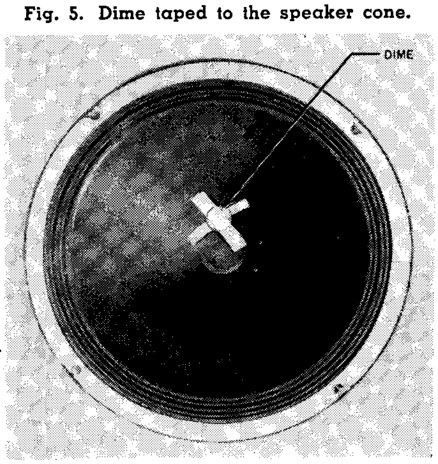
Fig. 4. (A) Determination of resonant frequency and (B) the source impedance.

is desired, it can be obtained by using the setup of Fig. 4A. Here R is non-critical as long as it is large compared to the nominal speaker impedance. As the frequency is varied throughout the desired range, the voltage measured on the voltmeter will be in proportion to the speaker impedance.

Source impedance: The magnitude of the impedance of the source which the speaker sees can be determined very easily by the half-voltage method. Electrical connections are shown in Fig. 4B. An audio tone is fed into the amplifier and the gain adjusted to give a convenient small voltage reading with switch S open. The switch is then closed and R is adjusted until the voltmeter reads half its original value. The value of R is then the magnitude of the source impedance. Source impedance is an important factor in loudspeaker damping.

Speaker resonance: Once an impedance curve has been run on a loudspeaker, the resonant frequency is quite obvious. Examination of Fig. 8 shows the resonant frequency to be that frequency at which the peak occurs, about 140 cps in this example. If it is necessary to know the resonant frequency only, to design a bass reflex enclosure for example, the scheme shown in Fig. 4A can be used very simply. The frequency of resonance is indicated by a peak in the voltage reading as the frequency is varied.

Fig. 5. Dime taped to the speaker cone.



Mass and compliance of speaker cone: In its most simple form, the loudspeaker can be represented as a series LCR circuit, as shown in Fig. 3. X_m, called the mass reactance, is determined by the mass of the cone. X_c, called the compliance reactance, is determined by the stiffness of the suspension system. R is the equivalent resistive component and is the result of the electrical losses and the radiation resistance of the loudspeaker. X_m and X_c are generally much larger than R so the point of resonance will occur when $X_m = X_c$ or:

$$2\pi f_r M = \frac{1}{2\pi f_r C}$$

This equation can be solved for f_r and yields the familiar equation:

$$f_r = \frac{1}{2\pi\sqrt{CM}} \quad \ldots\ldots (1)$$

where: f_r is the speaker resonant frequency in cps
M is the speaker cone mass in grams
C is the compliance of the suspension system in centimeters per dyne

Now f_r can be simply determined as has been described earlier. If either M or C is now altered, a new resonant frequency f_r' will be obtained. This will provide us with two equations in two unknowns, M and C, and these unknowns can be readily solved.

The compliance of the speaker cannot be altered readily but the mass of the vibrating cone may be changed by taping a small weight to the inner apex of the cone. See Fig. 5. Use only enough weight to give a significant change in resonant frequency. Too large a weight will make it impossible to detect resonance. Be sure to attach the extra mass securely with masking tape so that it does not rattle. After the extra mass, M', has been added, the new resonant frequency will be:

$$f_r' = \frac{1}{2\pi\sqrt{C(M+M')}} \quad \ldots\ldots (2)$$

Equations (1) and (2) can be solved simultaneously for C and M to yield:

$$C = \frac{\left(\frac{f_r}{f_r'}\right)^2 - 1}{4\pi^2 M' f_r^2} \text{ cm/dyne} \quad . (3)$$

and

$$M = \frac{M'}{\left(\frac{f_r}{f_r'}\right)^2 - 1} \text{ grams} \quad \ldots (4)$$

where: M' is the added mass in grams
f_r is the original resonant frequency
f_r' is the resonant frequency after the mass has been added

As an illustration, a certain 4" loudspeaker was found to have a natural resonance of 155 cps. A ½-gram weight, made up of a small crescent of #18 wire and the masking tape necessary to attach this weight to the cone, lowered the resonant frequency to 130 cps. When this information is inserted in equations (3) and (4) the mass of the cone is calculated to be 1.19 grams and the compliance 8.9×10^{-7} cm/dyne.

