CHAPTER 20.

LOUDSPEAKERS

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SECTION 1 : INTRODUCTION

(i) Types of loudspeakers (ii) Direct-radiator loudspeakers (iii) Horn loudspeakers (iv) Headphones (v) Loudspeaker characteristics (vi) Amplitude of cone movement (vii) Good qualities of loudspeakers (viii) Loudspeaker grilles.

(i) Types of loudspeakers

In this chapter the point of view is that of the receiver or amplifier designer, not that of the loudspeaker designer.

Loudspeakers are in three principal groups :

1. Direct radiators—where the cone or diaphragm is directly coupled to the air.
2. Horn loudspeakers—where the diaphragm is coupled to the air by means of a horn.
3. Ionic loudspeakers—in which no diaphragm is used. See Refs. 212, 227.

(ii) Direct radiator loudspeakers

(A) The moving coil or electro-dynamic type is by far the most popular for radio receivers. It has a voice-coil mounted in a strong magnetic field ; the a-f output power from the amplifier is passed through a suitable matching transformer (see Chapter 21) to the voice coil, and the a-f currents in the coil cause a force to be exerted in the direction of the axis of the coil. If the voice coil is free, it will move in the direction of the force. The voice coil is loaded by the cone which is (normally) firmly fixed to it.

The magnet may be either an electromagnet or a permanent magnet. In all cases, the higher the flux density the greater the efficiency and electro-magnetic damping. See Sect. 2 for detailed description.

(B) The electromagnetic type has two forms, the **reed armature** and the **balanced armature.** In the former there is a steel reed armature at a short distance from the pole piece wound with insulated wire carrying the a-f current, and supplied with steady magnetic flux by means of a permanent magnet. In the balanced armature type, only the a-f flux flows longitudinally through the armature. Both types require relatively stiff springs to return the armature to the position of equilibrium, so that the bass resonance usually occurs at a frequency above 100 c/s.

Refs. 13, 14, 15.

(C) **The inductor dynamic type** differs from the older electromagnetic type in that the two iron armatures are balanced so that they lie at rest without a strong spring tension. The natural resonance frequence may be about 70 c/s, and the amplitude of movement is about the same as that of a moving-coil type. Ref. 15.

(D) **The piezo-electric** (" **crystal** ") type generally uses Rochelle salt for the bimorph elements. It takes two forms, in which the elements either bend or twist with the voltage applied between the two electrodes. The crystal has a predominately capacitive impedance, the capacitance being of the order of 0.02 μF.

The impedance varies considerably with frequency in both scalar value and phase angle, so that it is difficult to provide a correct load impedance for the output valve. This is not very important with a triode or if another loudspeaker is connected in parallel.

The crystal loudspeaker naturally responds to the higher frequencies, and is sometimes used as a " tweeter " in a dual or triple speaker system (Sect. 5). In this case the response may be maintained roughly constant over the desired band of frequencies by a small inductance in series with the primary of the transformer. Crystal loudspeakers are not normally used alone.

Refs. 13, 14, 24, 60.

(E) **The condenser type** of loudspeaker usually takes the form of one large solid electrode, and a thin movable electrode which is mounted in such a way that it can vibrate (usually in sections) without touching the solid electrode.

Either the solid or the movable electrode may be corrugated ; the movable electrode may be of insulating material with a metal foil coating. An alternative form has both electrodes flexible (Ref. 22).

The sensitivity is dependent upon the total area ; with 300 to 500 square inches the sensitivity may approximate that of an electro-dynamic type.

The Kyle speaker (Ref. 58) has a capacitance of 0.004 μF for an area of 96 square inches. The impedance is predominately capacitive. A high polarizing voltage is required for high efficiency and good performance—500 to 600 volts is usual.

Condenser loudspeakers are little used at the present time.

Refs. 13, 14, 15, 22, 58.

(F) **Throttled air-flow loudspeaker**—this consists of a mechanical valve, actuated by an electrical system, which controls a steady air stream, the air flow being made proportional to the a-f input current. It is normally limited to very large sizes.

Ref. 14.

(iii) Horn loudspeakers

Any type of driving unit actuating a cone or diaphragm may be given greater acoustical loading by means of a horn, thereby improving the efficiency and power output. The horn ceases to have any beneficial effect below a frequency whose value depends on the law of expansion and the size of the horn. Horn loudspeakers have considerably greater efficiency than direct radiators. For further information see Sect. 4.

(iv) Headphones

Headphones (telephone receivers) make use of a diaphragm which is effectively sealed to the ear by means of a cap with a central opening. The pressure of the small quantity of air enclosed between the diaphragm and the ear drum varies in accordance with the displacement of the diaphragm.

The driving mechanisms of headphones resemble those for loudspeakers.

(A) **Magnetic diaphragm** : The a-f force caused by the a-f current in the electromagnet operates directly upon a steel diaphragm. The bipolar type is the most popular of all headphones at the present time. A permanent magnet supplies the steady flux. At frequencies above the second diaphragm resonance the response falls off rapidly. Refs. 13, 14, 59, 62.

(B) Moving armature type : The principles are the same as for loudspeakers. Ref. 14.

(C) Moving-coil type : This follows the same principles as the loudspeaker, and with careful design is capable of greater fidelity and wider frequency range than the magnetic types. One model has a response from 10 to 9000 c/s $\pm$ 2 db except for a dip of 6 db at 4000 c/s. (Ref. 63). Refs. 13, 14, 63.

(D) Crystal type : The impedance of one model is 80 000 ohms (predominately capacitive) at 10 000 c/s. A high resistance may be connected in series to raise the low frequency response relatively to the high frequency response. Refs. 13, 64.

(E) Ribbon type : The principle is the same as for a ribbon microphone. It is only used for high-fidelity reproduction. Refs. 13, 14, 65.

(F) Inductor type : This has been developed to deliver practically constant sound pressure to the ear cavity from 50 to 7000 c/s. It has a V shaped diaphragm driven by a straight conductor located in the bottom of the V. Refs. 13, 14, 66, 67.

Correction circuits for magnetic diaphragm headphones

A typical magnetic diaphragm unit has a pronounced peak slightly below 1000 c/s. A circuit for attenuating this peak and thereby giving reasonably uniform response is Fig. 20.1 (Ref. 25).

FIG. 20.1

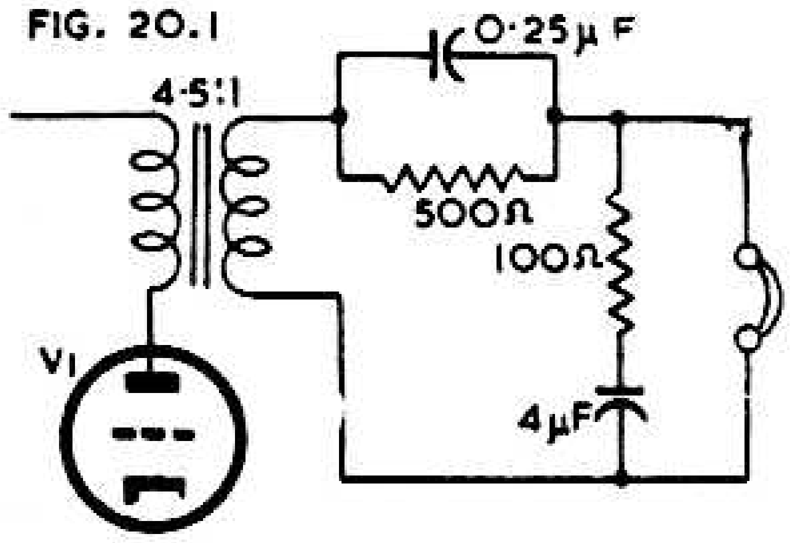


Fig. 20.1. Tone correction circuit for magnetic diaphragm headphones (d.c. resistance 4000 *ohms) giving* 15 *db bass and treble boost. Secondary impedance roughly* 500 *ohms, primary load resistance* 10 000 *ohms (Ref.* 25).

(v) Loudspeaker characteristics

The amplifier designer is mainly interested in a limited number of characteristics.

Impedance : See Sect. 2(iv) for moving-coil type, and Sect. 4(v)G for horn type. All types of electro-magnetic loudspeakers have a somewhat similar form of impedance characteristic, although the relative values vary considerably.

Phase angle : See Sect. 2(iv) for moving-coil-type.

Frequency response characteristic

This only applies under the strict conditions of the test, particularly generator impedance, and is very often misleading. The most serious defect is the existence of sharp upward peaks extending higher than say 5 db above the smooth curve, particularly in the region from 1500 to 4000 c/s. These are not always shown by the published curves, which frequently show signs of having been smoothed.

Reflections in rooms may result in changes of 15 db or more in level at certain frequencies, and the shape of the frequency characteristic may be altered out of all recognition by variations in placement of loudspeaker and microphone, or by a change of room.

It is important to know the source impedance (i.e. the output resistance of the amplifier) used in the test. A loudspeaker should always be used with the correct source impedance—too low a source impedance may result in loss of low frequency response, while too high a source impedance will usually result in over-accentuation of a narrow band of frequencies in the vicinity of the loudspeaker bass resonance. See Sect. 9(ii) for R.M.A. Standard loudspeaker measurement source impedance. Refs. 13, 14, 15, 26 (No. 1), 42, 70.

Damping

The source impedance (i.e. the output resistance of the amplifier) also affects the damping of the loudspeaker at the bass resonant frequency. One value of source impedance will normally (in loudspeakers of high flux density) give critical damping—this being the value for no overshoot—while a slightly lower degree of damping may be considered as the optimum. Too high a resistance will give insufficient damping for good transient response. For further information see Sect. 2(x).

Directional characteristics

The usual frequency response characteristic is measured with the microphone on the axis of the loudspeaker, this being the position of maximum high frequency response. There is a serious loss of higher frequencies (4000 c/s and above) even with the microphone 30° off the axis.

See Sect. 2(vii) for further information on moving-coil types, and Sect. 4(vii) for horn types.

Refs. 13, 14, 15, 26 (No. 1), 61, 70, 133.

Non-linear distortion : see Sect. 7(i) and (v).

Efficiency versus frequency characteristic : see Sect. 2(vi).

Transient response : see Sect. 7(iii).

Matching : see Chapter 21.

Characteristics of cones : see Sect. 2(ii).

(vi) Amplitude of cone movement

The peak amplitude of a rigid disc to give an acoustical power output of 0.2 watt (equivalent to 6.7 watts amplifier power with a loudspeaker efficiency of 3%) is given by the following table. Radiation from one side only is considered (derived from Chart No. 61, Ref. 69).

Diameter of cone	Frequency (c/s)					
	1000	400	200	100	50	30
13 in.	0.0003	0.002	0.008	0.03	0.12	0.34 in.
10 in.	0.0005	0.003	0.013	0.05	0.21	0.58 in.
8 in.	0.0008	0.005	0.02	0.08	0.32	0.90 in.
5 in.	0.002	0.013	0.051	0.21	0.82	2.26 in.

(vii) Good qualities of loudspeakers

A loudspeaker should have the following good qualities—

Satisfactory sensitivity
Broad directivity
Low distortion over the whole frequency range
Smooth frequency response
Balanced response
Good transient response
Sufficient damping at the bass resonant frequency
Adequate power-handling capacity

Any loudspeaker is necessarily a compromise. It has been demonstrated by Olson (Ref. 133) that with cone type loudspeakers, increased sensitivity brought about by reducing the thickness of the cone material is accompanied by increased distortion, ragged frequency response, bad transient response, and increased "break up" effects in the cone. On the other hand, the power output for the same distortion is many times greater for the loudspeaker having the more massive vibrating system.

If increased sensitivity is required, this may be achieved, within limits, by increasing the flux density. High flux density has the additional valuable property of increasing the damping.

A prominent peak in the response characteristic at the bass resonant frequency will increase the apparent sensitivity of the loudspeaker, particularly with small speakers having the bass resonance above 150 c/s. Similarly, prominent peaks in the response characteristic above 1000 c/s will also increase the apparent sensitivity, although both are features to be avoided for good fidelity.

When loudspeakers are compared in listening tests, the sound outputs should be adjusted to the same level before a comparison is made, because sensitivity is usually of secondary importance. Refs. 150, 155, 183.

(viii) Loudspeaker grilles

If the grille cloth is not to reduce sound pressure, its resistance must be small compared with the radiation resistance of the loudspeaker. The most suitable type of cloth is one which is very loosely woven and which has hard threads—cotton or plastic. Fuzzy threads increase the resistance of the cloth. Ref. 180.

SECTION 2 : CHARACTERISTICS OF MOVING-COIL CONE LOUDSPEAKERS

(i) Rigid (piston) cone in an infinite flat baffle (ii) Practical cones (iii) Special constructions for wide frequency range (iv) Impedance and phase angle (v) Frequency response (vi) Efficiency (vii) Directional characteristics (viii) Field magnet (ix) Hum bucking coil (x) Damping.

(i) Rigid (piston) cone in an infinite flat baffle

A rigid cone is one which moves forwards and backwards like a piston—at low frequencies a loudspeaker cone approaches this ideal. The air acts as a load on the piston at all frequencies when it is set in an infinite flat baffle. The **radiation resistance per unit area** of the piston is approximately constant for frequencies above

$$f_1 = 8120/\text{diameter of piston in inches} \qquad (1)$$

In tabular form :

Diameter of piston	12	10	8	6	4	inches
Frequency f_1	677	812	1015	1353	2030	c/s.

Below frequency f_1 the radiation resistance is approximately proportional to the square of the frequency.

For amplitude of cone movement see page 834.

(ii) Practical cones

In practical cones the voice coil is mounted towards the apex of the cone and the driving force is transmitted through the cone. The cone must be a compromise between strength and lightness. A cone for reproducing the low audio frequencies should be fairly rigid, and should have a large diameter. If such a loudspeaker is tested at high audio frequencies above 1000 c/s it will be found to have a poor response owing to :

1. Large mass, causing poor efficiency at the higher frequencies.

2. The vibrations tend to take the form of waves radiating from the voice-coil outwards through the cone. At the higher frequencies the outer portion of the cone may be out of phase with the inner portion, or there may be a phase difference greater or less than 180°

Cones for reproducing a wide frequency range often are designed with corrugations to reduce the effective vibrating area at the higher frequencies. Other designers achieve a somewhat similar result by a suitable choice of cone material together with changes in the thickness or compliance of the material. It is desirable that the material for the cone be no harder than is necessary to maintain response up to the highest frequency required, in order to give good reproduction of complex tones and transients.

The cone and voice-coil assembly is usually mounted at two points—an annular "surround" and a "spider" or centre mounting. Each of these introduces a restoring force which should increase linearly with the displacement. Actual mounting systems introduce a non-linear restoring force which usually increases rapidly when the displacement is fairly large. The suspension has two characteristics directly affecting the performance of the loudspeaker—compliance and mechanical resistance. Compliance is the inverse of stiffness (e.g. the stiffness of a spring), and controls the frequency of the bass resonance. The mechanical resistance of the suspension affects the Q of the vibrating system, but usually at the bass resonant frequency the damping due to the suspension is much less than that due to the output resistance of the amplifier, except when a high output resistance is used. There appears to be an optimum relationship between the stiffness of the cone and the compliance of the rim to minimize peaks and troughs in the 1500 to 3500 c/s region.

Non-linear suspensions are sometimes used with the object of preventing damage to the cone through overloading. This procedure causes serious distortion at high amplitudes, and the quoted value of " maximum power output " is that capable of being handled without the loudspeaker suffering damage. Loudspeakers for good fidelity should be capable of handling their full power without exceeding the limits of a fairly linear restoring force. These should be rated on the maximum power (within certain frequency limits) which they are capable of handling without the distortion exceeding a predetermined value.

As one result of the suspension system, the whole cone assembly tends to resonate at the **bass resonant frequency.** Below this frequency the cone is stiffness-controlled by the suspension and the output waveform tends to have flattened peaks resulting from the non-linear suspension.

Most loudspeaker manufacturers supply at least two alternative types of cone for use with each model—one with a nearly flat response up to 6000 c/s or higher (for broadly tuned or F-M receivers or amplifiers), the other with a response peaked in the 2000 to 4000 c/s region to compensate for sideband cutting in the receiver. It is desirable, in each case, to have a choice between a cone for pentode output (without feedback), and one for triode output (or pentode with negative voltage feedback).

Cones are also available with differing bass resonant frequencies (from below 50 c/s to about 225 c/s depending on cone size and application) and differing high frequency limits (say 4500 to 8500 c/s for single cone units). Most manufacturers also supply special cones to order. It is of the utmost importance for the receiver or amplifier designer to investigate the complete range of available models and cones, and to make comparative listening tests between different makes, models and cones.

A loudspeaker using a metal cone is described in Ref. 225.

Amplitude and phase measurements on cones are described in Ref. 179.

References to cones : 1, 13, 15, 16, 19, 23, 40, 54 (Part 4), 84, 133, 179, 225.

(iii) Special constructions for wide frequency range

As mentioned above, a simple cone loudspeaker is not capable of giving satisfactory performance over a wide frequency range. One solution is to use two or more loudspeakers with each designed to cover a limited frequency range or dual or triple integral units (see Sect. 5). Another solution is to use some device to extend the effective frequency range of a single unit, as described below.

(A) Multiple single coil, single cone : It is possible to employ two or more identical fairly light loudspeakers with fairly small cones, all connected in parallel, series or series-parallel and correctly phased. The number to be so connected depends on the maximum power to be handled at the lowest frequency. All loudspeakers operate at all frequencies, and no filter is used.

(B) Single coil, double cone : This consists of an ordinary voice coil with two cones, the smaller cone being firmly fixed to the voice coil, and the larger cone being flexibly connected by means of a compliance corrugation. At low frequencies both cones move together as a whole, but at high frequencies the small cone moves while the large cone remains stationary.

Refs. 13A, 14, 71.

(C) Double coil, single cone : This consists of a voice coil in two parts separated by a compliance, the smaller being fixed to a single corrugated cone. The larger part of the voice coil is shunted by a capacitance to by-pass the higher frequencies. At low frequencies the whole assembly moves together, but at high frequencies the larger part of the voice coil remains stationary, and the smaller part drives the cone. The corrugations in the cone are designed to decrease the effective cone area at the higher frequencies.

Refs. 13A, 14, 57.

(D) Double coil, double cone : This consists of a light coil coupled to a small cone, the light coil being connected by a compliance to a heavy coil which is firmly fixed to a large cone. The heavy voice coil is shunted by a capacitance to by-pass the higher frequencies. At low frequencies both voice coils are operative, and the whole assembly moves together, but at high frequencies the small cone is driven by the light coil, and the large cone and heavy coil remain stationary. This is really equivalent to two separate loudspeakers.

Refs. 13A, 71.

(E) Duode : This has a light aluminium sleeve as the voice coil former covered on the outside by a layer of rubber, over which is wound the voice coil. At high frequencies, it is claimed, the relatively heavy voice coil winding remains stationary, the aluminium sleeve acting as a one turn coil receiving its power by transformer action and driving the cone. Refs. 242, 243.

(iv) Impedance and phase angle

The impedance characteristic (e.g. Fig. 20.2) is drawn from measurements made on an electrical impedance bridge, or equivalent method, with the loudspeaker mounted on a flat baffle. The nominal value is usually taken as the impedance at 400 c/s and is reasonably constant from about 200 c/s to about 600 c/s. At lower frequencies it rises to a peak at the bass resonant frequency and then falls rapidly to the resistance of the voice coil at still lower frequencies ; at higher frequencies it rises steadily throughout.

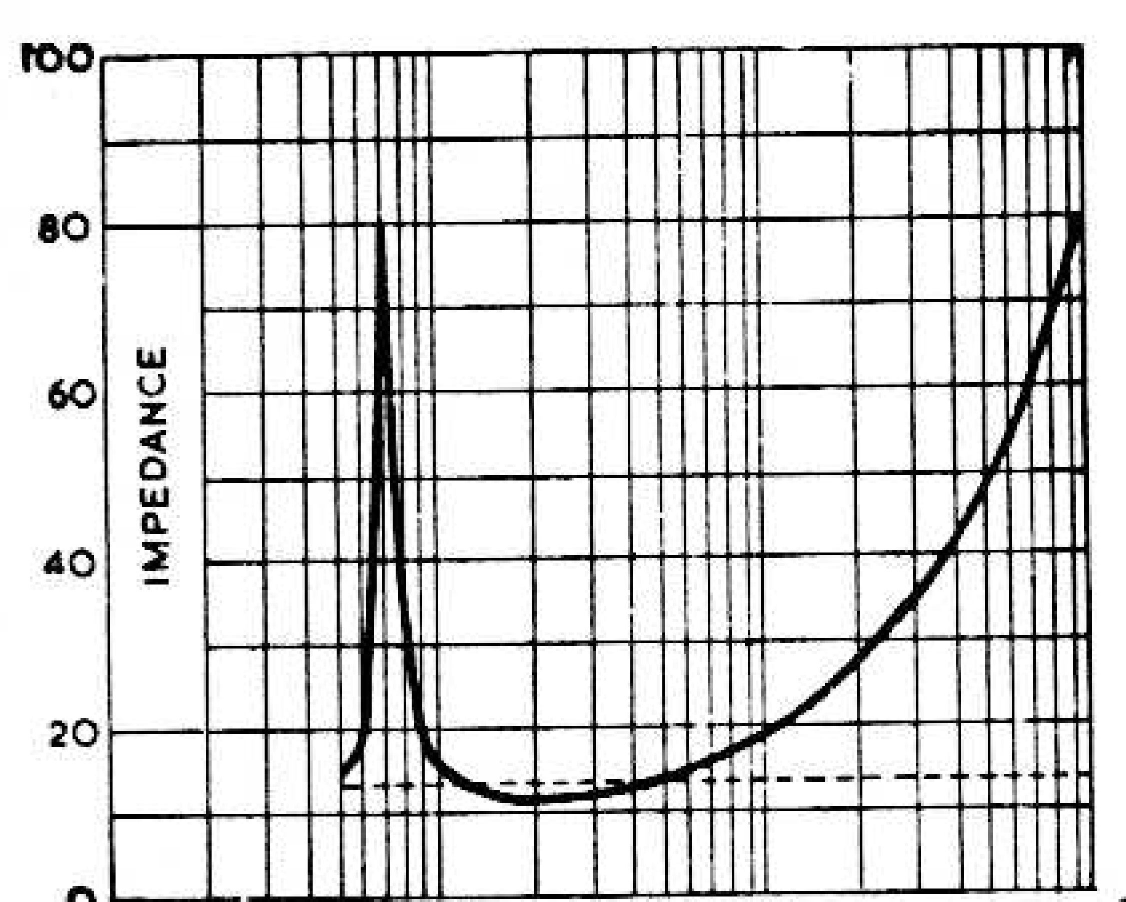


Fig. 20.2. Impedance versus frequency characteristic of a typical popular loudspeaker.