Flux density of the air gap: The efficiency, power handling capabilities, and general performance of a loudspeaker are related to the flux density, B, of the air gap times the length of wire, L, comprising the voice coil. In all but relatively few cases the term BL will appear rather than the quantity B alone. For that reason it will be sufficient to determine the product BL in our measurements and not be concerned with individual values. A simple way to make this measurement is to take advantage of the fact that the force exerted on a current-carrying conductor is:

$$F = BLI \quad \ldots\ldots (5)$$

where: F is the force in dynes
B is gap flux density in gausses
L is the length of wire on the voice coil in centimeters
I is the current flowing in the voice coil in abamperes (10 amps.)

We can measure this force with the arrangement shown in Figs. 6 and 10. The speaker is placed cone upward on a table or other flat surface. A battery, milliamp meter, and a variable resistor are connected in series with the voice coil so that a controlled current can be caused to flow in the voice coil. A thin cardboard disc is placed in the apex of the cone to provide a flat bottom. A small plumb-bob type weight is suspended on a string above the cone. With no current flowing in the voice coil, the plumb bob is adjusted to hang so that it just touches the cardboard disc at a point near the disc edge. A small known weight, such as a coin, is placed in the center of the cardboard disc. This will cause the cone to be depressed a certain distance depending on the weight of the coin. A depression of 1/32" or so will be sufficient. Current is now passed through the voice coil in such a direction as to raise the cone towards its original position. The current is adjusted so that once again the plumb bob just touches the edge of the cardboard disc. We have now balanced the $F = BLI$ equation. We know the mass of the added weight and can read the current flowing in the voice coil so the product of flux density times length becomes:

$$BL = 9.8 \times 10^6 \left(\frac{M}{I}\right) \text{ gauss-cm} \quad . (6)$$

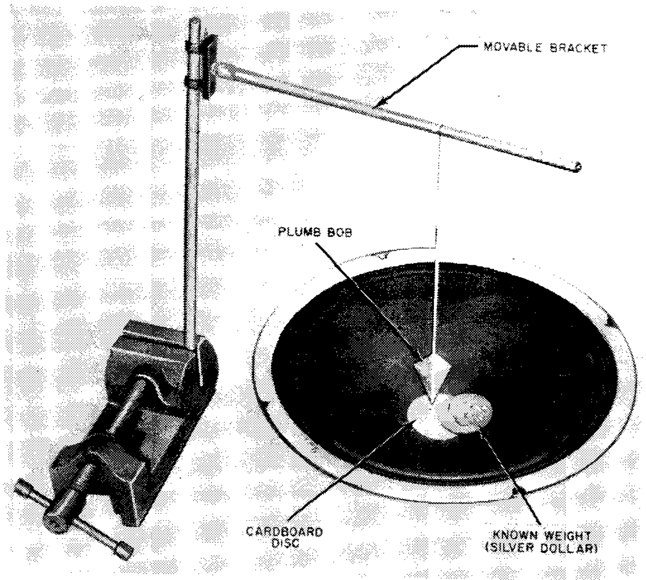


Fig. 6. Physical arrangement for determining flux density.