The impedance at the bass resonant frequency is a function of the damping of the cone suspension, the height of the peak decreasing as the damping is made heavier (see Ref. 13A, Fig. 6.31). It is also a function of the applied voltage, owing to non-linearity of the cone suspension.

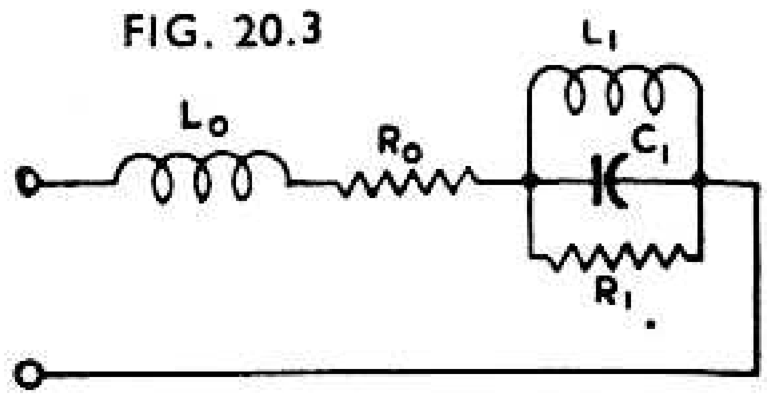
If the loudspeaker is mounted other than on a flat baffle, the impedance characteristic at frequencies in the vicinity of the bass resonant frequency may be modified. For example with a vented baffle there are normally two impedance peaks —— Sect. 3(iv).

The impedance at 10 000 c/s is always greater than that at 400 c/s, the ratio varying between 1.1 and 10 times. A high ratio is most undesirable with pentode operation, with or without feedback, but is of no great consequence with triodes. The impedance at 10 000 c/s is increased by any leakage inductance in the transformer.

The equivalent electrical circuit which gives approximately the same impedance characteristic up to 400 c/s is shown in Fig. 20.3 (Refs. 15, 72). If adjustments are made in the values of L_0, R_0, R_1 and C_1, the circuit may be extended to higher frequencies. The bass resonant frequency is the parallel resonance of L_1 and C_1.

Fig. 20.3. *Equivalent electrical circuit providing the same impedance characteristic up to* 400 *c/s as a moving-coil loudspeaker. Typical values are* $L_0 = 0.0024\ H$, $R_0 = 10.4$ *ohms*, $L_1 = 0.018\ H$, $R_1 = 71.6$ *ohms*, $C_1 = 282\ \mu F$.

FIG. 20.3



The impedance so far discussed is called the " free " impedance ; there is also the " blocked " impedance which is measured with the voice coil prevented from moving. The " motional " impedance is found by a vector subtraction of the " blocked " from the " free " impedance. The difference between the free and blocked impedances is small except in the vicinity of the bass resonant frequency, where the motional impedance becomes the major portion of the free impedance (Refs 13, 41).
Refs. 13, 13A, 14, 15, 26 (No. 2), 36, 37, 41, 47, 70.

Phase Angle : The phase angle versus frequency characteristic of a typical moving-coil loudspeaker is given in Fig. 20.4. The impedance is resistive at two frequencies only, being capacitive between the two points and inductive at lower and higher frequencies. The resistive and reactive components are shown in Fig. 20.5. Refs. 13, 13A, 37, 41, 61.

FIG. 20.4

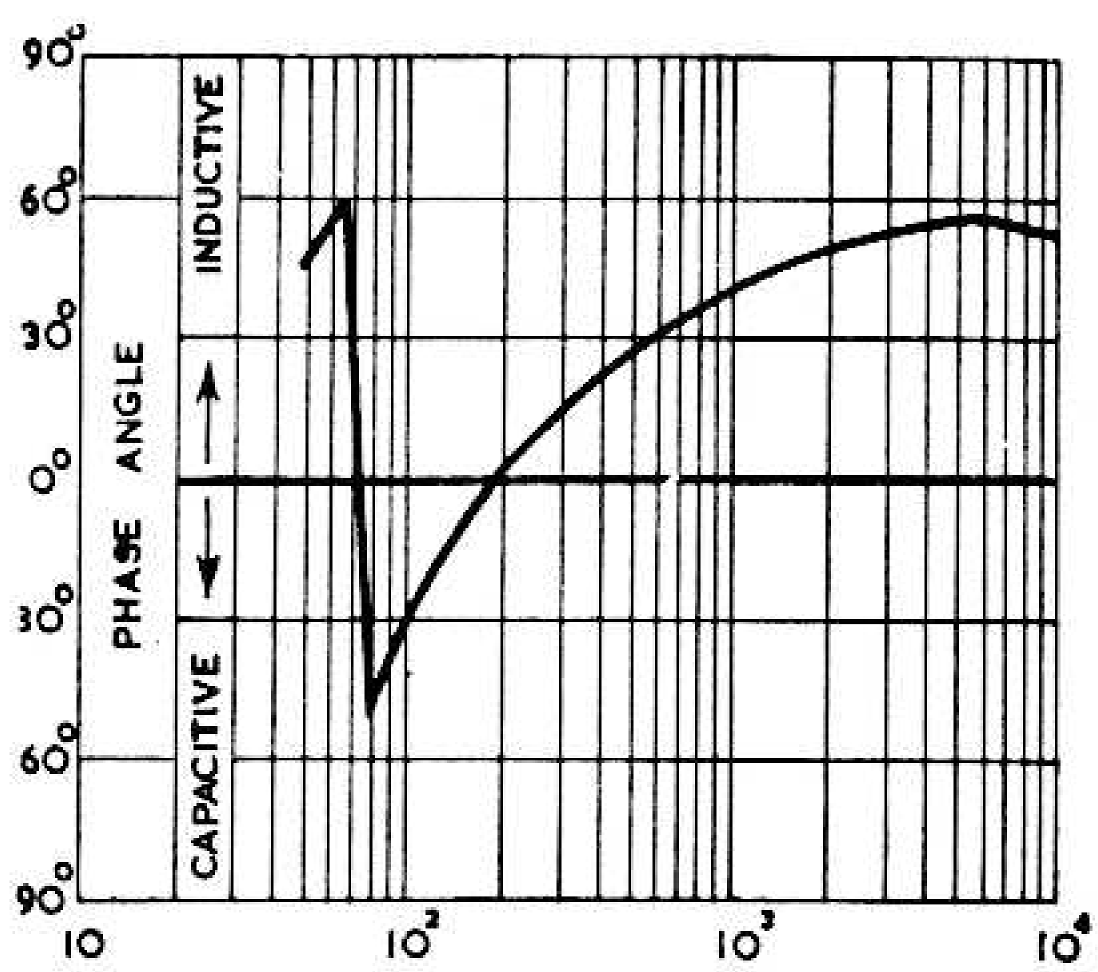


Fig. 20.4. *Phase angle versus frequency characteristic of a typical popular loudspeaker (same as Fig.* 20.2).

(v) Frequency response

When used on an adequate flat baffle, a moving-coil loudspeaker only operates satisfactorily at and above the bass resonant frequency. The upper limit of frequency response rarely extends much beyond 6000 c/s for popular 10 or 12 inch single coil, single cone loudspeakers. Any further extension adds appreciably to the cost. If wide frequency range is required, there is a choice between a complex single unit—see (iii) above—and a multiple unit (Sect. 5). Refs. 13, 13A, 14, 15, 26 (No. 1), 42, 61, 70.

FIG. 20.5

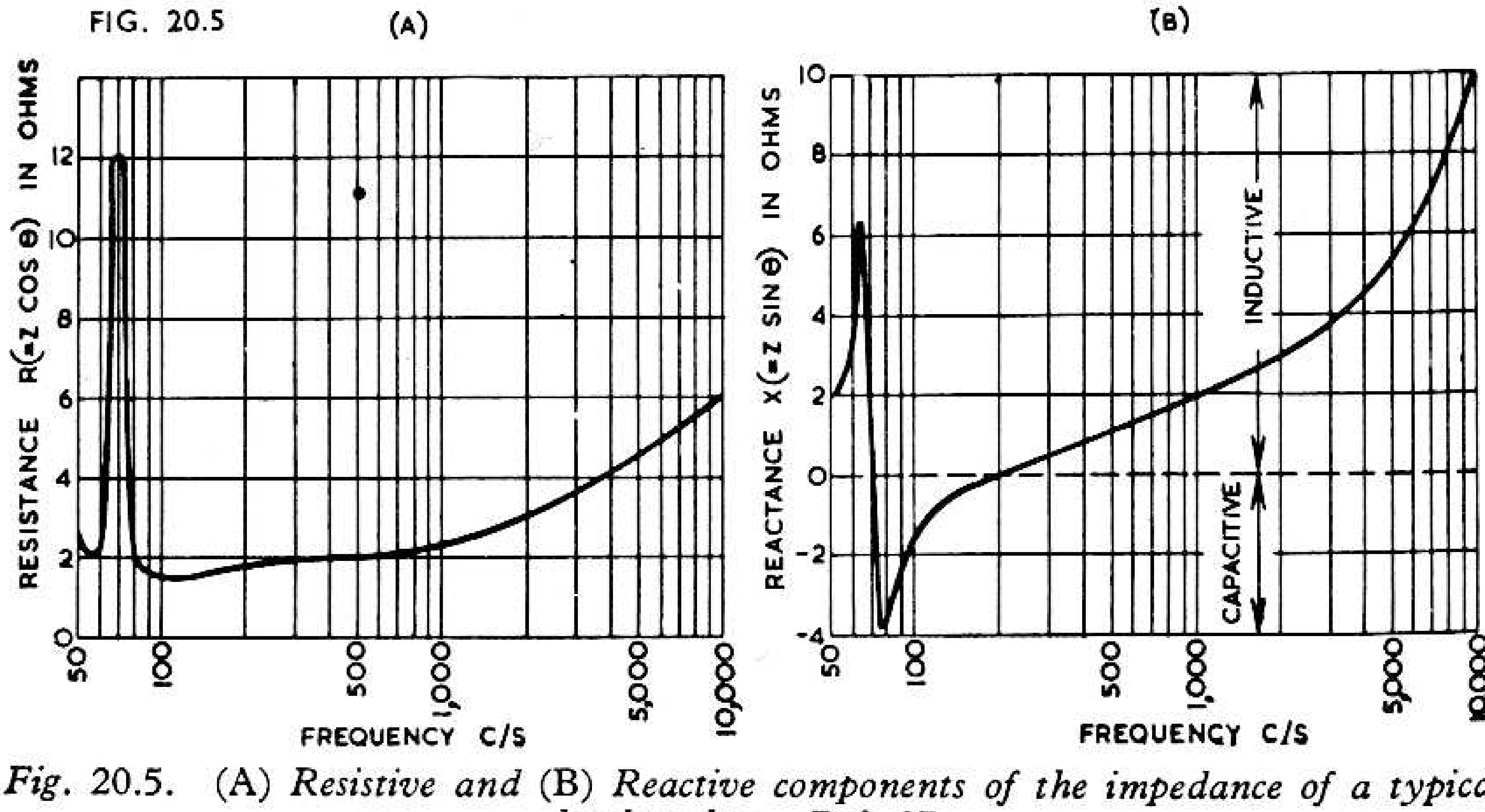


Fig. 20.5. (A) *Resistive and* (B) *Reactive components of the impedance of a typical loudspeaker. Ref.* 37.

(vi) Efficiency

Loudspeaker efficiency may be regarded either as the efficiency of the loudspeaker itself, or in the form of available power efficiency.

The efficiency of the loudspeaker itself is the ratio of the radiated acoustical power to the electrical input power, and is not influenced by the regulation of the power source. The efficiency of moving-coil cone loudspeakers on large flat baffles varies from about 2% to 10% at 400 c/s, depending on the design and on the flux density, 3% being a typical average. The maximum possible electro-acoustical efficiency is given by (Fig. 20.7)

$$\eta_{max} = R_2/(R_0 + R_2 + R_3) \quad (2)$$

and the electro-mechanical efficiency is given by

$$\eta_{em} = (R_2 + R_3)/(R_0 + R_2 + R_3) \quad (3)$$

where R_0 = blocked (d.c.) resistance of voice coil
R_2 = equivalent radiation resistance
and R_3 = equivalent frictional and eddy current loss resistance.

The electro-acoustical efficiency is always less than the electro-mechanical efficiency owing to the losses represented by R_3. It is possible to measure the electro-mechanical efficiency fairly readily because $(R_2 + R_3)$ is the motional resistance, and

$$(R_2 + R_3) = \text{free resistance} - R_0 \quad (4)$$

but the accuracy is very poor except at frequencies in the region of the bass resonant frequency.

The available power efficiency is usually defined as the electrical power available to the load when the loudspeaker is replaced by a resistance equal to the rated load impedance with a constant voltage applied in series with the loudspeaker measurement source impedance (see pp. 812, 874-876). The electrical power W_{AS} available to the loudspeaker is

$$W_{AS} = E_G^2 R_{SR}/(R_{SG} + R_{SR})^2$$

where E_G = r.m.s. value of constant source voltage
R_{SR} = loudspeaker rating impedance
and R_{SG} = loudspeaker measurement source impedance.

If R_{SG} = 40% of R_{SR} (as in SE-103, see page 874), then the power available to the load will be half that when $R_{SG} = 0$. Maximum available power efficiency occurs at the bass resonant frequency if the baffle is sufficiently large and if the amplifier output resistance is fairly high, but if the output resistance is low, the available efficiency may be less at this frequency than at some higher frequency. The available power efficiency/frequency characteristic at low frequencies is affected by the type of baffle—see Sect. 3(iii) and (iv).

FIG. 20.7

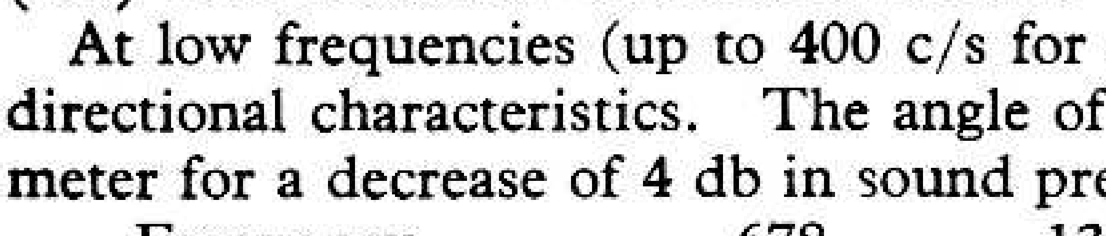


Fig. 20.7. *Equivalent circuit for power considerations, at frequencies above the bass resonance. The values of R_2, R_3 and C_m are not constant with change of frequency.*

Refs. to efficiency, 13, 13A, 14, 15, 36, 43, 54 (Part 3), 56, 70, 71, 74, 95, 121, 156.
References to equivalent circuits, 13, 13A, 15 (page 135), 41, 43, 54 (Part 3), 59, 75, 166, 188.

(vii) Directional characteristics

At low frequencies (up to 400 c/s for a 10 inch cone) a cone has only very slight directional characteristics. The angle of radiation from a flat disk 10 inches in diameter for a decrease of 4 db in sound pressure as compared with that on the axis is :

Frequency	678	1355	2710	5420	c/s
Angle (approx.)	180°	70°	35°	18°	

In practice it is possible to improve the angle of radiation by various expedients. One is to fit a deflector or diffusing lens (Ref. 120) so as to spread the higher frequency waves. Another is to use corrugations (compliances) in the cone so that only a small central area is vibrating at the higher frequencies. A further expedient is to use two co-axial loudspeakers and a frequency dividing network, the high frequency unit having some form of deflector. The ordinary popular loudspeaker is highly directional at high frequencies, but the best of the expensive designs are capable of 120° radiation up to 15 000 c/s (Refs. 36, 61).

General references : 13, 14, 15, 54 (Part 2), 70.

(viii) Field magnet

Field magnets may be either electro-magnets or permanent magnets. In either case the requirement is to provide the greatest possible flux density in the air gap. Fidelity loudspeakers should have a uniform field up to the limits of movement of the voice-coil ; cheaper models are usually lacking in this respect. Alternatively the voice coil may be longer than the gap length (Ref. 15).

Permanent magnets have advantages in that they have lower heat dissipation, leading to a lower ambient temperature for the voice coil, and the voltage drop across a suitable choke is less than that across the field coil.

Increasing the flux density increases the efficiency, the output and the damping ; it reduces the rise of impedance at the bass resonant frequency. There is an optimum value of flux density to give the most nearly uniform frequency response (Ref. 56, Fig. 3).

(ix) Hum-bucking coil

Some electro-magnetic fields are fitted with a hum-bucking coil connected in series with the voice coil such that the field coil induces equal hum voltages in the voice coil and hum-bucking coil. By connecting the hum-bucking coil and the voice coil in series opposition, the hum due to ripple in the field is minimized.

(x) Damping

Damping on a loudspeaker is partly acoustical, partly frictional, and partly electro-magnetic. The electro-magnetic damping is a function of the effective plate resistance of the power amplifier.

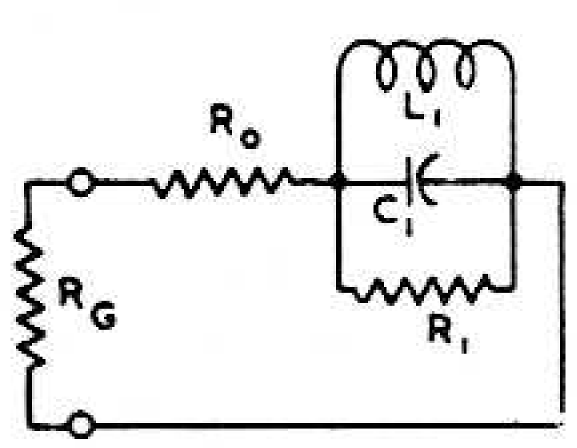


FIG. 20.8

Fig. 20.8. Equivalent circuit of loudspeaker at bass resonant frequency, showing damping effect of amplifier output resistance R_G.

With a direct radiator loudspeaker on a flat baffle the damping is slight, except in the vicinity of the bass resonant frequency, where the electro-magnetic damping may become quite large. The equivalent circuit at the bass resonant frequency is given by Fig. 20.8, which is identical with Fig. 20.3 except that L_0 has been omitted as being negligibly small, and R_G has been added to allow for the damping effect of the amplifier output resistance. R_1 is the damping resistance equivalent to the combined effect of friction and acoustical loading on the cone. R_0 is equal to the resistance of the voice coil plus that of the secondary of the transformer and leads plus that of the primary referred to the secondary. R_G is the effective plate resistance of the valve referred to the secondary. Any reduction in R_G below about one fifth of R_0 has only a very small effect on the Q of the tuned circuit.

At the bass resonant frequency, the Q of the tuned circuit is given by—

Case 1—R_G is infinite $Q = \omega_0 C_1 R_1$

Case 2—$R_G = 0$ $\qquad Q = \omega_0 C_1 \dfrac{R_1 R_0}{R_1 + R_0}$

Case 3—general $\qquad Q = \omega_0 C_1 \dfrac{R_1(R_0 + R_G)}{R_1 + R_0 + R_G}$

where ω_0 is angular velocity at the bass resonant frequency.

Critical damping is defined* as given by $Q = 0.5$.

Hence for critical damping,

$$R_G = \frac{R_1 + R_0 - 2\omega_0 C_1 R_1 R_0}{2\omega_0 C_1 R_1 - 1} \qquad (5)$$

*See " American Standard Definitions of Electrical Terms " A.S.A. C42—1941 (05.05.365) ; also C. E. Crede " Vibration and Shock Isolation " John Wiley & Sons, Chapman & Hall Ltd. 1951, p. 171, and other textbooks on vibration. However some engineers take $Q = 1$ as critical damping.

From Fig. 20.8 it will be seen that critical damping is only possible with a positive value of R_G when R_0 is not too high, that is to say with loudspeakers having fairly high efficiencies, since the values of L_1 and C_1 are functions of the flux density. The damping of any loudspeaker may be increased by increasing the flux density.

In general, there is no advantage in using greater than critical damping.

With enclosed cabinet loudspeakers as a result of the effect on frequency response, the value of Q should not fall below 1, and the most desirable all-round condition appears to be with Q from 1.0 to 1.5—see Sect. 3(iii).

Alternatively, if the impedance is carefully measured and the impedance curve plotted over the region of the bass resonant frequency, the frequencies of the two half-power points may be noted, and the Q of the loudspeaker may be computed from the relation

$$Q \approx \left(\frac{f_0}{2\Delta f}\right)\left(\frac{R_0 + R_G}{Z_m + R_G}\right) \tag{6}$$

where f_0 = bass resonant frequency
$2\Delta f$ = band width at half power point (3 db below maximum impedance)
R_0 = voice coil resistance
Z_m = maximum value of the impedance (i.e. at f_0)
and R_G = effective plate resistance, referred to the secondary.

The measurements of frequency during this test are somewhat critical. The article by Preisman (Ref. 146) gives further details, although it does not derive eqn. (6) in this form.

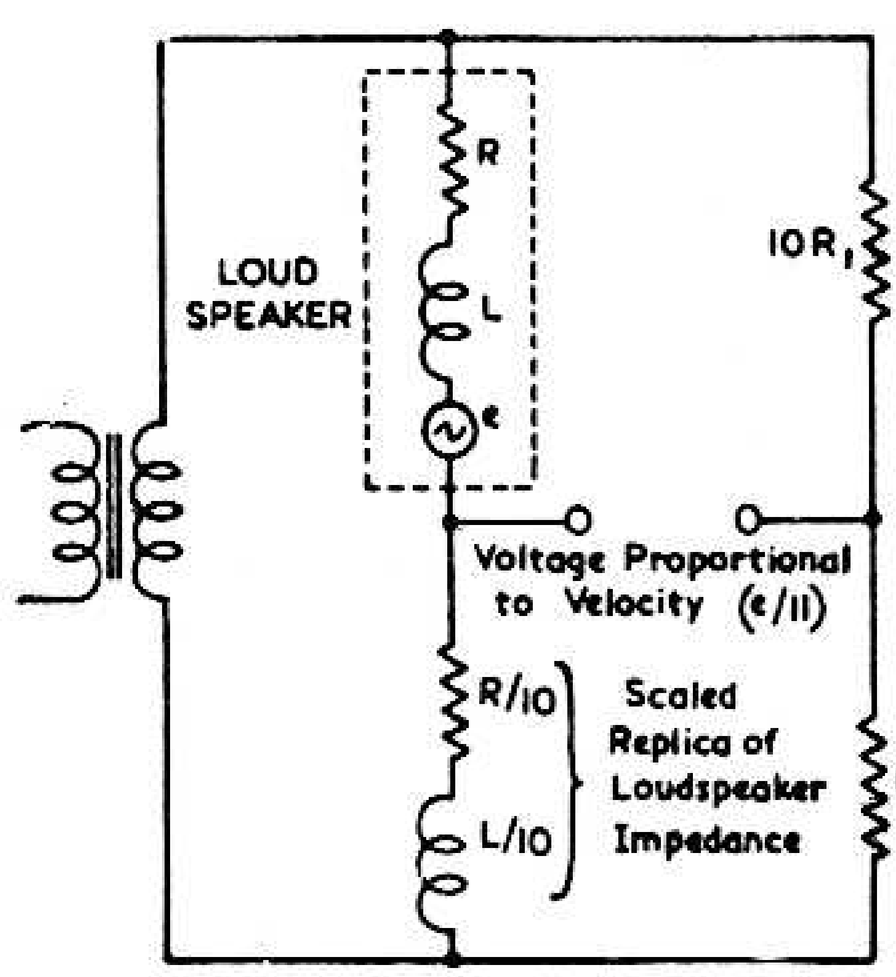


FIG. 20.9

Fig. 20.9. *Bridge circuit to give a voltage proportional to the velocity of the cone, to be fed back to the input.* (*Ref.* 184).

The majority of direct radiator loudspeakers have values of Q between 8 and 18 with a high valve plate resistance, although this may be decreased very considerably with a low effective plate resistance. Tests have been carried out to demonstrate the effect of a low plate resistance on the damping of a loudspeaker at the bass resonant frequency (Refs. 13A Fig. 6.30, 126, 132).

While there is no doubt that it is desirable to achieve a near approach to critical damping, the results are very much masked by the effects of the listening room which behaves as a resonant structure. In one case a room showed 8 resonances below 100 c/s, with Q values of the order of 12 to 15, and all above 8 (Ref. 155).

There are some loudspeakers in which the efficiency is too low to permit sufficient damping by reduction of the effective plate resistance. In these cases improvement in the electrical damping is possible by the use of positive current feedback combined with negative voltage feedback which can make the effective plate resistance zero or negative (Refs. 157, 158, 200, 201).

Alternatively, the e.m.f. generated by the movement of the cone may be fed back degeneratively to the input of the amplifier. One method of achieving this result is to arrange a special bridge circuit (Fig. 20.9) to provide the voltage to be fed back. The amplifier must have good characteristics with very low phase shift. This has been used satisfactorily over a frequency range from 10 to 1000 c/s, using a standard 12 inch loudspeaker, whose coil velocity was made proportional to input amplitude and completely independent of frequency (Refs. 184, 186, 148B, 191, 198).

Another method is to wind a separate feedback coil of very fine wire over the existing voice coil in a conventional loudspeaker. The voltage induced in this coil by the motion of the voice coil is a pure motional voltage at most frequencies. This voltage may be used as a feedback voltage to increase the damping and to reduce distortion arising from non-linearity of the cone suspension and from fringing of the magnetic field..

At very high audio frequencies the mutual inductance between the driving voice coil and the feedback coil produces in the feedback voltage a component which is dependent on the induction between the coils rather than on the motion alone. This difficulty is overcome by incorporating additional mutual inductance equal in value but of opposite sign, between the voice coil and feedback circuits at a point external to the magnetic field (Ref. 148).

SECTION 3: BAFFLES AND ENCLOSURES FOR DIRECT-RADIATOR LOUDSPEAKERS

(i) Flat baffles (ii) Open back cabinets (iii) Enclosed cabinet loudspeakers (iv) Acoustical phase inverter (" vented baffle ") (v) Acoustical labyrinth loudspeakers (vi) The R-J loudspeaker (vii) Design of exterior of cabinet.

(i) Flat baffles

The baffle is intended to prevent the escape of air pressure from the front to the back of the cone, which is out of phase at low frequencies. In many theoretical calculations an infinite flat baffle is assumed, but in practice a baffle is only made just large enough to produce the desired results. When the air distance around the baffle from front to back of the cone is equal to the wavelength, there is a dip in the response. For this reason, baffles are frequently made of irregular shape or the loudspeaker is not mounted in the centre, the object being to produce air distances from front to back of the cone which vary in a ratio at least 5 : 3. For example, with a 4 ft. square baffle and a 10 in. cone, the centre of the cone may be 18 inches from each of two adjacent sides.

The size of a square baffle, with the loudspeaker mounted off-centre, for desired bass response is given approximately by the following table for a 10 inch diameter cone (Ref. 13A) :

Minimum frequency	300	170	80	c/s
Side of square	2	4	8	feet

Below this minimum frequency, and below the loudspeaker bass resonant frequency, the response falls off at the rate of 18 db/octave.

The best flat baffle is provided by mounting the loudspeaker in a hole in the wall of a room, with the rear of the cone radiating into another room.

In a very large flat baffle, a loudspeaker which has a sufficiently high Q will have a peak in the response characteristic at, or somewhat above, the bass resonant frequency; below the peak frequency the attenuation is at the rate of 12 db/octave. Refs. 13, 13A, 14, 15 (Part 2), 75, 147, 151, 177.

(ii) Open back cabinets

An open back cabinet, as commonly used with radio receivers of the console, table or mantel type, has a resonance in the enclosure to the rear of the cone. This resonance causes a peak in the response characteristic at a frequency which is mainly a function of the cabinet although also influenced by the loudspeaker characteristics. For example, a cabinet 2 feet $\times$ 2 feet $\times$ 8 inches depth gave peak response at 180 c/s with a loudspeaker having a bass resonant frequency of 20 c/s, while the same cabinet gave a peak response at 110 c/s with a loudspeaker having a bass resonant frequency of 100 c/s. (Ref. 13A, Figs. 6.19B and 6.20B). The height of the peak is about 3 to 6 db for shallow cabinets or about 6 to 10 db for deep cabinets.

Open back cabinets are undesirable for good fidelity. If unavoidable, they should be as shallow as possible, with the minimum of acoustical obstruction, particularly at the back of the cabinet. Open back cabinets should be placed at least 6 inches out from the wall.

References to open back cabinets : 13A, 14, 147, 151, 188.

(iii) Enclosed cabinet loudspeakers

An enclosed loudspeaker is one which is totally enclosed so that there can be no interference between the front and back of the cone. There is no critical value for the volume of the enclosure but a large volume is desirable because it reduces the rise in resonant frequency above that on a large flat baffle. The increase in resonant frequency for Jensen speakers is shown in Fig. 20.10 and may be taken as fairly typical There is practically no non-linear distortion caused by the suspension, since this only contributes a small part of the total stiffness.

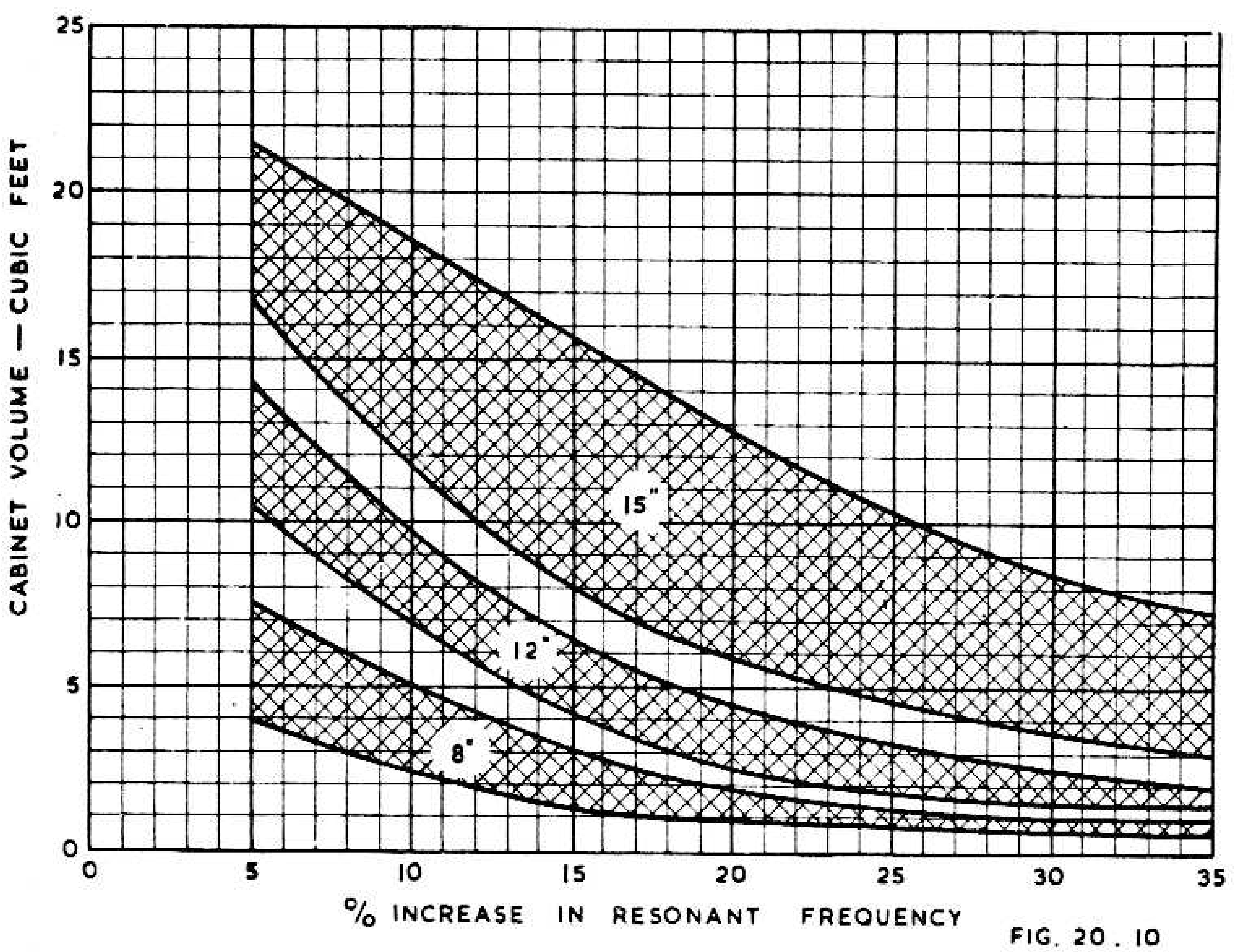


Fig. 20.10. *Increase in resonant frequency of totally enclosed cabinet loudspeaker versus cabinet volume. For each nominal speaker size, the upper limit of the shaded area corresponds to speakers with highest compliance, while the lower limit corresponds to those with least compliance.* (*Ref.* 151).

The enclosure should be adequately damped by at least ½ inch thickness of damping material all over (or double this thickness on one of each pair of parallel surfaces), although this is insufficient to eliminate standing waves at low frequencies.

The ineffectiveness of sound absorbent linings at low frequencies is due to the fact that all the absorbent material lies within a small fraction of a wavelength of the inner wall, which is a velocity node. Without motion of the air particles, or of the material itself, acoustical power cannot be absorbed and dissipated. Shorter (Ref. 135) has examined the problem of standing waves, and puts forward the method of damping by partitions, concentrating the absorbing material (¼ inch carpet felt) into one or two partitions strategically placed across the cabinet. Those sections of the cabinet which are separated from the cone by one or more sound absorbing partitions receive little sound at the high frequencies and therefore require very little acoustical treatment. The loudspeaker back e.m.f. may be measured by a bridge circuit and should show a single peak without subsidiary peaks over the range from 30 to 200 c/s.

It is desirable for the enclosure to have appreciably different values of linear dimensions—typical ratios are 1 : 1.5 : 2.7. However, the longest dimension should be less than ¼ of the wavelength at the lowest working frequency. The enclosure should have rigid walls—timber at least ½ inch thick, and braced where necessary.

One limitation of the enclosed cabinet loudspeaker is the bass attenuation which occurs with critical damping, although this only happens with high flux density and low amplifier output resistance—see below under Equivalent Circuit.

Some loudspeakers have been designed specially for use with enclosed cabinets. These usually have very low bass resonant frequencies, when measured on a flat baffle, so that when loaded by the enclosure, the resonant frequency is still sufficiently low.

The volume of the cabinet is given below for some examples :

Western Electric 755A	8 in.	70 c/s	2	cubic feet
" " 756A	10 in.	65 c/s	2½	" "
" " 754A	12 in.	60 c/s	3	" "
Goodmans "Axiom 80"	9½ in.*		3½	" "
R.C.A. (Ref. 84)	peak at 80 c/s		1.5	" "
Stromberg-Carlson	8 in.		1.7	" "
	12 in.		3.9	" "

Equivalent circuit

Fig. 20.11 is a simplified electrical equivalent circuit of the acoustical system of the loudspeaker (Ref. 135).

L_U represents cone mass + effect of radiation reactance.
R_U represents radiation resistance (which varies with frequency)†.
C_C represents acoustical capacitance of cabinet volume.
C_U represents equivalent capacitance of cone suspension.
R_S represents effect of electrical circuit of loudspeaker and driving amplifier reflected into acoustical circuit. The mechanical resistance of the cone suspension may be taken as being included with R_S.
E_S = constant voltage generator.
I_U = alternating air current produced by cone, which is proportional to cone velocity.

Now $R_S \propto \dfrac{B^2}{R_G + R_0}$ at low frequencies

where B = flux density in gap,
R_G = output impedance of amplifier referred to voice coil circuit,
and R_0 = resistance of voice coil.

The acoustical response of the loudspeaker is proportional to $I_U f$.

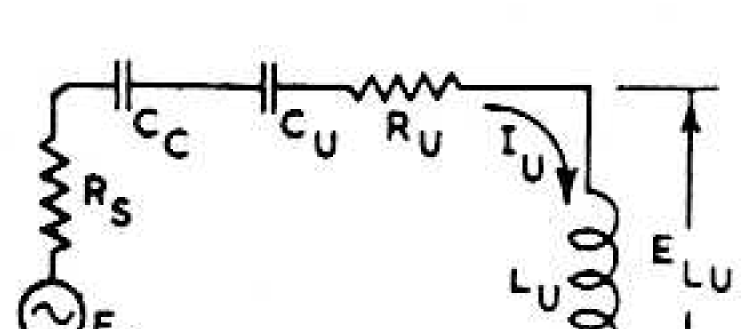


Fig. 20.11. *Equivalent electrical circuit of acoustical elements in an enclosed cabinet loudspeaker at low frequencies.* (*Ref.* 135).

The voltage $E_{LU} = 2\pi f L_U I_U \propto I_U f \propto$ acoustical response. Therefore the variation with frequency of E_{LU}, for any one value of L_U, gives the frequency response of the loudspeaker. Over the frequency range for which the equivalent circuit is valid, and the wavelength is large compared with the size of the cone, the loudspeaker can be reduced—so far as frequency response is concerned—to a half-section high-pass filter working into open circuit. Small values of R_S, resulting from low flux density or high amplifier output impedance, give a resonance peak and bad transient response, while large values of R_S, corresponding to high flux density and low amplifier output impedance, can give a serious loss of bass.

Fig. 20.12 shows curves for three values of R_S and hence for three values of Q, for a resonant frequency of 45 c/s. This indicates that values of Q less than about 1 result in serious attenuation of low frequencies.

*Resonance 17 c/s on flat baffle.

†Note that R_U is small compared with other impedances in the circuit.

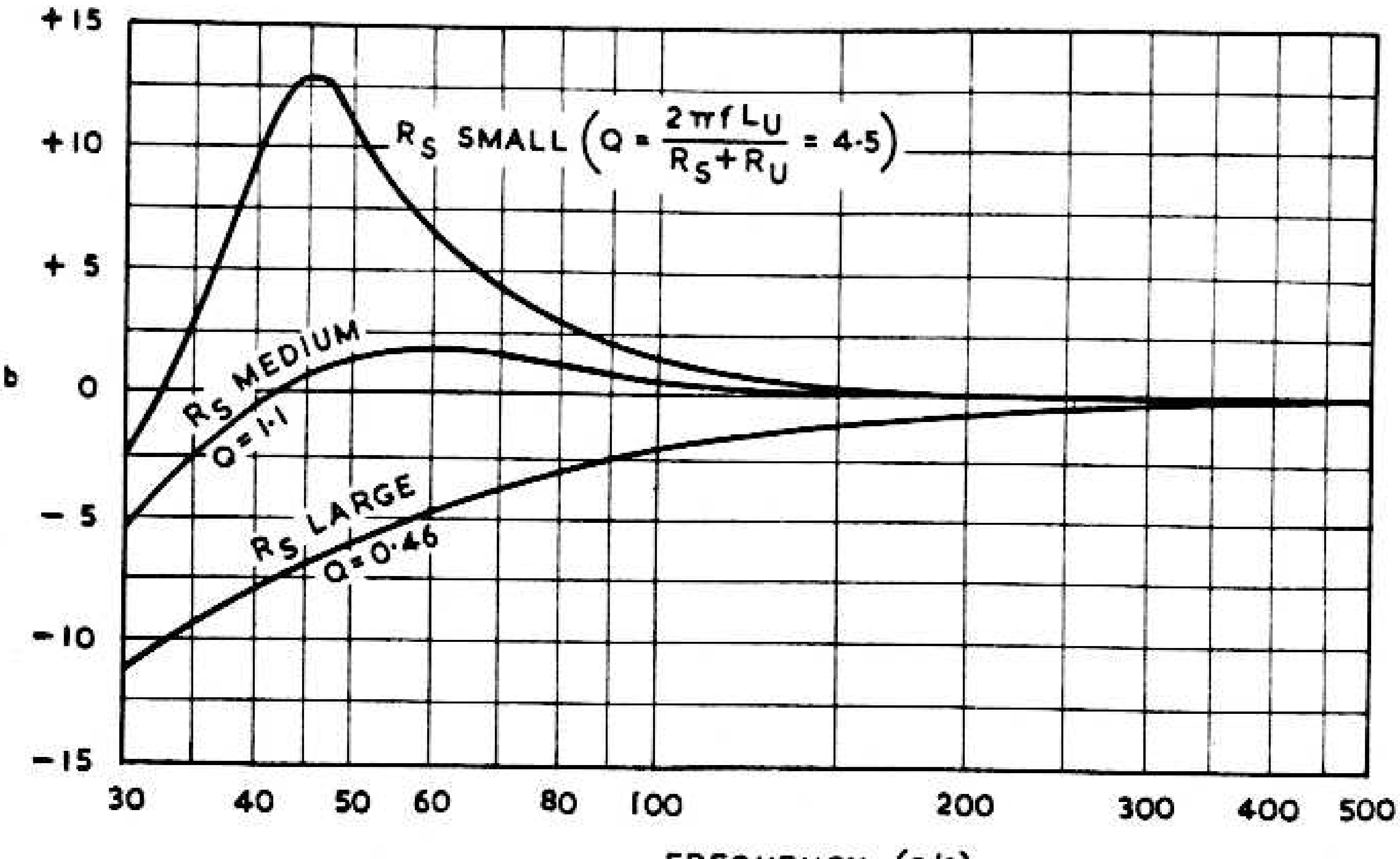


FIG. 20.12

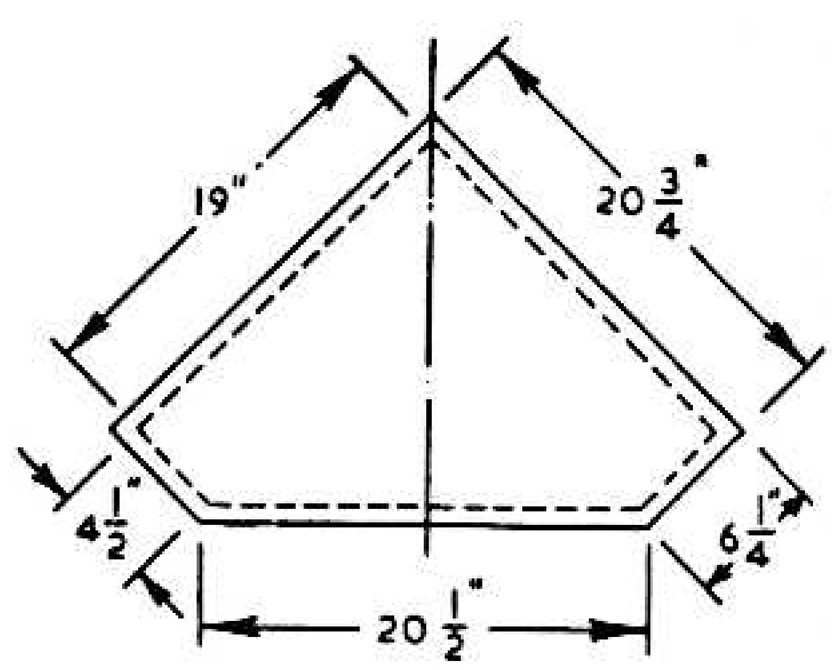
Fig. 20.12. *Theoretical response of enclosed cabinet loudspeaker for various values of the effective damping resistance R_s. Resonance frequency* 45 *c/s.* (*Ref.* 135).