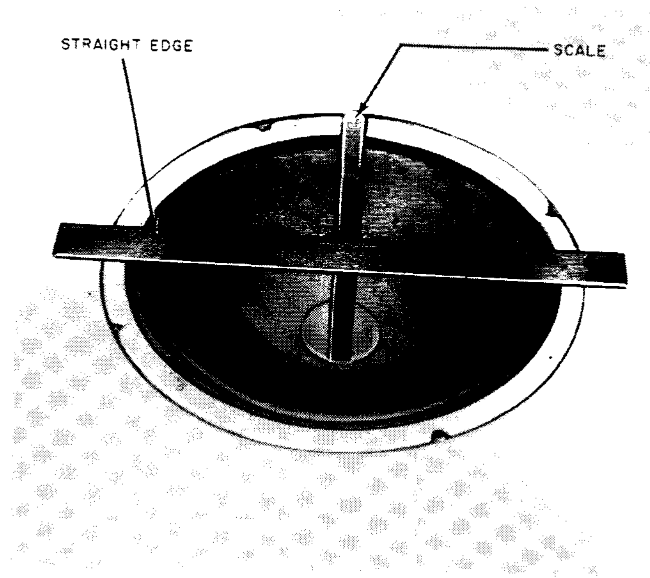


Fig. 7. Cone depth measuring arrangement for cone travel.

where: M is the added mass in grams
I is the voice coil current in milliamps

The term 9.8×10^6 is included to convert to proper units. An approximation of the length of wire on the voice coil can be made by measuring the d.c. resistance. 30 cm-per-ohm will get you close.

Linearity of cone travel: If the flux density of the air gap is not uniform throughout the distance traveled by the voice coil or if the flux does not adequately cover the path traveled by the voice coil, the force applied to the cone through the driving mechanism will not represent the true shape of the voice coil current. This, of course, results in a non-linearity and, consequently, distortion. Nonlinearity can also be caused by driving the speaker cone beyond the suspension limits of the suspension system. A static plot of cone displacement *vs* voice-coil current can be obtained with the same electrical connections shown in Fig. 10. Again the speaker is placed cone upward on a flat surface and the cardboard disc is placed in the apex of the cone as described previously. Instead of using the suspended plumb bob arrangement, however, a straight-edge is placed across the diameter of the speaker basket. A good scale is used to measure the distance from the top surface of the cardboard disc to the lower edge of the straight-edge. See Fig. 7. The distance is first measured with no current flowing in the voice coil. As a small current is caused to flow through the voice coil this distance is again measured. This procedure is repeated until the maximum current of the speaker is reached. Maximum current is determined from $I = \sqrt{P/Z}$ where P = power rating of the speaker and Z is the impedance of the speaker. The current is reduced to zero and the connections to the voice coil reversed. The measurements are repeated for the reversed polarity. A plot similar to Fig. 9 will be obtained. The cone will faithfully reproduce the waveform of the current flow if the plot is a straight line. Curvature near the extremes indicates non-uniform flux density or over extension of the suspended system.

Speaker efficiency: The speaker efficiency is one hundred times the ratio of acoustic power output to electric power input. A simple method for determining the efficiency of a speaker is described in reference 1 and attributed to Kennelly & Pierce. The electrical power delivered to the loudspeaker when operating normally is the product of the current squared times the resistive part of the loudspeaker impedance. The acoustic power delivered by the loudspeaker can be found by subtracting the electrical losses from the power input. Electrical losses can be found by blocking the speaker cone with shims so that movement is impossible and then measuring the resistive component of the speaker impedance under these conditions. The product of current squared times this resistive component represents the electrical losses of the speaker. The acoustic power is then the difference between total electrical power input and the electrical power losses. The efficiency equation becomes:

$$n = \frac{r_f - r_b}{r_f} \times 100 \quad . \ . \ . \ . \ (7)$$

where: n is the speaker efficiency
r_f is the resistive part of the speaker impedance with the cone itself free
r_b is the resistive part of the speaker impedance with the cone firmly blocked

A convenient adaptation of this method is easily achieved with the electrical connections shown in Fig. 11. The oscillator is set at a frequency well above the natural speaker resonance. C is adjusted for minimum voltage across the speaker-capacitor series circuit.