References to enclosed cabinets: 13A, 29, 36, 80, 81, 84, 116, 116A, 135, 147, 151, 166, 168, 188.

(iv) Acoustical phase inverter (" vented baffle ")

Also known as a bass reflex baffle.

This has a vent* or duct in the front of the cabinet which augments the direct radiation from the cone at low frequencies (Fig. 20.13). The box should be at least partially lined with sound absorbent material to absorb the higher frequencies, but should not be too heavily damped at frequencies below about 150 c/s. The cabinet should be strongly made of heavy timber with adequate bracing to prevent vibration. Many cabinets which have been built for this purpose are partially ineffective as the



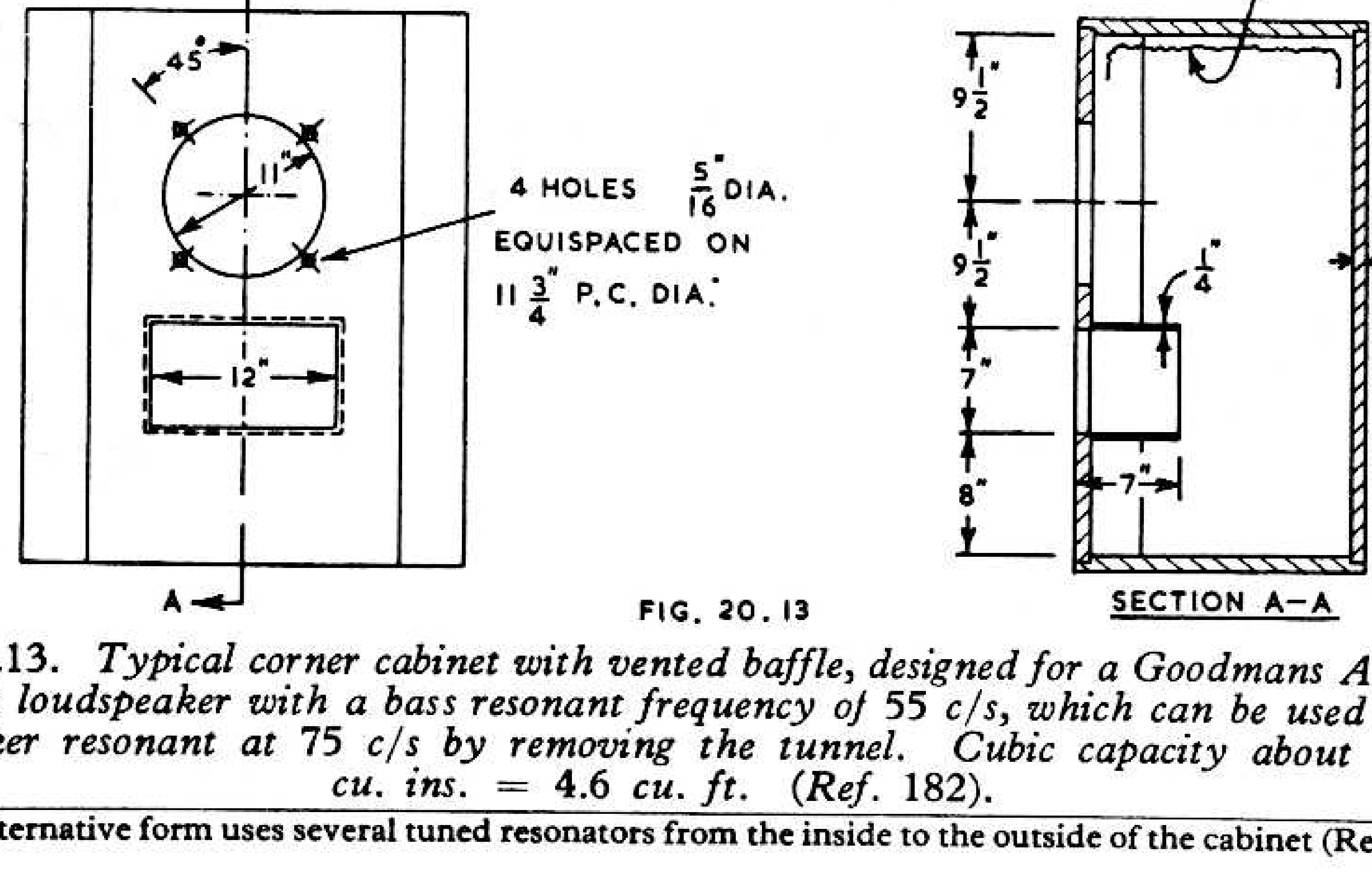


FIG. 20.13

Fig. 20.13. *Typical corner cabinet with vented baffle, designed for a Goodmans Axiom* 12 *inch loudspeaker with a bass resonant frequency of* 55 *c/s, which can be used with a speaker resonant at* 75 *c/s by removing the tunnel. Cubic capacity about* 8000 *cu. ins.* = 4.6 *cu. ft.* (*Ref.* 182).

*An alternative form uses several tuned resonators from the inside to the outside of the cabinet (Ref. 18).

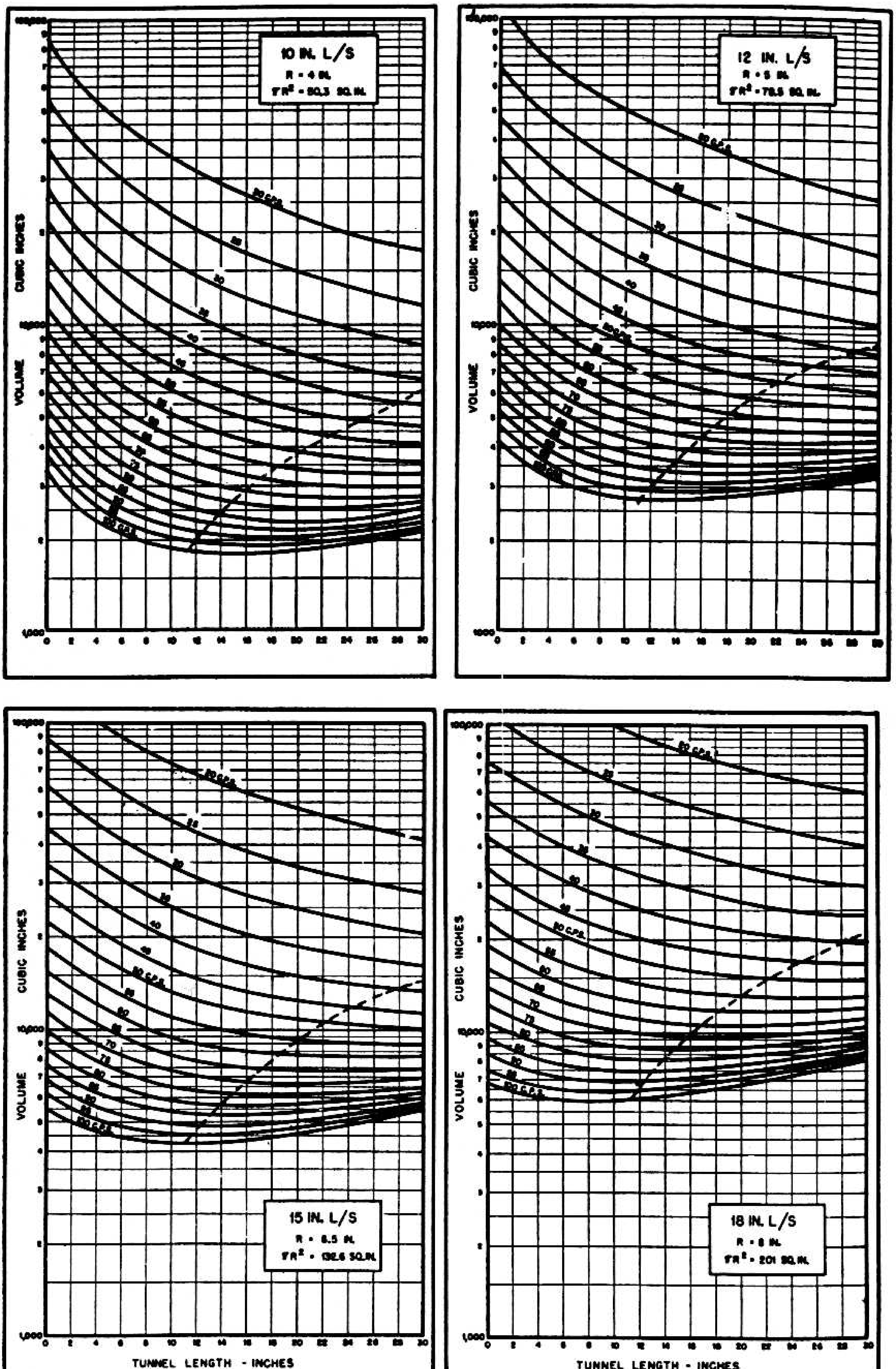


FIG. 20.14

Fig. 20.14. *Curves showing tunnel length for* 10, 12, 15 *and* 18 *inch Goodmans loudspeakers with bass resonant frequencies from* 20 *to* 100 *c/s. The tunnel length should not exceed* 1/12 *of the wavelength of the speaker bass resonant frequency* ; *points to the right of the broken lines should not be used* (*Ref.* 182).

result of wall vibration under the extremely arduous conditions of operation. Other materials which have been used satisfactorily include brick and concrete.

Vented baffles are normally designed so that the loudspeaker bass resonant frequency is matched to the acoustical resonant frequency of the cabinet ; unmatched combinations are, however, also used.

(A) Matched vented baffles

The merits of the vented baffle, when correctly applied, include

(1) improved bass response,
(2) much reduced amplitude of movement of the cone at the resonant frequency, and hence reduced distortion at the same power output or increased power handling capacity for the same distortion,
(3) decreased peak electrical impedance,
(4) increased acoustical damping over a limited range of low frequencies (not including the vent resonance),
(5) increased radiation resistance and decreased reactance of the loudspeaker at low frequencies.

The vented baffle appears to be preferred by most engineers to the acoustical labyrinth, even apart from cost.

The vent area should normally be the same as the effective radiating area of the cone, and the curves in Fig. 20.14 are based on this relationship. Thus the cross-sectional area of the tunnel is given by πR^2 where R is the radius of the piston equivalent to the speaker cone at low frequencies. The tunnel may have any length from the thickness of the timber (i.e. when no tunnel, as such, is constructed) to a maximum of approximately one twelfth wavelength, indicated by the broken curves in Fig. 20.14 ; consequently the area to the right of the broken curves should not be used.

A reasonable length of tunnel enables a smaller enclosure to be used, but the space between the end of the tunnel and the rear of the cabinet should not be less than R. The volume occupied by the loudspeaker must be added to the volume derived from the curves to obtain the volume of the enclosure. If the volume of the loudspeaker is not known, the following approximation may be used :

Volume displaced by loudspeaker :

Loudspeaker dia.	8	10	12	14	16	18	ins.
Volume (approx.)	250	400	650	1000	1600	2300	cu. ins.

Also add volume displaced by internal timber bracing.

Note that no allowance should be made for the thickness of the damping material.

The curves for the volume of the cabinet are derived from the relationship

$$V = \pi R^2 \left[\frac{1.84 \times 10^8}{\omega^2} \times \frac{1}{1.7R + L} + L \right] \quad (1)$$

where V = volume in cubic inches,
R = radius of equivalent piston in inches,
L = length of tunnel in inches
and $\omega = 2\pi \times$ frequency of vent resonance.

When correctly designed a vented baffle loudspeaker has two impedance peaks, one above and one below the vent resonance, while the same loudspeaker on a flat baffle has only one impedance peak—see Fig. 20.15. In all cases where optimum performance is desired, the electrical impedance characteristic below 150 c/s should be measured using a constant current source and a voltmeter. The ratio of the two peak frequencies should not exceed about 2.4 : 1 nor be less than about 1.5 : 1 and any subsidiary peaks caused by standing waves should be removed by the addition of damping material (as for the enclosed cabinet above ; see also Fig. 20.17).

Tests for acoustical output should not be made until the cabinet dimensions and damping have been adjusted to give the correct impedance characteristic. In general, with a vented baffle the bass range over an octave or more is increased by several decibels over either a very large baffle or an enclosed cabinet loudspeaker.

The power gain may be greater than 3 db since the air loading of the cone may be increased appreciably over the lower octave of the frequency range. For about one third of an octave above and below the resonant frequency of the system, the greater

FIG. 20.15

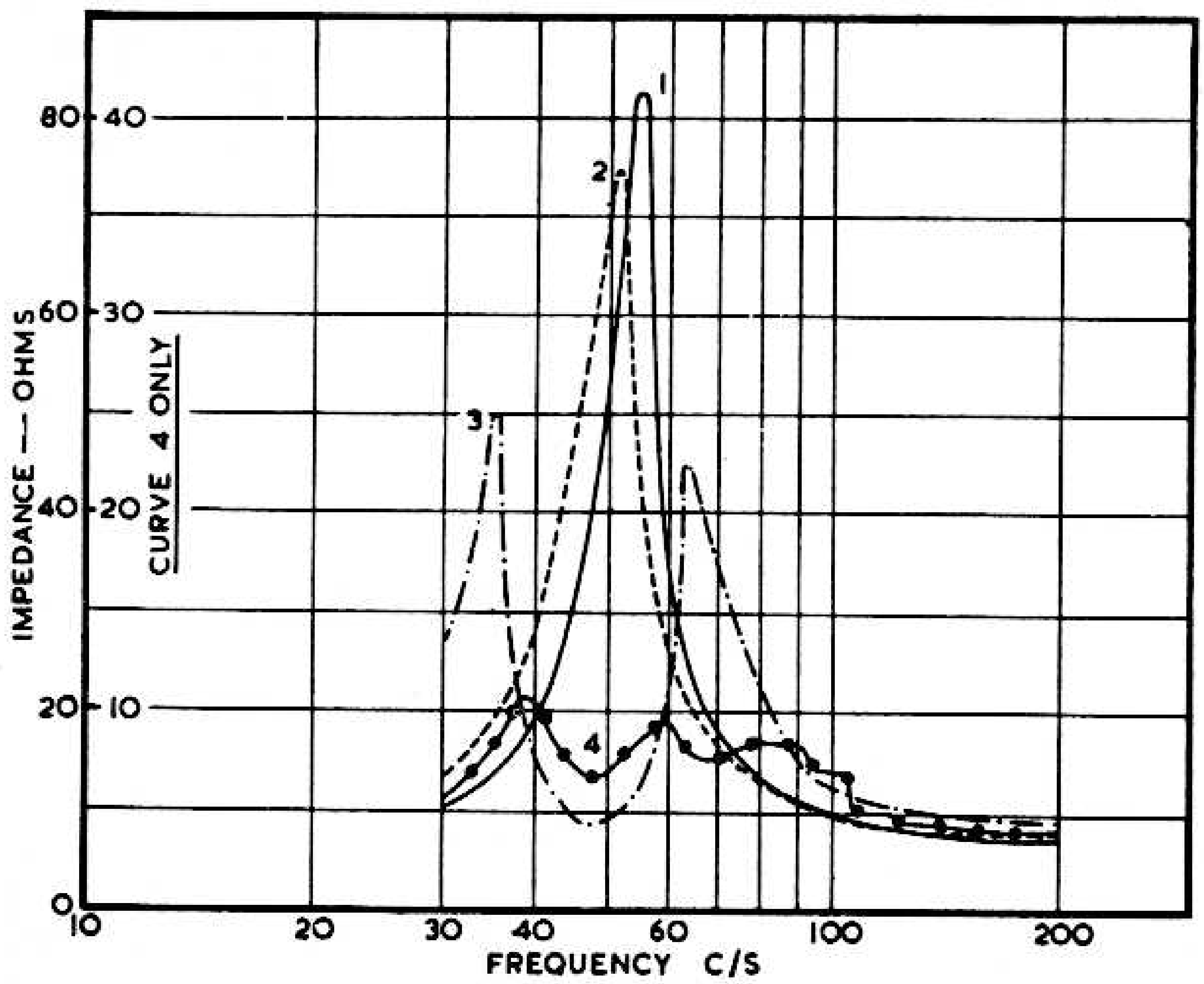


Fig. 20.15. *Impedance versus frequency characteristics of a* 12 *inch loudspeaker* (1) *in free air* (2) *on a flat baffle and* (3) *in a vented baffle. Curve* (4) *shows two "staggered" vented baffle loudspeakers in parallel (measured).*

part of the energy is radiated by the port. Phase shift occurs suddenly at the resonant frequency, so that the radiation below this frequency is reduced, being the vector difference between the two sources.

At the resonant frequency, the air in the vent or tunnel moves vigorously while the acoustical impedance on the rear of the cone is resistive and reaches a maximum.

At the frequency of the vent resonance, the motion of the speech coil is so small that no appreciable electrical damping can take place and there is no danger of bass attenuation such as may occur with the enclosed cabinet or flat baffle. The energy dissipated by mechanical resistances and by the radiation of sound is generally small, so that the vent resonance on which the maintenance of the bass response depends, is quite lightly damped. For this reason, when good fidelity is required, the vent resonance frequency should not be greater than 60 c/s. (Ref. 135).

A corner speaker cabinet for 15 inch cones is described in Ref. 100, while one for 12 inch cones is described in Ref. 105. These both employ a vented baffle arrangement with the vents on both sides of the cabinet, between the cabinet and the walls.

Equivalent circuit (Ref. 135)

The Fig. 20.16 is a simplified electrical equivalent circuit of the acoustical system and is based on Fig. 20.11 with the addition of R_V representing the radiation resistance of the vent, and L_V, the acoustical inductance of the vent plus the effects of its radiation reactance. Let f_V be the frequency of the vent resonance (i.e. the resonance of L_V and C_C) and assume that the cone resonant frequency is the same. Below f_V, the series combination $L_U C_U$ appears as a capacitance and the parallel combination $L_V C_C$ as an inductance, and the overall effect is to produce a series resonance at some frequency f_1 in this region. At frequencies above f_V, $L_U C_U$ appears inductive and $L_V C_C$ capacitive, and a second series resonance appears at some frequency f_2. The resonances at f_1 and f_2 are responsible for the characteristic double hump in the electrical impedance frequency characteristic of a vented baffle and are subject to the circuit damping represented by R_S.

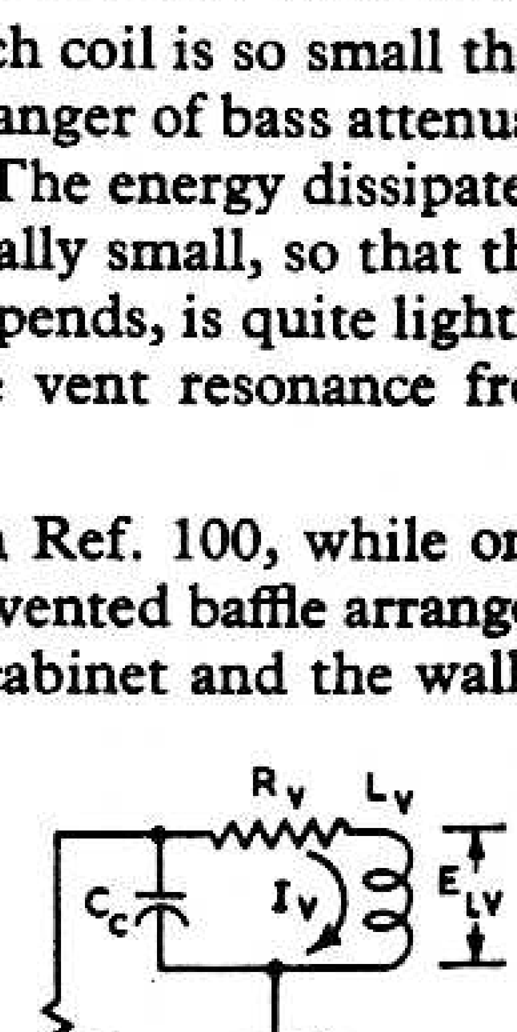


FIG. 20.16

Fig. 20.16. *Equivalent electrical circuit of vented baffle loud-speaker at low frequencies.* (*Ref.* 135).

I_U and I_V represent the alternating air current flow in the cone and vent respectively. I_U originates at the back of the cone and I_V must therefore be reversed in phase with respect to I_U if the acoustical output of the vent is to reinforce that from the front of the cone. At f_V, the frequency of the vent resonance, I_U and I_V are approximately in quadrature ; above f_V the desired reinforcing condition is approached, but below f_V the two air currents oppose one another.

The performance of the system is determined by the value of f_V, which is normally made equal to the loudspeaker bass resonant frequency, and also by the ratio L_V/C_C. With a large cabinet and large vent (C_C large, L_V small) the response can rise to a peak in the bass, while a small cabinet and a small vent (C_C small, L_V large) will give much the same effect as a completely closed cabinet, i.e. there may be bass loss.

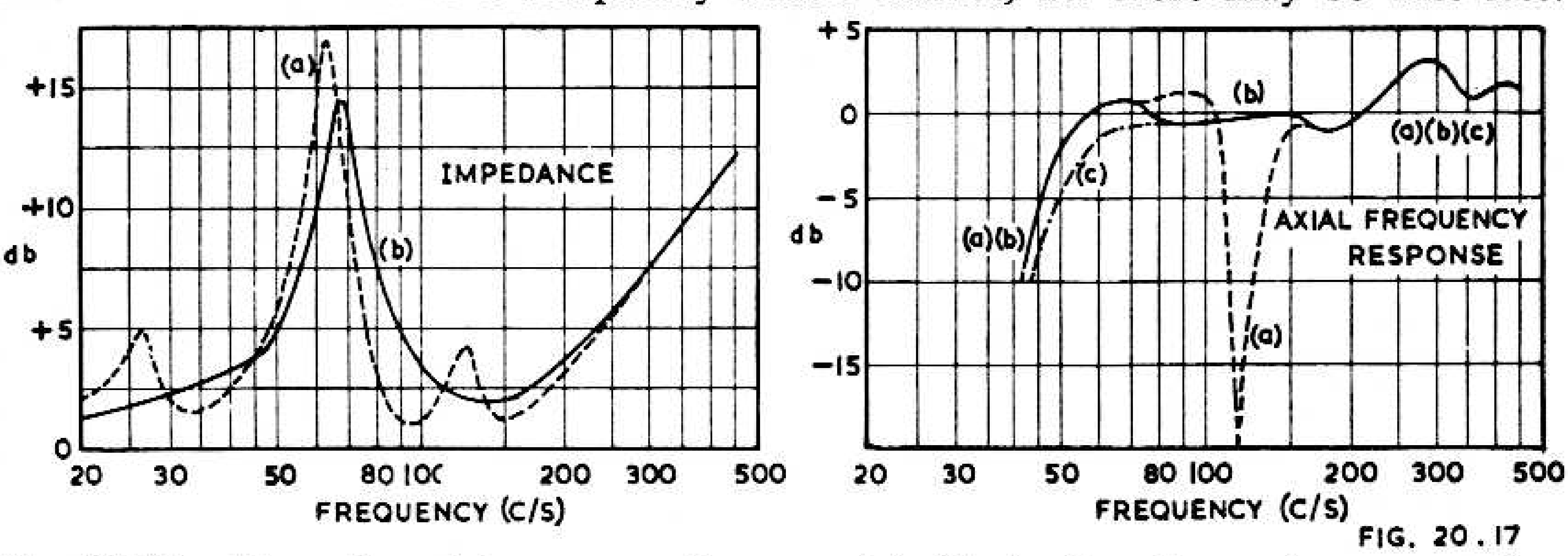


Fig. 20.17. Free-air axial response of a vented baffle loudspeaker and corresponding impedance characteristics (a) conventional damping (b) optimum damping (c) excessive damping of partition. (Ref. 135.)

Between these two extremes, there will be a pair of values for L_V and C_C which gives the best approximation to a flat frequency response. The cabinet volume required for this last condition depends on the efficiency of the loudspeaker unit and the output impedance of the amplifier. Fig. 20.17 shows the response and impedance characteristics of a 15 inch unit in a 10.5 cu. ft. cabinet. Curves (a) are for the cabinet lined with kapok quilt and carpet felt, with internal space left free ; a serious internal resonance shows at about 120 c/s. Curves (b) show how this resonance was suppressed by introducing into the cabinet a partition having a window of area about 170 sq. ins. covered with three layers of ¼ in. carpet felt. This damping does not seriously affect the frequency response but it suppresses the lower frequency impedance hump. Curve (c) shows the effect of too many layers of felt on the window. In this case the design aimed at a nearly level frequency characteristic down to 50 c/s.

See also Ref. 211 for other form of equivalent circuit.

(B) Unmatched vented baffles

Unmatched vented baffles preferably incorporate a loudspeaker having a very low bass resonant frequency. The cabinet is designed quite independently of the loudspeaker and may, if desired, be fitted with an adjustable vent. In the latter case the maximum vent area may be somewhat greater than the loudspeaker cone area, while in the other extreme the vent may be completely closed. Using a cabinet 2 ft. × 2 ft. × 18 ins. (volume 6 cu. ft.) and a loudspeaker with a bass resonant frequency of 30 c/s, the frequency of maximum acoustical pressure increases from 40 to 60 thence to 75 c/s as the vent area is varied from small to medium to large respectively. The height of the maximum response above that at 400 c/s rises from 4 db (small vent) to 6 db (medium vent) thence to 8 db (large vent) (Ref. 13A).

In other cases a loudspeaker is used which has a bass resonant frequency of conventional value, but here also the vent frequency is higher than the loudspeaker bass resonant frequency. This permits the use of a smaller cabinet while still retaining some of the good features of a matched vented baffle, although it is not so good on transients. In all cases an adjustable vent area is desirable. Tunnels are not generally used, since they give no flexibility of adjustment.

The vent frequency should not exceed (say) 90 c/s, otherwise the tone will be poor. This limits the reduction in cabinet volume to about 2½ cubic feet.

The shape of the cabinet, as well as its volume, affect its frequency response and impedance characteristics. For example, a change from 30 ins. × 15 ins. × 8 ins. to an equivalent volume 15 ins. × 15 ins. × 16 ins. caused an increase from 45 to 58 c/s in the lower impedance peak and from 75 to 110 c/s in the higher peak (Ref. 117). In the cubic shape of cabinet the addition of an internal partition between loudspeaker and vent, extending approximately half-way from front to back, causes a reduction in the frequencies of both impedance peaks, and is claimed to give improved results on both speech and music (Ref. 117).

Careful attention to damping, along the lines given above for both enclosed cabinets and matched vented baffles, would be well repaid.

(c) **Special types of vented baffle loudspeakers**—see Supplement.

Refs. to vented baffles : 4, 9, 12, 13, 13A, 29, 36, 76 (Part 2), 85, 96, 113, 116, 116A, 117, 118, 135, 143, 144, 147, 150, 151, 166, 168, 175, 182, 188, 202, 209, 211, 225, 229, 230.

(v) Acoustical labyrinth* loudspeaker

The acoustical labyrinth gives a performance somewhat similar to that of a vented baffle. The rear of the cone drives a long folded tube, lined with sound absorbing materials, the mouth of which opens in front of the cabinet (Fig. 20.18). The length of the tube is approximately 7 feet (measured on the centre-line) for nearly linear response down to 70 c/s. The loudspeaker bass resonance loaded by the labyrinth is preferably at a frequency at which the wavelength is four times the length of the tube ; in the example this is 40 c/s. If this latter condition is not fulfilled, the frequency response will not be linear. The loudspeaker resonance frequency in the example was reduced from 50 to 40 c/s by the loading of the labyrinth. This is the only form of baffle which reduces the bass resonant frequency of a loudspeaker. The rise of impedance from 400 c/s down to the bass resonant frequency is reduced considerably by the acoustical labyrinth—in one case the ratio was reduced from 10 : 1 to 4.3 : 1 (Ref. 27).

References 13, 14, 27, 28, 188, 204, 225 Part 2.

FIG. 20.18

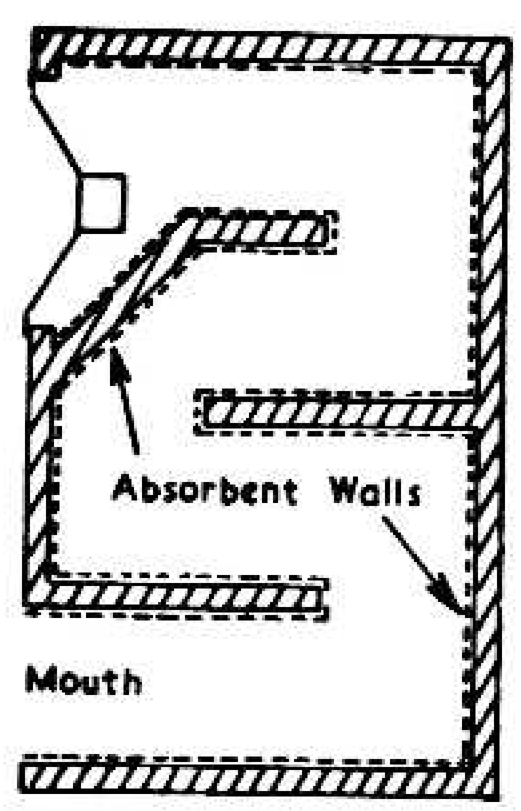


Fig. 20.18. *Sectional view of acoustical labyrinth loudspeaker.*

(vi) The R-J loudspeaker

The R-J loudspeaker has a particularly compact bass unit, to which any desired tweeter can be added. In one design a 15 inch woofer unit is mounted in a cube with 18 inch sides and fundamental bass reproduction is claimed down to 20 c/s. Only the forward radiation is used, but both sides of the cone are loaded. The back of the speaker is completely enclosed within a small stiff cavity. The front of the woofer works into a carefully designed rectangular duct, and the sound issues from a slot extending across the base of the enclosure.

Refs. 189, 190, 219, 234. See also Supplement.

(vii) Design of exterior of cabinet

The sharp corners on the usual more-or-less box shaped cabinet, particularly those of the side in which the loudspeaker is mounted, produce diffraction effects causing a sequence of peaks and valleys up to ± 5 db in the response characteristic. The best shaped enclosure is a complete sphere, while the worst is a cube with the loud-speaker in the centre of one side. A rectangular parallelepiped (box shaped) with the loudspeaker closer to one of the short sides than to the other is an im-

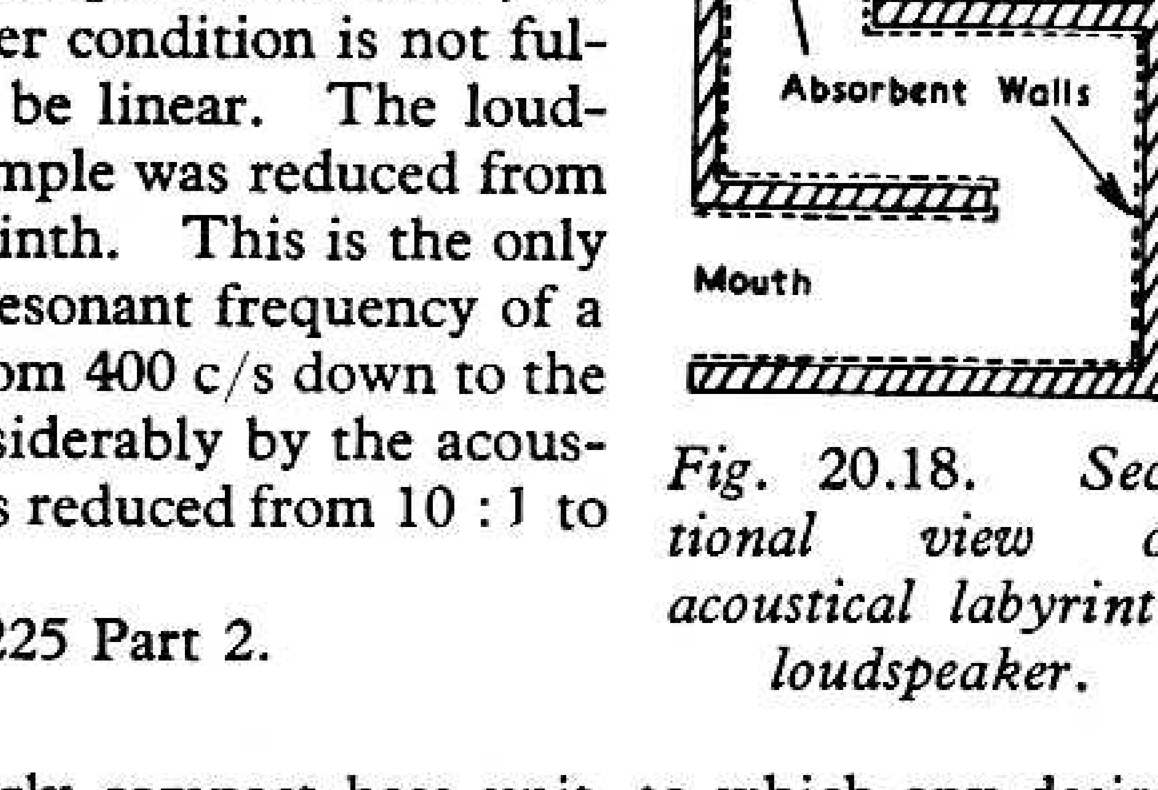


FIG. 20 19

Fig. 20.19. *Loudspeaker enclosure to minimize diffraction effects* (*after Ref.* 193).

*"Acoustical Labyrinth" is a registered trademark of Stromberg-Carlson Co.

provement over the symmetrical cube, but still far from the ideal. A very close approach to the ideal is given by a rectangular truncated pyramid mounted on a rectangular parallelepiped (Fig. 20.19) Ref. 193.

In all cases it is desirable that the edge of the cone should be flush with the front of the cabinet.

SECTION 4 : HORN LOUDSPEAKERS

(i) Introduction (ii) Conical horns (iii) Exponential horns (iv) Hyperbolic exponential horns (v) Horn loudspeakers—general (vi) Folded horn loudspeakers (vii) High frequency horns (viii) Combination horn and phase inverter loudspeakers for personal radio receivers (ix) Materials for making horns.

(i) Introduction

A horn is used in conjunction with a diaphragm or cone loudspeaker for the purposes of increasing the acoustical loading on the diaphragm (over a limited frequency range) and thereby increasing the efficiency and reducing non-linear distortion. With a horn, inside the useful frequency limits, the movement of the diaphragm is much less and the acoustical damping is much greater than on a flat baffle, for the same acoustical power output. Thus with a horn, a smaller diaphragm can radiate a given acoustical power.

A horn is essentially a device which transforms acoustical energy at high pressure and low velocity to energy at low pressure and high velocity.

Horns are of various shapes—conical, parabolic, hypex and exponential, but the exponential is most widely used.

A well designed and well executed exponential or hypex horn loudspeaker is capable of giving a flatter frequency output characteristic with less distortion than any other form of loudspeaker.

References 13A, 14, 15, 59, 82, 142, 177, 188.

(ii) Conical horns

A conical horn is one having straight sides ; it functions in a manner similar to an exponential horn but its throat resistance is less than that of an equivalent exponential horn, except at high frequencies. Conical horns are sometimes used with cone type loudspeakers to give directional characteristics ; the angle of propagation is then approximately equal to the angle of the horn.

Conical horns are sometimes used, for economy, as the first section at the throat end, with an exponential horn forming a second section at the mouth end. This device is frequently used with folded horns. Sometimes an approach to an exponential horn is made with a number of conical sections, each with a different degree of taper. All such compromises lead to inferior performance.

References 13A, 14, 15, 142, 175.

(iii) Exponential horns

In an exponential horn, the cross-sectional area (S) at any point distant x feet along the axis is given by

$$S = S_0 \epsilon^{mx} \tag{1}$$

where S_0 = cross-sectional area at throat (sq. feet)

ϵ = 2.71828 (Naperian base)

and m = flaring constant.

Equation (1) may be put into the form

$$S/S_0 = \epsilon^{mx}$$

Therefore $mx = \log_\epsilon (S/S_0) = \log_{10} (S/S_0) \times 2.3026$.

If $(S/S_0) = 2$ then $mx = 0.3010 \times 2.3026 = 0.6931$.

If $(S/S_0) = 4$ then $mx = 0.6020 \times 2.3026 = 2 \times 0.6931$.
We may therefore deduce that

(1) the cross-sectional area doubles itself each time the distance along the axis is increased by $0.6931/m$.
(2) the value of m (in inverse feet) is equal to 0.6931 divided by the distance along the axis in feet for the cross-sectional area to double itself.

This is shown more clearly below where $l' = 0.6931/m$:

Distance along axis	Cross-sectional area
$x = 0$ (at throat)	S_0
$x = l'$	$S = 2S_0$
$x = 2l'$	$S = 4S_0$
$x = 3l'$	$S = 8S_0$ etc.

The corresponding diameter and side of circular and square cross sections are :

Distance along axis	Diameter	Side of square
$x = 0$ (at throat)	d_0	a_0
$x = l'$	$1.414d_0$	$1.414a_0$
$x = 2l'$	$2d_0$	$2a_0$
$x = 3l'$	$2.818d_0$	$2.818a_0$
$x = 4l'$	$4d_0$	$4a_0$

The flare cut-off frequency for an infinite exponential horn is given by

$$f_0 = m \times 89.5 \tag{2a}$$

while the corresponding cut-off wavelength is

$$\lambda_0 = 12.6/m \tag{2b}$$

where m is expressed in inverse feet.

FIG. 20.20

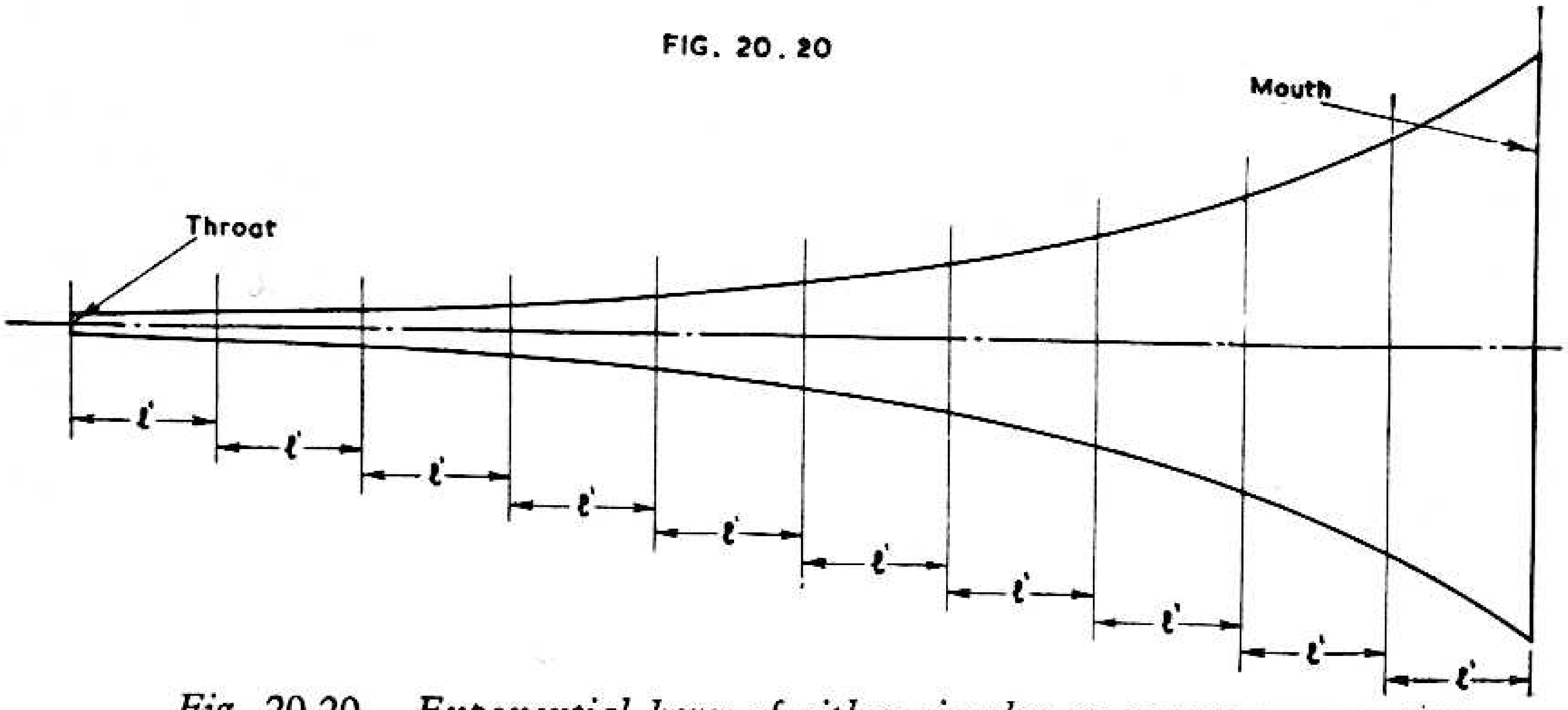


Fig. 20.20. Exponential horn of either circular or square cross-section.