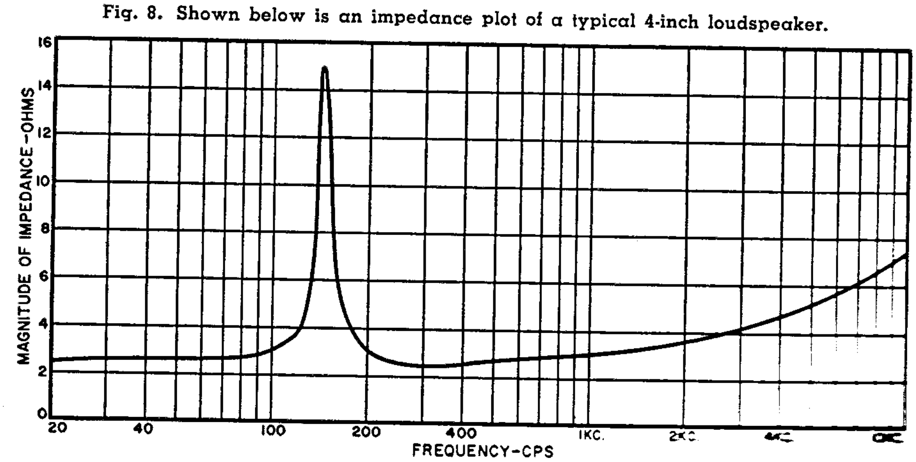
This minimum voltage is noted as E_r. The cone is blocked and C again adjusted for minimum voltage across the speaker-capacitor series. This minimum is noted as E_r'. If R is large, it can be shown that equation (7) can be closely approximated by:

$$n = \frac{E_r - E_r'}{E_r} \times 100 \quad . \ . \ . \ (8)$$

where: n, E_r and E_r' carry the notations mentioned above.

In one instance, R was made 1000 ohms and twelve volts of 2400—cps

Fig. 8. Shown below is an impedance plot of a typical 4-inch loudspeaker.



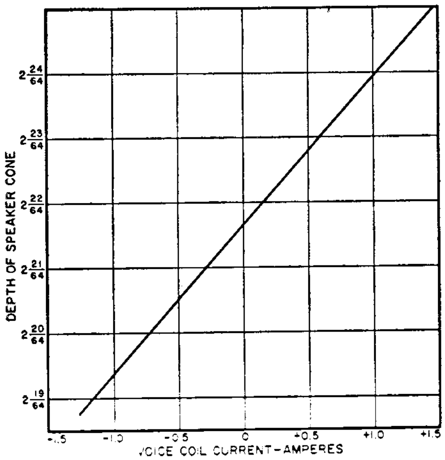


Fig. 9. Cone displacement of 10″ speaker.

voltage was applied from the oscillator. A minimum of .1 volt across the speaker and capacitor was observed when C was adjusted to 10 μfd. The cone was blocked and a minimum of .09 volt observed when C was adjusted to 14 μfd. In this case E_r = .1 volt and E_r' = .09 volt. From equation (8) the efficiency can be calculated to be 10% at 2400 cps.

Similar determinations can be made at other frequencies. As frequency is lowered, the value of capacitance necessary for a minimum becomes larger. An infinitely large capacitor is indicated at the natural resonant frequency of the speaker. There are several factors which affect the accuracy of this method. Not the least of these is the large possible error contributed in measuring when E_r' is nearly equal to E_r. This method also assumes mechanical losses negligible.

Speaker response: To take a speaker response curve the arrangement shown in Fig. 12 may be used. The oscillator must supply constant output to the speaker over the range of frequencies which interest you and the response of the microphone and amplifier must be known. It is true that this is asking a lot, for few of us have calibrated microphones lying about. In general, however, a broadcast quality microphone will be so much better than the average loudspeaker that it may be assumed to be flat. If it is desired to calibrate the microphone (and it is a worthwhile endeavor), several methods are available. A unique method is described in reference 3 as the "Reciprocity Technique of Calibration." This method is reasonably simple and does not require any more test equipment than is necessary for other tests described in this article. The process is quite lengthy so we will not go into the details but will refer the reader to Meyer[3] should he be sufficiently interested. Because the characteristics of the room will affect the response as measured by this arrangement, the microphone should be placed about one foot along the axis of the speaker. A reference is set at 400 cps and the frequency varied above and below this point and the reading of the db meter is noted. The response can then be plotted.