Design on basis of minimum useful frequency

The characteristics below are given as a function of f', the lower frequency limit for satisfactory horn loading. The value of f' is arbitrarily taken as 1.2 times the flare cut-off frequency for an infinite exponential horn ; it is, of course, influenced also by the dimensions of the mouth and the length of the horn.

The value of m, the **flaring constant**, should then be equal to or less than $10.5/\lambda'$ (2c)
where λ' = wavelength (in feet) of the minimum useful frequency (f') for satisfactory horn loading.

Minimum useful frequency (f')	50	100	150	200	c/s
Maximum useful wavelength (λ')	22.6	11.3	7.52	5.65	feet
Maximum value of m	0.47	0.93	1.40	1.85	inverse feet

The **diameter of the mouth** should preferably not be less than one third of the maximum useful wavelength, while the effect of resonances at the low frequencies may be made negligible by increasing the mouth diameter to two thirds of the maximum useful wavelength (λ'). Taking the smaller mouth diameter, and referring to the minimum useful frequency:

Minimum circumference $= 1.05\lambda' = 1200/f'$ (3)

Equation (3) may be applied to any shape of mouth—see table below:

Circumference (min.)	24	12	8	6	feet
Maximum useful wavelength (λ')	22.6	11.3	7.52	5.65	feet
Minimum useful frequency (f')	50	100	150	200	c/s

Horns with circular cross-section

In this case the equation is

$d = d_0 \epsilon^{mx/2}$ (4)

where d = diameter in feet

and d_0 = throat diameter in feet.

Diameter at mouth: $d_m = d_0 \epsilon^{ml/2}$ (5)

or $d_m = d_0 \text{ antilog}_{10} (ml/4.605)$ (6)

Length: $l = (4.605/m) \log_{10} (d_m/d_0)$ (7)

For chart see Ref. 39 Chart III.

Flaring constant: $m = (4.605/l) \log_{10} (d_m/d_0)$ (8)

If the minimum useful frequency is taken as f' where $f' = 1.2 f_0$, and if the minimum diameter of mouth is taken as one third of the wavelength at frequency f', then m may have its maximum value of $10.5/\lambda'$ and

minimum diameter of mouth $= \lambda'/3 = 380/f'$ (9)

minimum length $= (500/f') \log_{10} (380/f'd_0)$ (10)

where λ' = wavelength in feet at frequency f'

$f' = 1.2 f_0 = 1.2 \times$ flare cut-off frequency

d_0 = diameter at throat in feet

and all dimensions are in feet.

Horns with square cross-section

These follow the same laws as horns of circular cross-section having diameters equal to the sides of the square, but the length and mouth dimensions are slightly less for the same minimum useful frequency f':—

minimum side of mouth $= 300/f'$ (11)

minimum length* $= (500/f_0) \log_{10} (300/f'a_0)$ (12)

where a_0 = length of side at throat.

For cutting the side of a square horn from a sheet, see Ref. 77.

General references to horn dimensions: 39, 46, 82.

References to theory of exponential horns: 13A, 142, 175, 188.

(iv) Hyperbolic exponential horns ("hypex")

A hyperbolic exponential horn is one which follows the law

$S = S_0[\cosh mx + T \sinh mx]^2$ (13)

where S = cross-sectional area at distance x along the axis in sq. ft. (or in any other convenient units)

S_0 = cross-sectional area at throat in sq. ft.

m = flaring constant

x = distance along the axis in feet

and T = shape parameter (T may have any value from zero to infinity).

*For chart see Ref. 46 Figs. 1 and 2

The cut-off frequency is the same as for an exponential horn having the same flaring constant—see eqn. (2a). When $T = 1$, the horn is exponential. When $T = 1/(mx_0)$ and m is allowed to go to zero, the horn is conical. When $T = 0$ the horn is hyperbolic cosine or catenoidal (Refs. 13A, 123, 140, 142, 188). The name " hypex " is usually applied when T is greater than zero and less than unity.

The value of T in " hypex " horns is usually between 0.5 and 0.7, and within these limits the throat resistance of an infinite horn is more nearly constant than that of an exponential horn, at frequencies slightly above the cut-off frequency. These comparisons are for constant throat, mouth and length of horn ; under these conditions a " hypex " horn with T of the order of 0.6 has improved low frequency characteristics as compared with those of an exponential horn. Consequently, for equivalent performance, a " hypex " horn may be made more compact than an exponential horn. A further useful feature of the " hypex " characteristic is that it makes possible a gradual transition from conical, via " hypex " with varying T, to exponential (Ref. 123).

An analysis of the response peaks in finite hyperbolic horns, and design procedure for horns to have peaks at pre-determined frequencies, is given in Ref. 139.

(v) Horn loudspeakers—general

(A) Frequency limitations

The resistive component of the throat impedance drops rapidly below about 1.2 times the flare cut-off frequency. Some output is obtained with finite horns at, and even below, the flare cut-off frequency (Ref. 45), but at frequencies below the flare cut-off frequency there is a strong tendency towards the production of harmonics. Care should therefore be taken to eliminate from the amplifier any appreciable output power below about 1.2 times the flare cut-off frequency, unless the loudspeaker is known to be capable of handling such frequencies without damage or distortion.

The resonant frequency of the diaphragm should not be less than the flare cut-off frequency, and preferably not less than 1.2 times this value, in order to ensure sufficient loading at the resonant frequency.

The only known method for handling frequencies below the flare cut-off frequency of an exponential horn, with good fidelity, is the use of an enclosed air-chamber behind the diaphragm, resonant at a frequency in the vicinity of the flare cut-off frequency, as used with the Klipsch loudspeaker. This air-chamber is designed to provide a capacitive reactance approximately equal to the inductive reactance (inertance) of the horn at the flare cut-off frequency. The volume of the air-chamber is given by eqn. (15) in Sect. 4(vi) below—see also Ref. 31.

(B) Diaphragm and throat

Some horns are designed with the throat area the same as the diaphragm area. Greater acoustical loading is obtained by the use of a larger diaphragm and a sound chamber. The maximum size is limited by the high frequency response—the distances from any parts of the diaphragm to the throat opening should vary by less than one quarter of the wavelength of the highest frequency to be reproduced. The simplest form of sound chamber is Fig. 20.21A, with a single hole in the centre. Better high frequency performance is obtained with the more complex forms of Figs. 20.21 B, C.

Relatively large throats are necessary for high efficiency at low frequencies, and relatively small throats are necessary for high efficiency at high frequencies. Consequently any loudspeaker must be a compromise, and **the highest efficiencies are obtained with limited frequency ranges.**

Second harmonic **distortion** is generated in the throat and follows approximately the theoretical relationship

$$\text{Percentage second harmonic} = (\sqrt{W}/81)\,(f_1/f_0) \times 100 \tag{14}$$

where W = acoustical watts per square centimetre of throat area
f_1 = frequency being radiated
and f_0 = cut-off frequency due to flaring (eqn. 2).

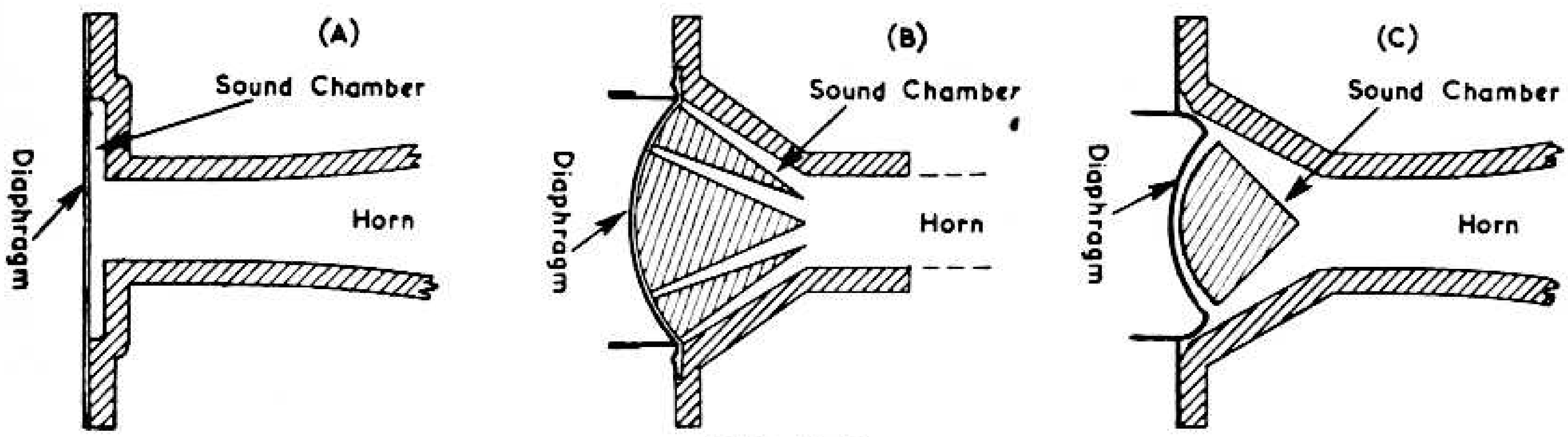


FIG. 20.21

Fig. 20.21. *Different forms of sound chambers in horn loudspeakers.*

For example if a horn with a 40 c/s cut-off is to reproduce a 4000 c/s note, there will theoretically be 8% second harmonic distortion for an acoustical power of 0.01 watt per square inch of horn throat. The actual distortion appears to be about half the theoretical value.

Thus for reasons of both efficiency and distortion, a horn should only be designed to cover a limited frequency range.

(C) Driving units

The simplest driving mechanism is the electro-magnetic type with an iron diaphragm as the armature (Fig. 20.21A).

All modern driving units are of the moving-coil type and may have a diaphragm of aluminium alloy or some form of paper or a cloth base impregnated with synthetic resin.

Well designed direct-radiator loudspeakers may be used as the driving mechanisms for horns. The most suitable size is from 8 to 12 inches diameter for medium power requirements. Some loudspeakers are designed specially for this application.

Where high power output at very low frequencies is required, a 15 inch unit may be used, as in the Klipsch corner horn described below.

(D) Distortion

In addition to the distortion caused by the throat, there is also **distortion due to the sound chamber.** The acoustical capacitance is a function of the position of the diaphragm, and the effect is most apparent at low frequencies where the amplitude is large. This distortion may be reduced by the use of a large sound chamber, thus limiting this unit to low frequencies only. A separate high frequency unit with a small sound chamber may be used, since it will not be required to handle large amplitudes.

Distortion is also caused by a non-linear suspension ; this is only serious at low frequencies (Ref. 14, Fig. 8.21). **Distortion at low frequencies due to non-linear suspension** may be reduced by

(1) the use of a large dynamic driver,
(2) increased compliance in the suspension,
(3) an enclosed air-chamber at the rear of the diaphragm as in the Klipsch corner horn.

Another cause of distortion is **frequency modulation** through the Doppler Effect—see Sect. 7(ii)—which can be reduced to small proportions by the use of separate high and low frequency units.

It is impossible to design a horn loudspeaker that covers a wide frequency band and is simultaneously free from non-linear distortion. Thus two separate units for low and high frequencies are essential for fidelity. This is a limitation peculiar to horn loudspeakers.

(E) Efficiency

With horn loudspeakers efficiencies as high as 80% can be achieved over a limited frequency range. A typical horn of the type used in cinema theatres has an efficiency from 30% to 45% over its useful range. The Klipsch corner horn has an efficiency around 50% over its useful range.

(F) Directional characteristics

For wavelengths larger than the mouth diameter, the directional characteristics are approximately the same as for a cone the same size as the mouth. At frequencies above 3000 c/s the directional characteristics are only slightly affected by the flare or length. At intermediate frequencies the directional characteristics are broader than those obtained from a piston (cone) the size of the mouth.

When a number of horns are arranged in a line, a sharp beam will be obtained when the horns are parallel to one another, while a broader beam will be obtained when they are arranged radially.

(G) Electrical impedance

A rise in electrical impedance normally occurs at the diaphragm resonance, although less than that with a direct radiator, with a rise at the higher frequencies as with a moving-coil direct radiator.

On account of the variation in impedance, triode power valves (preferably with negative voltage feedback) are highly desirable for fidelity.

(H) Damping and transients

With a well designed horn of sufficient mouth area, or with a smaller horn in which the resistive loading has a peak in the vicinity of the bass resonant frequency, the acoustical damping at that frequency will be high. The damping due to the output resistance of the amplifier will be additive.

At higher frequencies, experimental results indicate that spurious transients are much lower in horn loudspeakers than with direct radiators. For example (Ref. 155), a 12 inch direct radiator loudspeaker, when loaded by a short horn, gave 12 to 15 db less transient level than when used without the horn, using Shorter's method—see Sect. 7(iii).

(I) Horns and rooms

Horns require larger rooms than direct radiator loudspeakers, even apart from the space occupied by the speaker itself, owing to the roughness of the overlap between the low and high frequency units. The distance from loudspeaker to listener must be greater than that with cone type direct radiators (Ref. 150).

Horn loudspeakers are frequently mounted in the corner of the room, so that the angle between the walls acts as sort of continuation of the horn. Some ingenious corner horns have been developed, for example one in which the difference between the solid angle between the walls and the desired exponential flare is taken up by a suitably shaped "plug" (Ref. 130).

(vi) Folded horn loudspeakers

Owing to the large space required by a horn, much attention has been paid to folded horns. These all involve some loss of the higher frequencies, due to reflections and differences in length of path causing cancellation of some frequencies. The number of folds should be kept to the minimum—most folded horns have only one or two folds. Any increased number of folds, if unavoidable, must be accompanied by a reduction in the maximum frequency.

Concentric folded horns

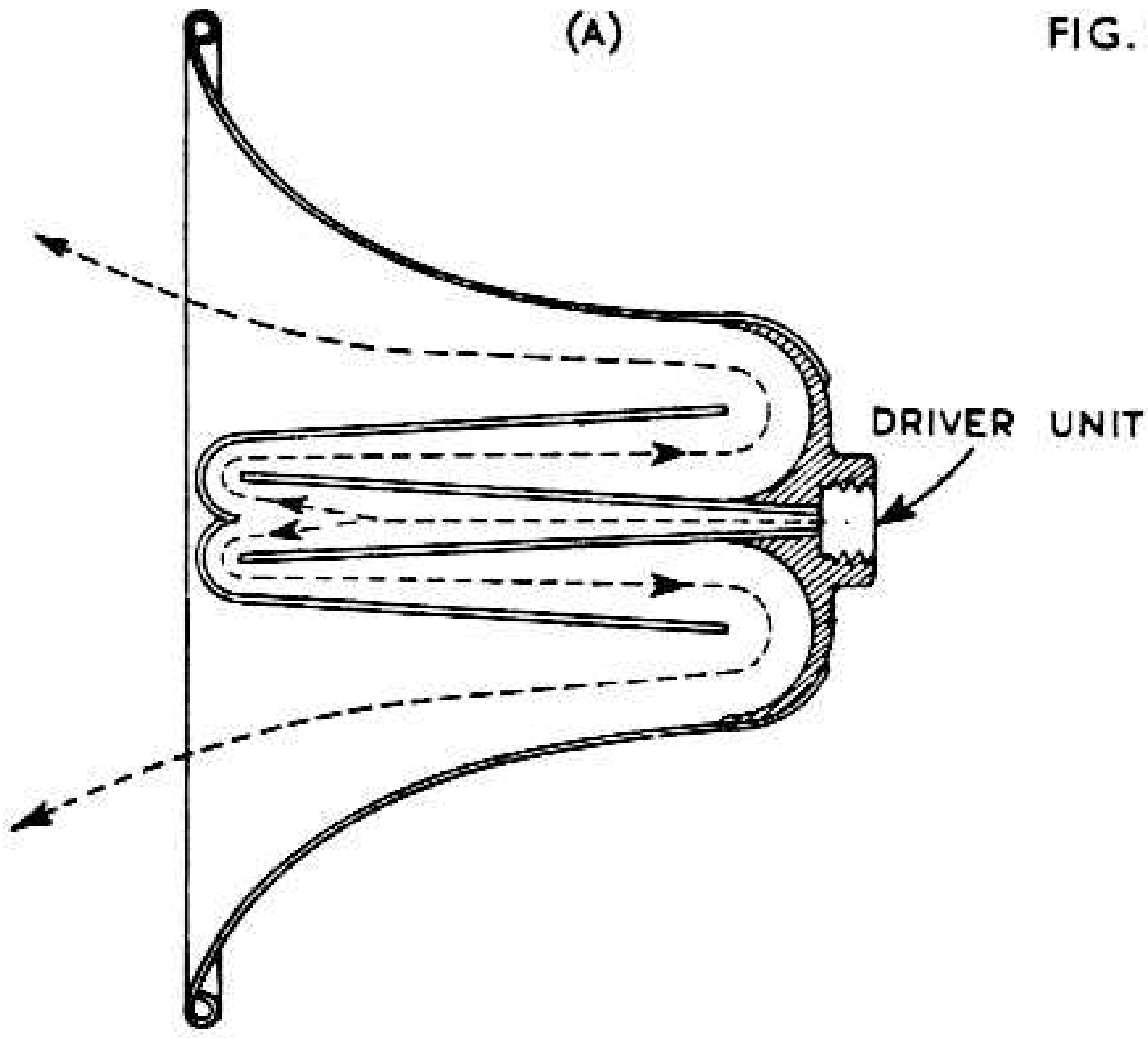
In these, the high frequency loss may be made fairly small, and it is possible to cover a range from below 200 up to 8000 c/s. One popular form is the three section directional reflex (Fig. 20.22A) in which the total length of the air column is nearly three times the overall length ; this gives useful radiation over an angle of 60°. A modified form is the radial type which may be suspended from the ceiling to give an angle of 360° ; this gives almost the same radiation upwards as downwards. Refs. 2, 82.

Low frequency folded horns

When a folded horn is used as the low frequency unit in a dual system, the loss of higher frequencies is actually an advantage, and the inner surface of the horn is frequently covered with sound absorbent material to eliminate high frequency reflections. Some of the forms which such a horn may take are given in Figs. 20.23 (Ref. 51) and 20.24 (Ref. 52).

FIG. 20.22

(A)



(B)

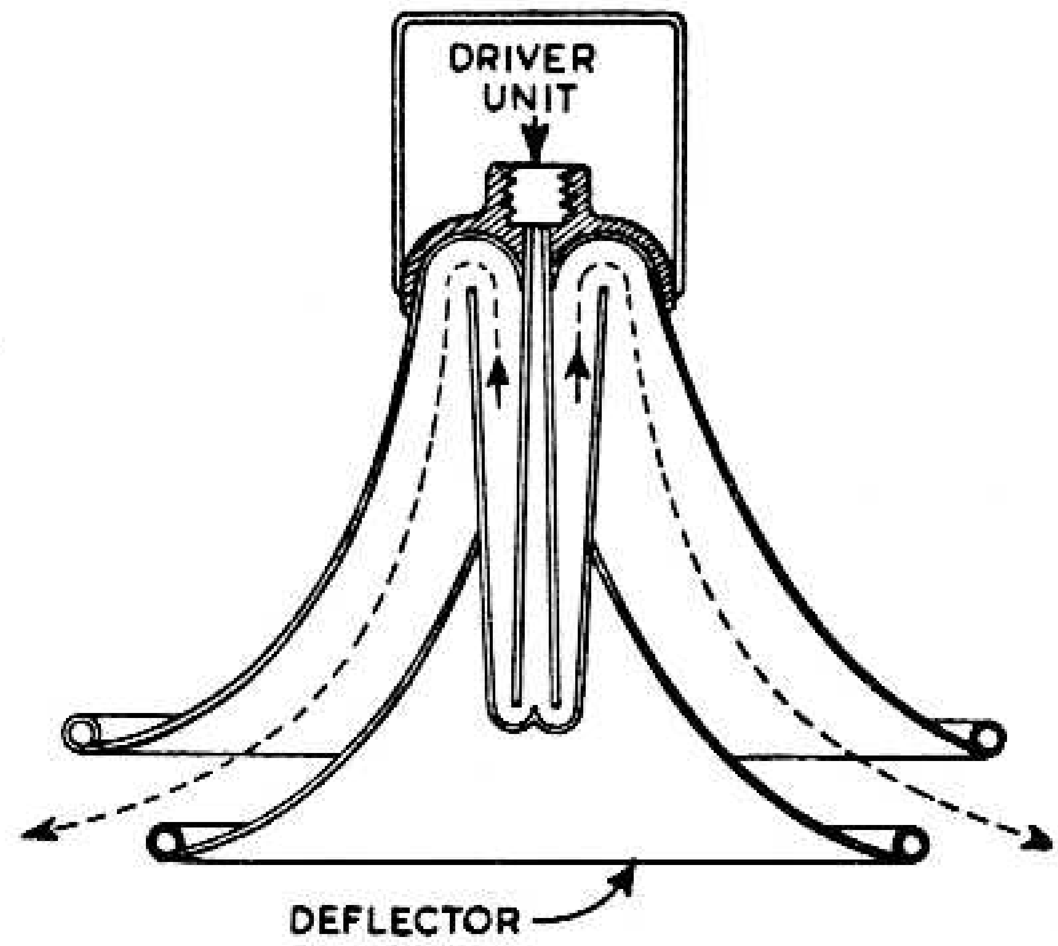


Fig. 20.22. *Concentric folded horns* (A) *three section reflex* (B) *three section reflex radial type.*

One outstanding design is the Klipsch corner type speaker which includes a folded low frequency horn.

This is notable mainly on account of the comparatively small space which it occupies —20 cubic feet in all—compared with other dual horn systems having equivalent performance. The efficiency of the low frequency woofer unit (K-3-D) is not less than 50% down to 36 c/s, and still fairly high at 32.7 c/s ; it is capable of radiating a clean fundamental at reduced power down to 27 c/s. Maximum electrical power input is 15 watts, so that the maximum acoustical power output is over 7 watts.

FIG. 20.23

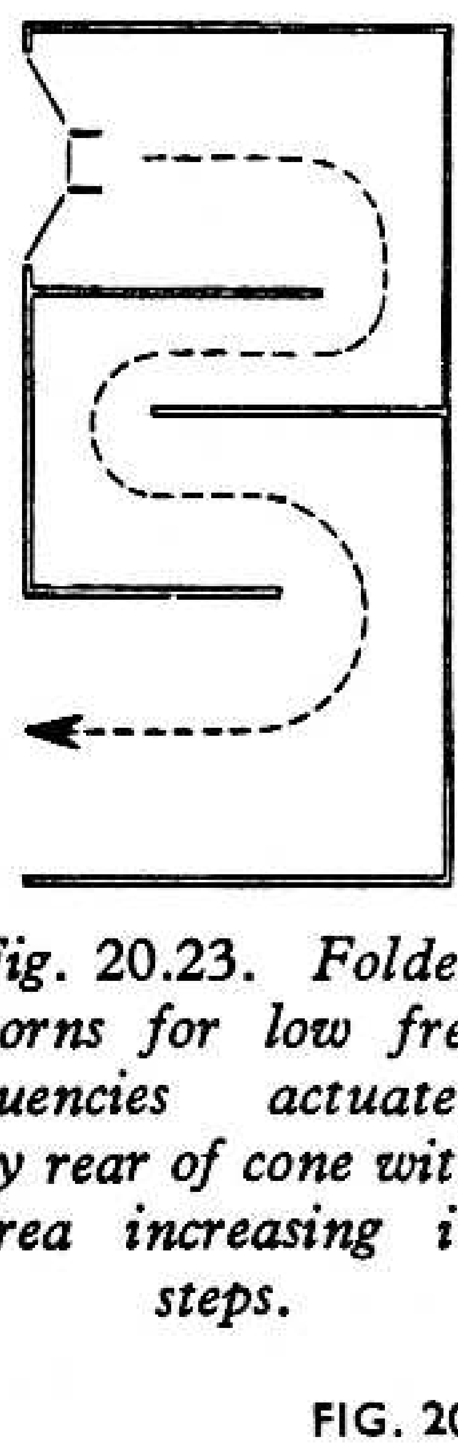


Fig. 20.23. *Folded horns for low frequencies actuated by rear of cone with area increasing in steps.*

The low frequency unit has a 15 inch direct radiator loudspeaker with an enclosed cabinet baffle at the rear and a folded horn in front. The recommended driver for ordinary home and small theatre power levels is the Stephens P-52-LX-2 woofer motor ; this is specially treated to increase its compliance before installation. The enclosed cabinet baffle at the rear is designed to offset the mass reactance of the throat impedance at low frequencies. The volume of the enclosed baffle is given theoretically by

$$V = 2.9\ AR \qquad (15)$$

where V = volume in cubic inches

A = throat area in square inches

and R = length of horn in inches within which the horn area doubles.

FIG. 20.24

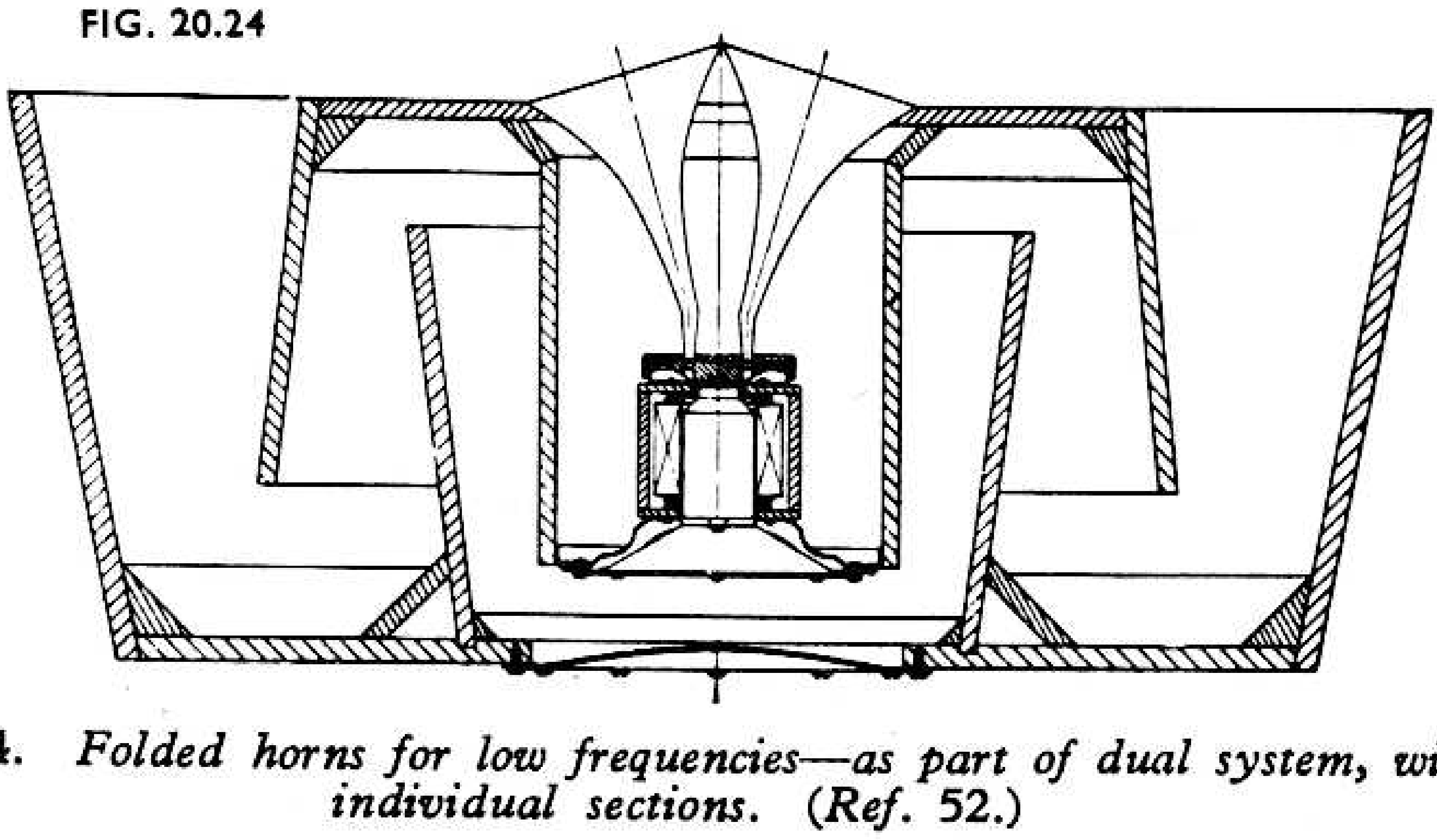


Fig. 20.24. *Folded horns for low frequencies—as part of dual system, with conical individual sections.* (*Ref.* 52.)

For the K-3 woofers, $A = 88$ sq. ins., $R = 16$ and the calculated volume is 5270 cubic inches. This equation presumes the suspension compliance to be infinite, which is not the case ; some experimental adjustment is therefore necessary.

The general construction is shown in Figs. 20.25 and 20.26. The voice coil impedance of two designs, the second with increased cone compliance, is shown in Fig. 20.27.

The cross-over frequency is 500 c/s, so that the **high frequency horn** has to be of unusual design, capable of handling 15 watts input down to 500 c/s. The recommended driver for home use is the Stephens P-15 HF motor, with which the response is good to 15 000 c/s. Other suitable drivers are the Jensen XP-101, Western Electric 713A, or Stephens P-30, P-40. The cell structure is contained in the region of small cross-section of the horn, so that the individual mouths of the cells are small. The construction of the high frequency horn is shown in Fig. 20.28.

References to Klipsch horn loudspeakers : 30, 31, 32, 33, 34, 48, 99, 115, 119, 122.

(vii) High frequency horns

High frequency horns are frequently used in dual or triple systems in conjunction

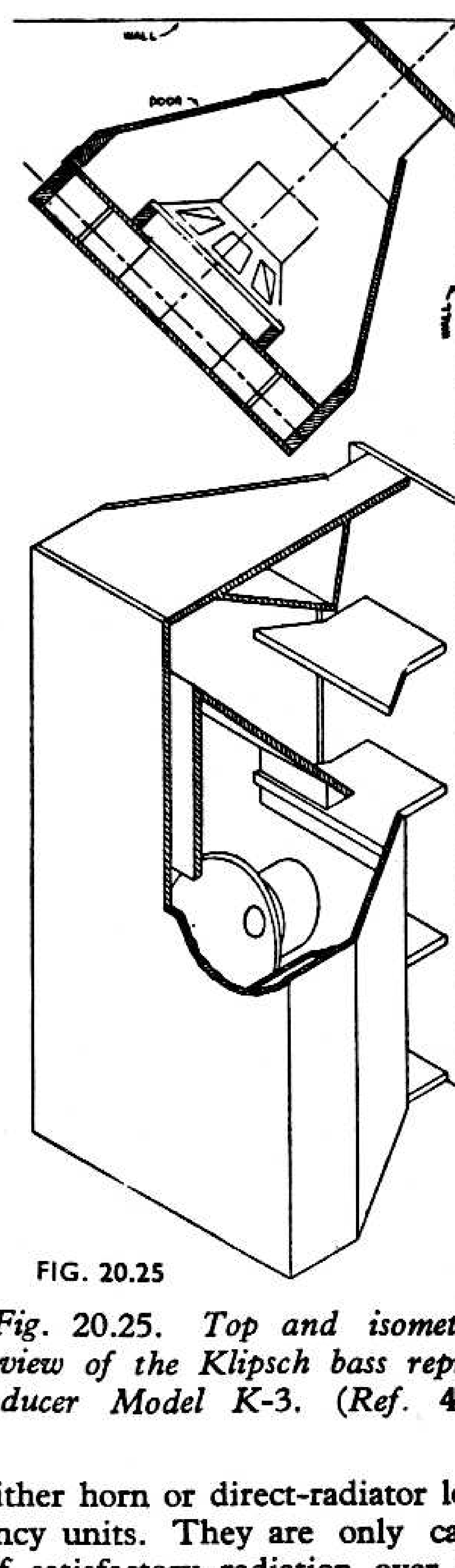


FIG. 20.25

Fig. 20.25. *Top and isometric view of the Klipsch bass reproducer Model K*-3. (*Ref.* 48).

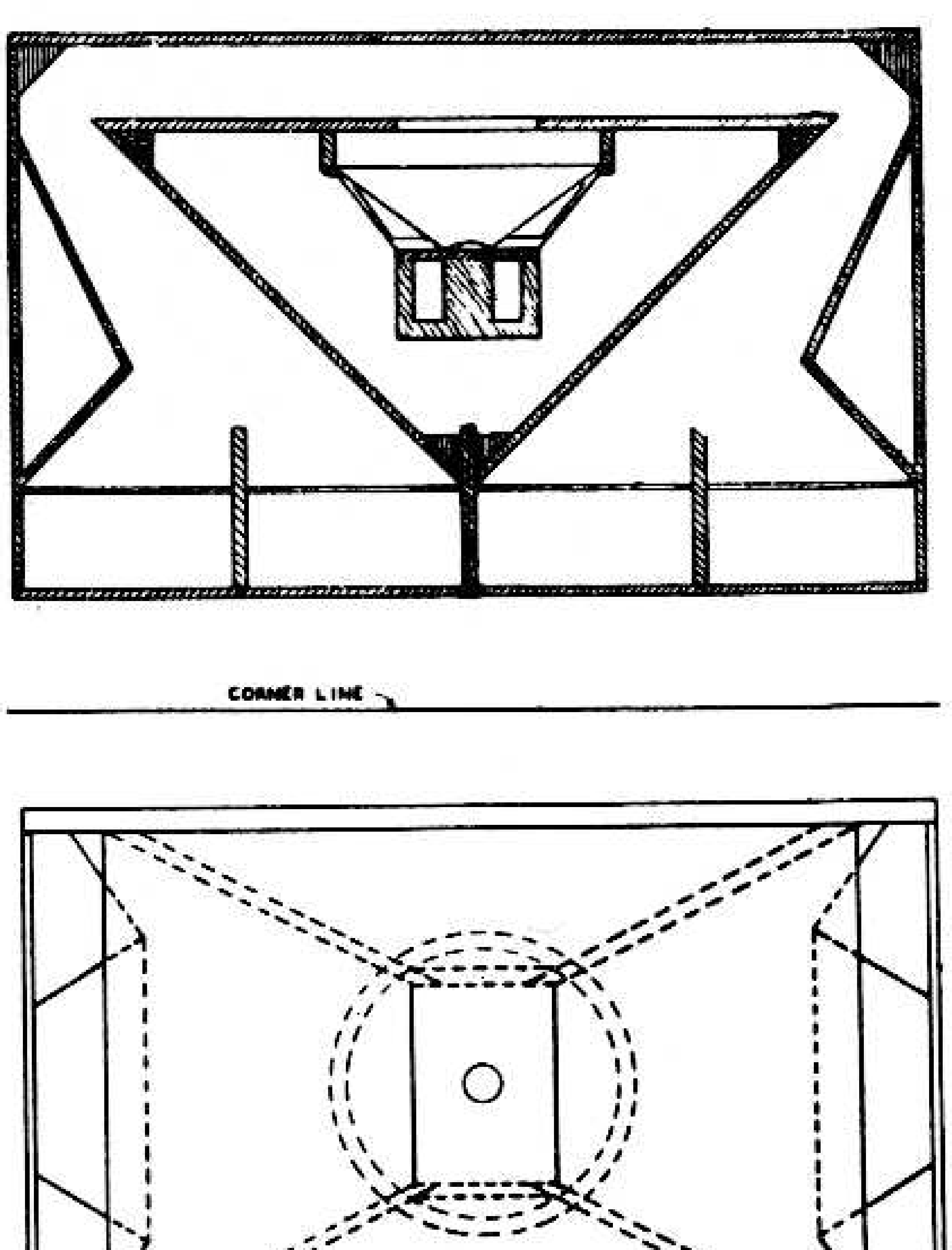


FIG. 20.26

Fig. 20.26. *Side and front views of the Klipsch bass reproducer Model K*-3. (*Ref.* 48).

with either horn or direct-radiator low frequency units. They are only capable of satisfactory radiation over an angle of from 20° to 40° for a single horn, but up to six similar horns may be built into one unit operated by a single driver to give good coverage over a wide horizontal angle (Refs. 76 Part 1 ; 52, 136). Alternatively the multi-cellular construction may be adopted (Ref. 33). Diffusing lenses may also be used with high frequency horns as an alternative to multi-cellular construction, to spread the sound over a wider angle. In one example the angle was increased from about 20° to over 50° at 8000 c/s by the use of such a diffusing lens. The construction appears cheaper than a multi-cellular

FIG. 20.27

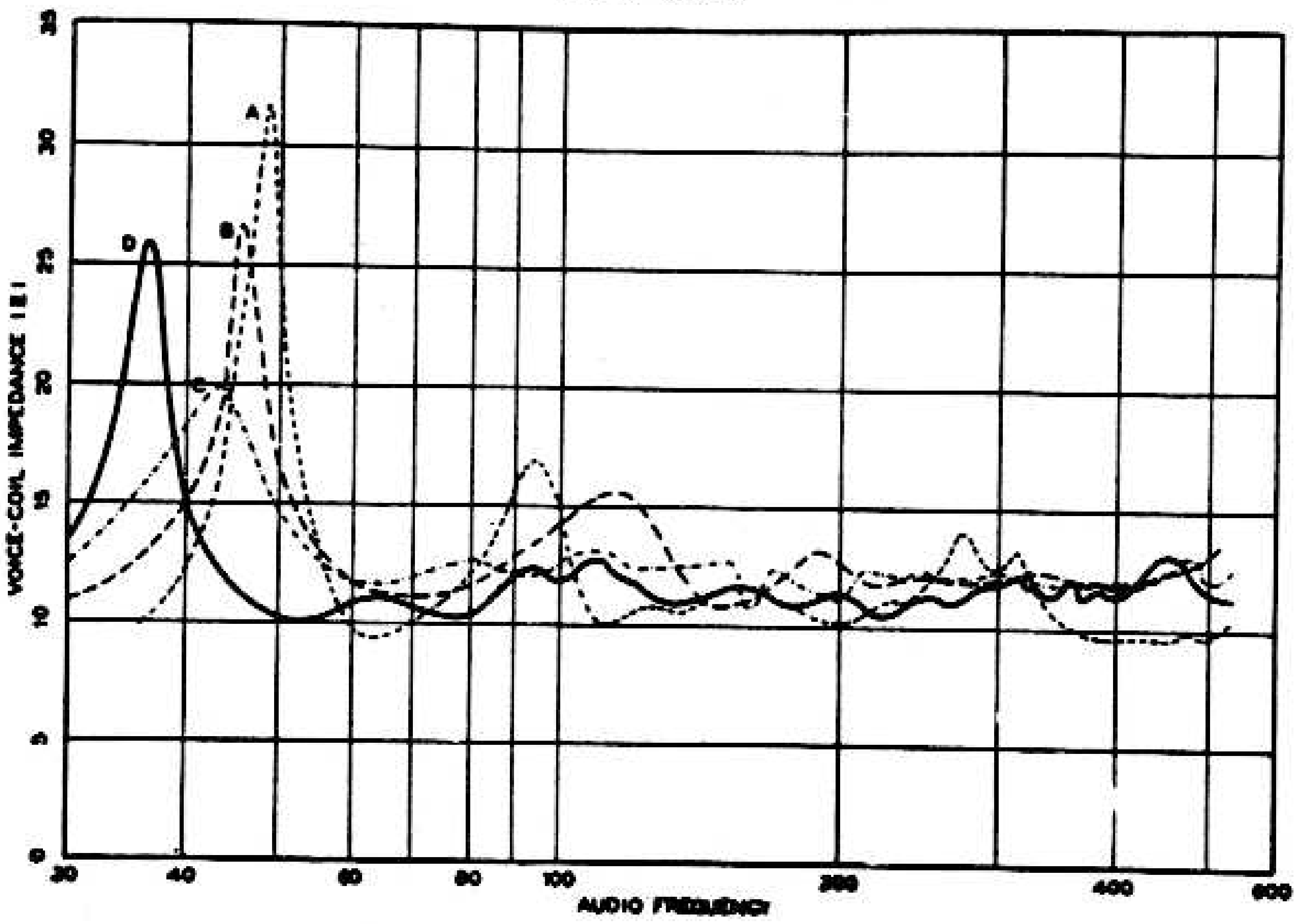


Fig. 20.27. *Voice coil impedance of two designs of Klipsch bass reproducer (A) Model K-3-B as Ref.* 48 *(B) Model K-3-C (cone with increased compliance).* (*Ref.* 99).

horn and permits greater flexibility in application (Refs. 120, 141).

References to corner ribbon high frequency horn : 229, 230.

Reference to reverse flare principle in high frequency horn : 232.

See also the Klipsch high-frequency horn (Fig. 20.28) and page 858.

References to high frequency horns : 33, 76 (Part 1), 80, 137, 165.

General references to horns : 2, 13A, 14, 15, 30, 31, 32, 33, 34, 38 (Parts 1 and 2), 39, 45, 46, 47, 48, 51, 52, 59, 69, 73, 76, 77, 82, 99, 115, 119, 128, 130, 137, 142, 147, 165, 175, 177, 206, 209, 211, 217, 232.

(viii) Combination horn and phase inverter loudspeakers for personal radio receivers

A combination horn and phase inverter loudspeaker for personal radio receivers has been described (Ref. 124) which has an efficiency of about 25% and a frequency range from 300 to 4000 c/s. A sound level of 84 db is obtained at 3 feet from the loudspeaker with an input of 10mW—this level is somewhat higher than conversational speech. Sub-miniature valves and very small B batteries may therefore be used to give an acceptable sound output from a receiver having a cubic capacity of 41 cubic inches.

(ix) Material for making horns

The best materials are concrete, brick and masonry. The most practical material for horn construction, with many good features, is untempered 3/16 inch Masonite. It should be reasonably strutted and backed with absorbent material (Ref. 191).