Multiple speakers: When multiple speakers are used, care must be taken to insure the proper phasing of the units. Out-of-phase speakers tend to counteract each other's efforts. A very simple method for checking the phasing of speakers is to connect a flashlight cell across the speaker leads and observe the direction of travel of each cone. All cones should travel outward at the same polarity. Reversing the voice-coil connections will reverse the direction of travel for a given battery polarity. Do not forget to phase both tweeter and woofer in a dual system. There will be frequencies in the region of crossover which will be reproduced by both. If a horn-type tweeter is used or if cone movement is not discernible, phasing can be done by using the two speakers as microphones and phasing for maximum output.

In a woofer-tweeter combination the selection of a crossover point should be governed by the low-frequency unit. The acoustic output of a speaker will be constant up to that frequency at which the cone no longer vibrates as a piston. This is called the break-up frequency of the cone. For a rigid cone this frequency is governed by the expression:

$$f_b = \frac{v}{2\pi R} = \frac{2100}{R} \quad . \; . \; . \; . \; (9)$$

where: f_b is the break-up frequency in cps

v is the velocity of sound in inches per second (13,200)

R is the radius of the speaker cone in inches

A rigid 15″ cone has a break-up frequency of 280 cps. For this reason wide-range speakers have concentric compliance rings formed into the cone to allow only a small portion of the cone to vibrate at the higher frequencies. This gives the advantage of a large cone at low frequencies and a small cone at the higher frequencies. The crossover point may be any point below that where piston action ceases. This frequency may be calculated from equation (9) where the radius of the smallest compliance ring is used for R. Once the choice of crossover point has been made, suitable crossover network design can be found in almost any reference book. Reference 2 provides excellent instructions, including coil winding information. Be sure to use as large wire as possible in the inductors to keep the resistive losses low.

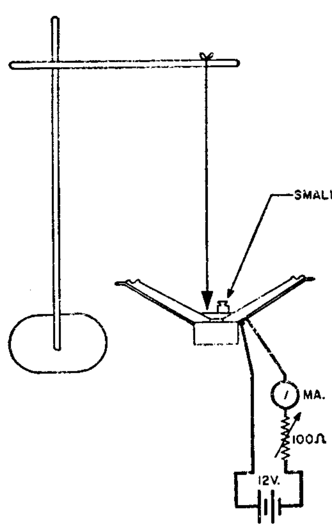
Speaker enclosures: Speaker enclosures may be divided into three general classes; infinite baffle, horn, and vented port or reflex. The infinite baffle makes an effort to separate the acoustic radiation at the rear of the speaker from that of the front. The horn-type baffle makes similar efforts and, in addition, an attempt is made at matching the radiation impedance of the speaker to the air load by means of the horn.

The bass-reflex cabinet not only has an adjustment which will vary performance but one which is mandatory for top performance. This is the tuned port which should be adjusted for the particular speaker to be used.

The bass-reflex cabinet is essentially a Helmholtz resonator tuned to the resonant frequency of the loudspeaker. See Fig. 14A. A simplified electrical equivalent circuit of a loudspeaker in such an enclosure is shown in Fig. 13. The speaker appears as a series-resonant circuit and the cabinet as a parallel-resonant circuit. If the cabinet is adjusted so as to have the same resonant frequency as the speaker, the effects of each tend to cancel. The result is an impedance curve having two humps of equal magnitude spaced equidistant on either side of the speaker resonant frequency. It appears as though the cabinet were of higher "Q" than the speaker and just notches out a portion of the resonant energy of the speaker in a manner not unlike the action of an absorption-type wavemeter. This notch can be moved along

(*Continued on page* 124)

Fig. 10. Electrical and physical arrangement for the measurement of flux density.



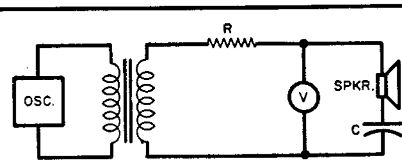


Fig. 11. Finding the speaker efficiency.

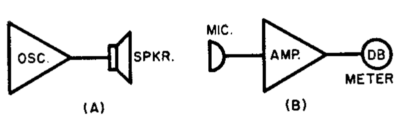
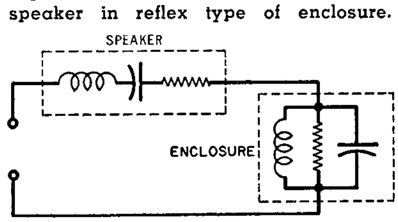


Fig. 12. Setup used to check response.

Fig. 13. Simplified equivalent circuit of speaker in reflex type of enclosure.



Loudspeaker Properties

(Continued from page 42)

by tuning and should be adjusted to coincide with the impedance peak produced by the speaker. The effect of high- and low-cabinet resonance is shown in Fig. 14B. These characteristics furnish a very convenient way to tune the reflex type cabinet, simply make the port larger than necessary then cover portions of it until an impedance plot possessing the desired double humps of equal amplitude is obtained. The amplitude of these humps can be reduced somewhat by lowering the "Q" of the cabinet to approach the "Q" of the speaker. Acoustic resistance is a property of the viscosity of the air and can be increased by stretching tightly woven fabric, such as silk or nylon, across the cabinet port. Be sure to stretch the fabric tightly so as not to add to the mass of the system, thus changing the resonant frequency.

As an interesting aside at this point, we may mention that almost any volume may be made into a Helmholtz resonator of practically any frequency. The limitation on size of a reflex type cabinet lies not in the volume necessary for resonance but in the size of the port necessary for sufficient low-frequency radiation. Any convenient volume may be used provided the tuning is carried out and the port size is sufficiently large. Port area is generally taken to be equal to or larger than the area of the speaker cone that is used.

The frequency of resonance of a Helmholtz resonator is given in reference 4 as:

$$f_c = 2070 \sqrt{\frac{\sqrt{A}}{V}} \quad . \; . \; . \quad (10)$$

where: f_c is the cabinet resonant frequency in cps
A is the port area in square inches
V is the cabinet volume in cubic inches. The equation may be solved for either A or V depending upon which is specified in a particular case. The value of f_c is, of course, determined by the speaker used. The area of the port should be made larger than the value obtained by calculation to permit tuning.

Speaker damping: One of the functions of a speaker enclosure is to load the speaker at its resonant frequency to remove the ringing effect. Whether or not this is being accomplished can be determined easily by what has come to be known as the "click-boom" method. This is merely listening closely to the speaker as a flashlight cell is alternately disconnected and connected to the voice coil. As the cell is connected a "click" will be heard as the cone is moved suddenly. When the cell is disconnected, the speaker is no longer damped by the cell and if the enclosure does not load the speaker properly, it will be free to vibrate at its resonant frequency and a mellow "boom" will be heard. The ideal would be where the sound were the same when the cell was disconnected as it was when the cell was connected to the voice coil.

Listening room acoustics: Most authorities agree that a reverberation time of ¾ second appears to be optimum for a listening room. This means

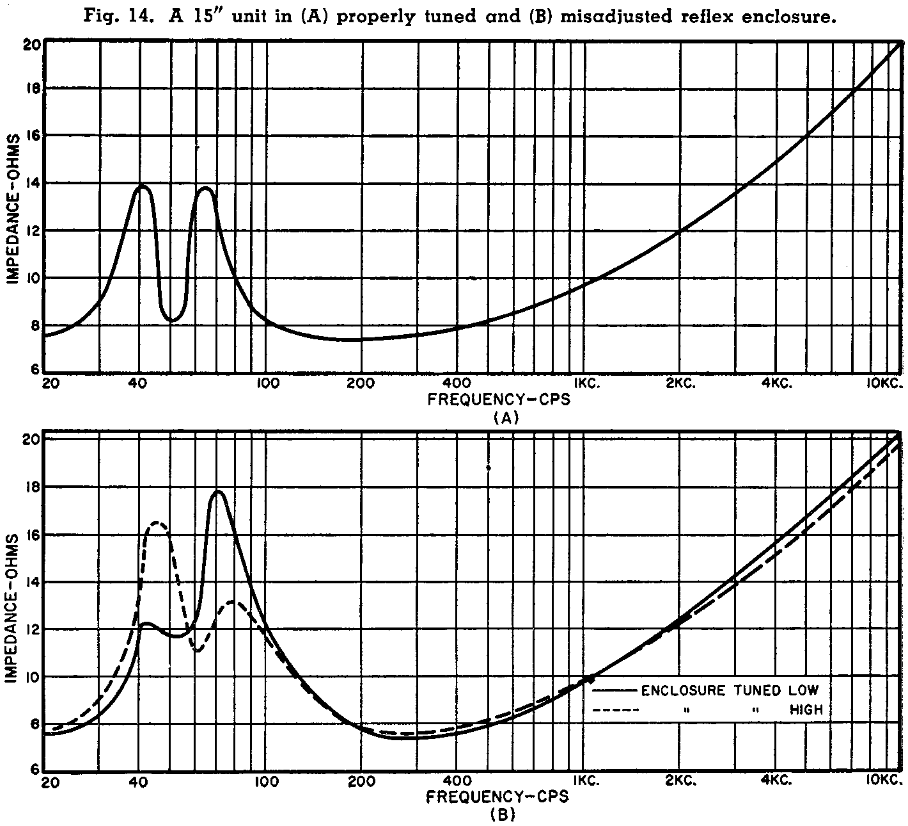


Fig. 14. A 15" unit in (A) properly tuned and (B) misadjusted reflex enclosure.

it should take ¾th of a second for the sound intensity to decay to one millionth of its original value. In the absence of a reverberation time meter, calculations can be made to get a reasonably good idea of the characteristics of the listening room. The reverberation time, T, is given by Sabine to be:

$$T = .05\left(\frac{V}{A}\right) \quad . \; . \; . \; . \quad (11)$$

where: V is the volume of the room in cubic feet and A is the total absorption units in the room in equivalent square feet of open window. A brief listing of absorption units for various materials is given in Table 2. For a more complete table see references 1, 5, and 6.

Perhaps the best way to explain the use of this relationship, is with an example. Consider a room 15 feet wide, 20 feet long, and 8 feet high. The floor is covered with waxed cork slabs, the ceiling and walls are of varnished knotty pine. There are 125 square feet of drapes which completely cover the windows. The room contains a couch, three large chairs, and is usually occupied by two adults. To calculate the total absorption units, multiply the surface of each type material by its absorption coefficient and add the results for different materials. In the room just described, the absorption units total 98 sabines. The volume of this room is 15′ x 20′ x 8′ = 2400 cubic feet. From equation (11):

$$T = \frac{.05 \times 2400}{98} = 1.225 \text{ sec.} \quad . \quad (12)$$

If it is desired to decrease this reverberation time to ¾ second, equation 11 can be solved for A when T and V are known. In our example:

$$A = \frac{.05V}{T} = 160 \text{ sabines} \quad . \; . \; . \quad (13)$$

This means we must add sufficient absorption to provide 160 — 98 or 62 sabines. If we place carpets on the floor we will lose 15 sabines and gain 75 sabines for a net of + 60 sabines. This is sufficiently close to our desired 62 sabines, thus placing a carpet in our hypothetical room would solve the problem.

The links in the chain of audio reproduction which lie between the amplifier and the ear are truly complex. The tests and measurements described are not intended to oversimplify the problem. Conversely, they are intended to provide interested persons with an appreciation of the complexity of the loudspeaker, its baffle, and the room in which we listen.

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