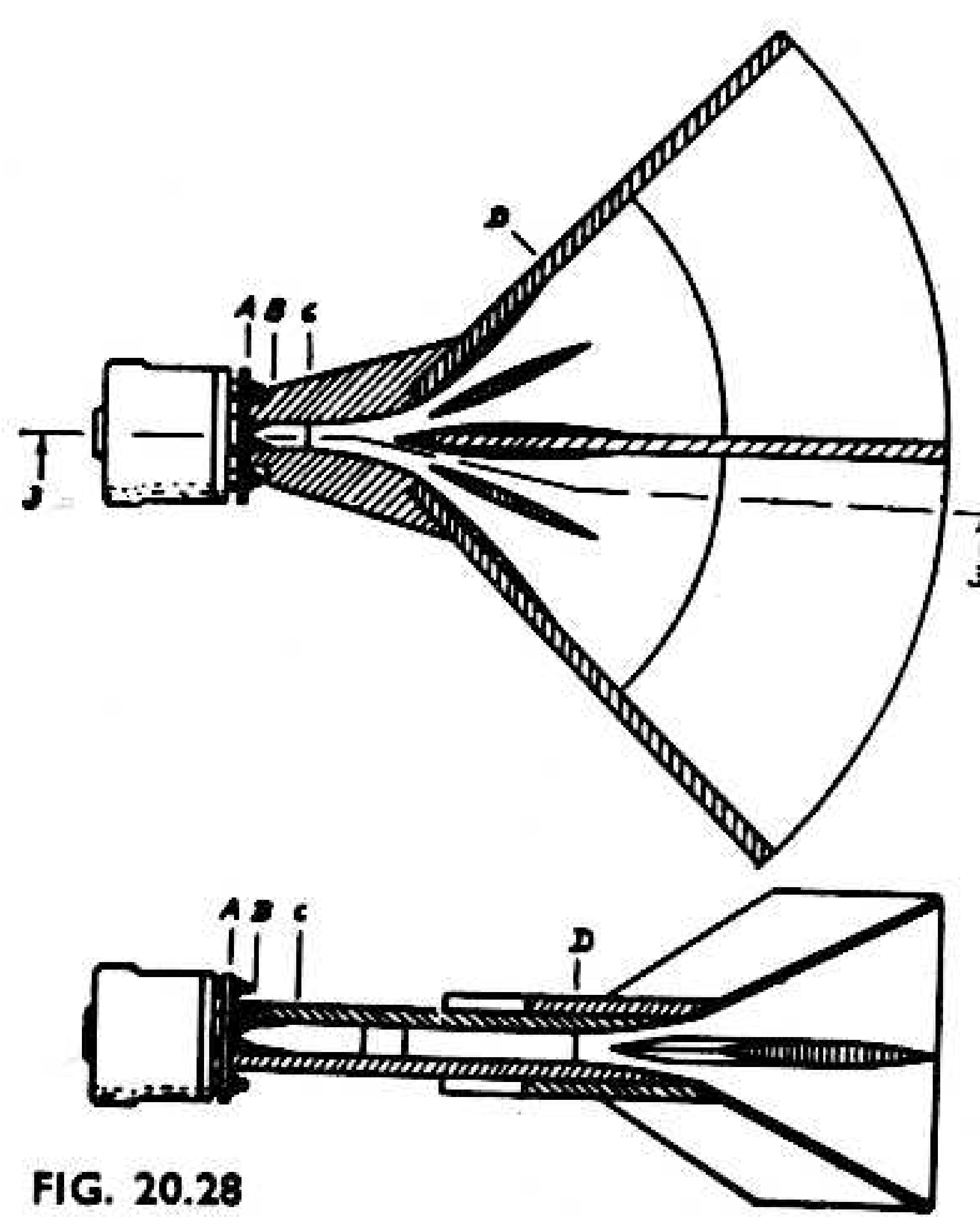


FIG. 20.28

Fig. 20.28. *Klipsch high frequency horn loudspeaker : Upper view—sectional view from top ; lower view—sectional view from the side through line* 3-3 *of upper view.* (*Ref.* 33).

SECTION 5: DUAL AND TRIPLE SYSTEM LOUDSPEAKERS

(i) Introduction (ii) Choice of the cross-over frequency (iii) The overlap region (iv) Compromise arrangements.

(i) Introduction

When a wide frequency range has to be covered there is a choice between

1. A single unit employing some special construction to extend the frequency range—see Sect. 2(iii),
2. Two or three separate loudspeakers, each covering a limited frequency range,
3. An integral dual or triple system with two or three loudspeakers mounted (usually co-axially) in one equipment, and
4. Some compromise arrangement—see (iv) below.

If separate loudspeakers are used, they should be mounted as closely together as possible, with co-axial mounting as the ideal; this is not so important when the cross-over frequency is below 500 c/s. In addition, they should be co-planar, with the plane of horn loudspeakers taken as the plane of the diaphragm. The loudspeakers should be correctly phased, so as to be additive in the overlap region.

A frequency dividing network is used to split the output between two loudspeakers so that neither unit is called upon to handle large amplitudes of frequencies beyond its range. This has the advantage that Doppler Effect distortion (see Sect. 7) is much reduced. Other advantages are that, on account of limitations in the frequency range of each unit, the system efficiency is increased, while the directivity characteristic is improved due to the smaller diaphragm (or horn mouth) for the high frequency unit. In addition, the transient response is improved, and there is less intermodulation and reduced frequency modulation.

The loudspeakers should preferably have the same efficiency, otherwise one will have to be attenuated. The directional characteristics of the high frequency unit should receive careful attention.

Integral dual systems

One excellent arrangement employs two cones, co-axial and co-planar, with the small high-frequency cone mounted near the apex of the large cone. Both cones vibrate in unison in the overlap region. (Refs. 61, 116, 116A).

Another arrangement employs a large low-frequency cone and a co-axial high-frequency horn. This has the disadvantage that the sound sources are not co-planar, since the diaphragm of the high frequency unit is mounted to the rear of the cone.

Integral triple systems

In one design there is a large low-frequency cone, a mid-frequency horn using the flared low-frequency cone for its mouth, and a high-frequency horn mounted in front of the mid-frequency horn.

(ii) Choice of the cross-over frequency

In a dual system, that is one having a low-frequency and a high-frequency loudspeaker, the cross-over frequency is usually between 400 and 1200 c/s. The following points must be satisfied.

1. The low frequency unit (" woofer ") must be capable of handling at least half an octave above the cross-over frequency, at full power. See also (4) below.
2. The high frequency unit must be capable of handling at least half an octave below the cross-over frequency at full power. See also (4) below.
3. Provided that point (2) can be satisfied, the cross-over frequency should be as low as possible, say 400 to 500 c/s.
4. If a 6 db/octave frequency dividing network is used, each unit should be capable of handling one octave beyond the cross-over frequency at full power, and about 3 octaves beyond the cross-over frequency at reduced power.

Some systems have cross-over frequencies from 1200 to 2000 c/s, in which case the high frequency unit approaches more closely to a true " tweeter," since the latter is generally limited to frequencies above 2000 c/s (Refs. 11, 45).

In a triple system there is no necessity for such a compromise regarding the cross-over frequency. The " woofer " may handle up to between 300 and 600 c/s, the middle unit up to between 2000 and 5000 c/s, and the " tweeter " will then look after the higher frequencies (e.g. Ref. 134).

The design of frequency dividing networks is covered in Chapter 21, Sect. 3

(iii) The overlap region

Serious distortion often occurs in the overlap region when both units are contributing to the total acoustical output. In a dual direct-radiator system in which the distances from each cone to the listener are not equal, the response characteristic will have pronounced peaks and valleys at frequencies where the two sources are in and out of phase. In the case of a dual horn system, particularly when one horn is folded, the acoustical paths may differ sufficiently to cause the same effect. Similar effects occur with other combinations, but the trouble is minimized in all cases by a low cross-over frequency (less than 600 c/s) and, in the case of dual horns, by a fairly steep attenuation characteristic (12 or 18 db/octave nominal). Most loudspeakers give serious distortion below the frequency of minimum rated response, even though the level is attenuated by the cross-over network—hence the desirability of an extended frequency range.

There tends to be a peak of distortion at the cross-over frequency owing to incorrect impedance matching and the partially reactive load. This is most serious with pentodes and beam power amplifiers (see Chapter 21).

(iv) Compromise arrangements

One possibility is to use a single moving-coil loudspeaker with the front acting as a direct radiator, and the rear as bass horn (Ref. 51). Another arrangement uses a single driver with a high-frequency horn facing forwards and a folded bass horn driven by the rear of the diaphragm (Ref. 52). Owing to the use of a single cone or diaphragm, the high frequency response does not extend as far as with two separate units, each specially designed.

General references to dual and triple systems : 11, 30, 33, 45, 48, 51, 52, 61, 76, 130, 134, 150, 165, 167, 194, 215.

SECTION 6 : LOUDSPEAKERS IN OPERATION

(i) Loudness (ii) Power required (iii) Acoustics of rooms (iv) Loudspeaker placement (v) Stereophonic reproduction (vi) Sound reinforcing systems (vii) Open air Public Address (viii) Inter-communicating systems (ix) Background music in factories.

(i) Loudness

Loudness may be measured in loudness units—see page 827 and Fig. 19.8.

(ii) Power required

(A) Direct radiation (outdoor)

Some loudspeaker manufacturers quote the intensity level on the axis at a distance of, say, 30* feet with a stated electrical input power, usually with a warble frequency (from 300 to 3300*, 500 to 2500 or 500 to 1500 c/s). The intensity level decreases by 6 db each time the distance is doubled, or increases by 6 db each time the distance is halved.

The intensity level increases by 3 db when the power input to the loudspeaker is doubled, by 6 db when the power is quadrupled, and so on.

*R.M.A. SE-103. See Sect. 6(x).

Example : At a distance of 30 feet on the axis, a certain loudspeaker is stated to produce a level of 81.5 db above 10^{-16} watt per square centimeter at 8 watts input with a warble frequency from 500 to 2500 c/s. Find the intensity level on the axis at a distance of 6 feet with 3 watts input.

Effect of change of distance $= 20 \log_{10} (30/6) = + 14$ db.
Effect of change of power input $= 10 \log_{10} (3/8) = - 4.3$ db.
Net change $= + 9.7$ db.
Intensity level at 6 feet with 3 watts input $= 81.5 + 9.7 = 91.2$ db.

In other cases loudspeaker manufacturers (such as R.C.A.) publish the curves of sound pressure versus frequency, from which it is easy to estimate the average over the frequency range of interest. In this case the reference level is 0 db = 10 dynes per square centimetre, and the microphone distance is 24 inches. In a typical case the output level is $-$ 1 db for an input of 0.1 watt to the loudspeaker. The level at a distance of 30 feet for a power input of 8 watts will therefore be

$$- 1 + 19.03 - 23.52 = - 5.49 \text{ db } (0 \text{ db} = 10 \text{ dynes/cm.}^2)$$

or converting to the basis of 0 db = 0.0002 dyne/cm^2,

$$- 5.49 + 93.98 = + 88.5 \text{ db } (0 \text{ db} = 0.0002 \text{ dyne/cm}^2).$$

Alternatively the R.M.A. loudspeaker pressure rating may be quoted. For calculation and example see page 812.

FIG. 20.29

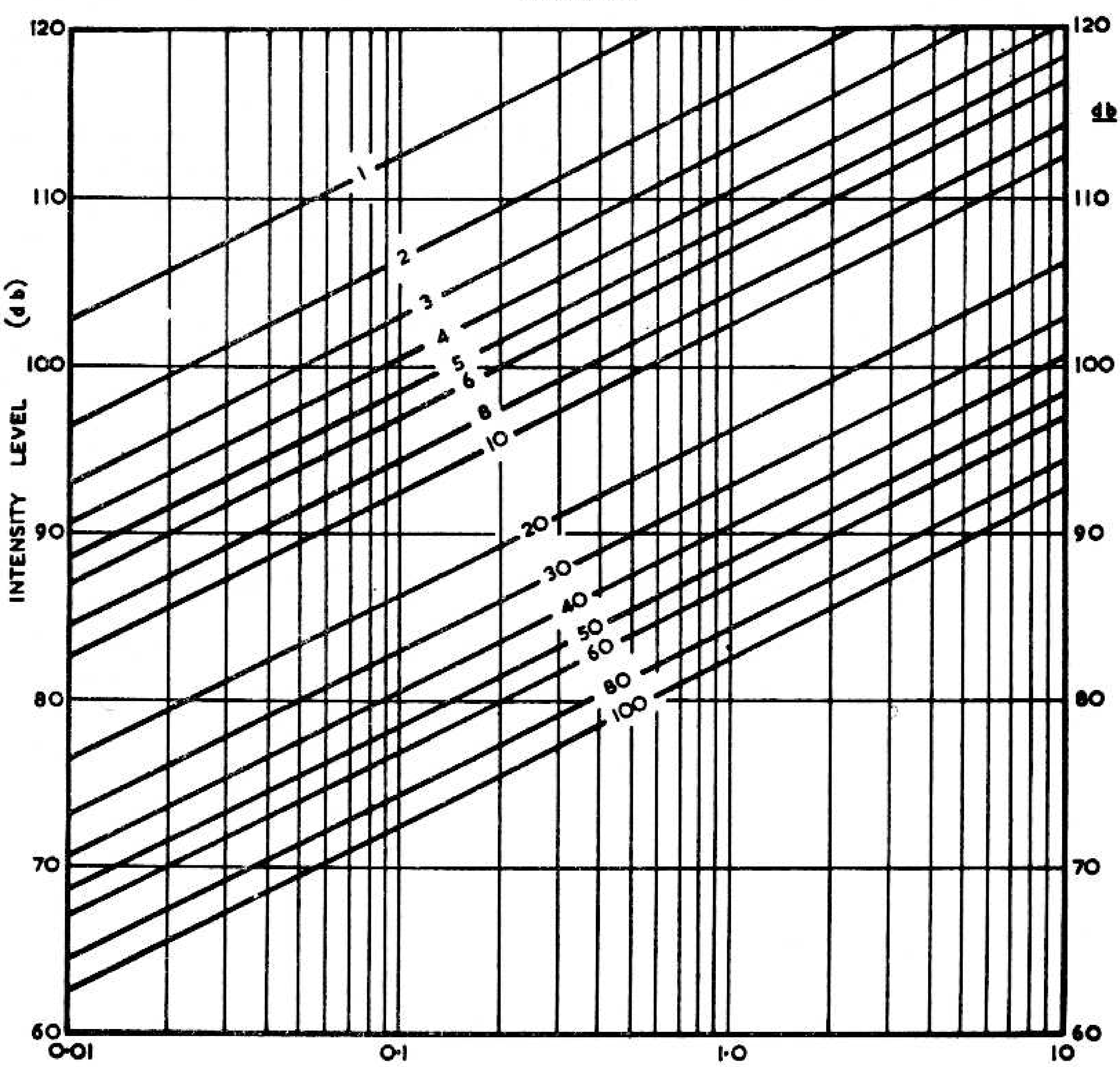


Fig. 20.29. Intensity in db versus acoustical watts output at specified distances along the axis of a loudspeaker which is considered as a small source in an infinite baffle. This holds fairly accurately at low frequencies, with an error of 1 db at 500 c/s with a cone 10 inches in diameter. The numbers on the sloping lines indicate the distance in feet from the loudspeaker (based on F. Massa, Ref. 69, but extended and modified).

Again, the loudspeaker percentage efficiency may be known, say 3%, and it is desired to calculate the intensity at a distance of, say, 30 feet with an input of 8 watts. The unknown factor is the angle of radiation, but at frequencies up to 500 c/s this may be taken as approximately 180° (assuming a direct radiator with a large flat baffle) and we may therefore make use of Fig. 20.29. In this case the acoustical output will be $8 \times 0.03 = 0.24$ watt and the intensity will be 87 db at 30 feet (Fig. 20.29).

At higher frequencies there will be an increasingly greater focusing effect, and the intensity on the axis will be somewhat greater than the value calculated above.

(B) Power required indoors

When a loudspeaker is operated indoors the direct radiation is supplemented by the reflected sound.

The **reverberation time** is the time in seconds for a sound to fall to one millionth of its original intensity (− 60 db) after stopping the source. Eyring* gives

$$T = \frac{0.05\ V}{-S \log_\epsilon (1 - \alpha)}$$

where T = reverberation time in seconds,
V = volume of room in cubic feet,
S = surface area of walls, ceiling and floor in square feet,
and α = average absorption coefficient per square foot, in sabins (values are less than unity).

The sabin is an absorption unit representing a surface capable of absorbing sound at the same rate as does 1 sq. ft. of perfectly absorbing surface, such as an open window.

In a typical living room, the reverberation time is about 0.5 second at 500 c/s and the absorption coefficient is about 0.25. The reverberation time falls to possibly 0.3 second at 5000 c/s and rises to possibly 0.75 second at 200 c/s.(Ref. 14, Fig. 13.26).

In a very large living room, the reverberation time would probably be about 0.8 second at 1000 c/s.

FIG. 20.30

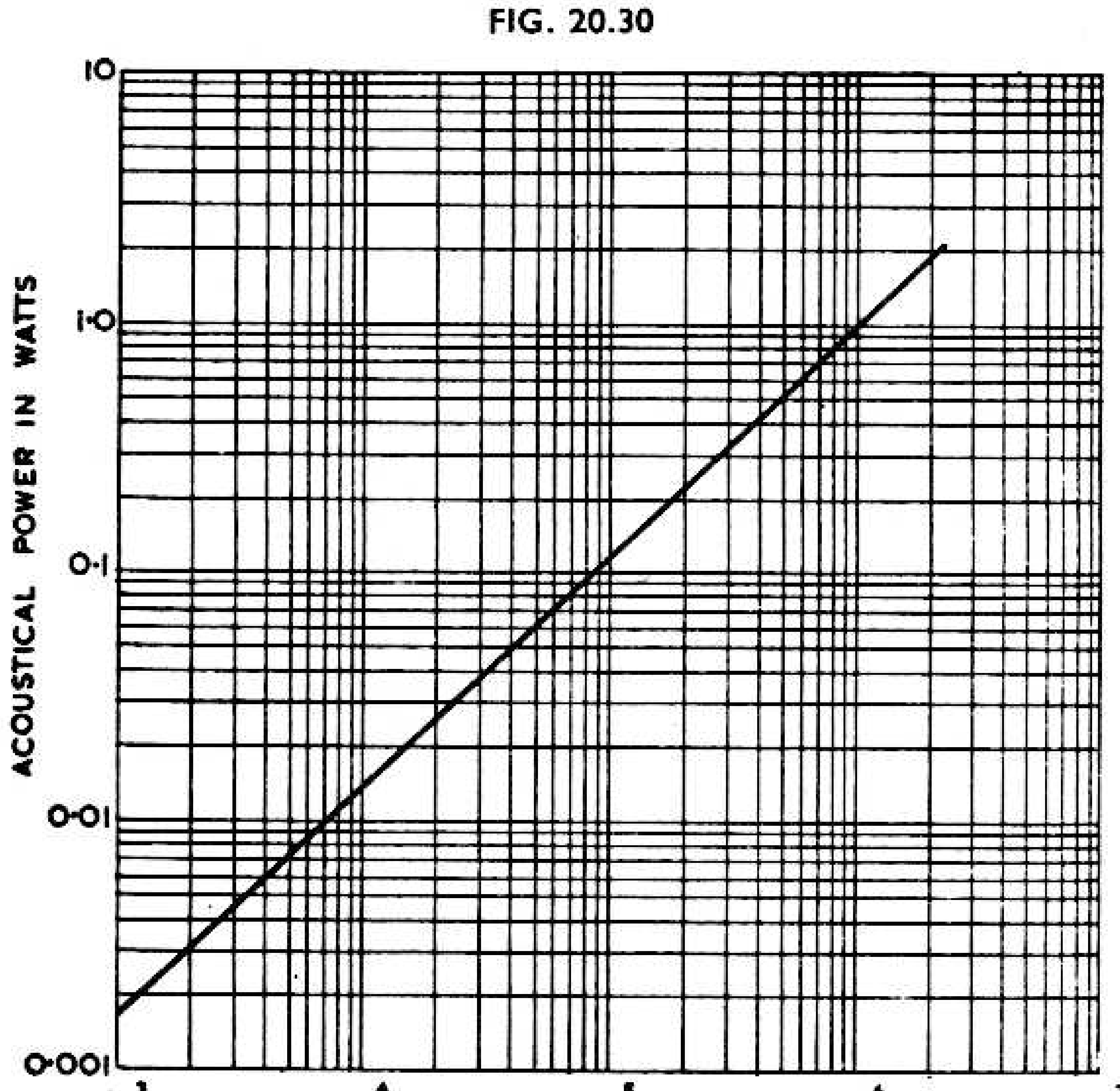


Fig. 20.30. *Acoustical power required to produce an intensity level of* 80 *db as a function of the volume of the auditorium for optimum reverberation times (after Olson, Ref.* 13, *by kind permission of the author and of the publishers and copyright holders, Messrs. D. Van Nostrand Company Inc.).*

*See Ref. 13A page 399.

The " optimum reverberation time " at 1000 c/s depends on the volume of the room (Ref. 13A) :

Volume	1000	2000	4000	20 000	100 000	500 000	cu. ft.
T_{opt}	0.7	0.75	0.82	1.01	1.28	1.62	secs.

In a typical room 20 ft. × 15 ft. × 10 ft., with a reverberation time of 0.5 second, and with the listener 6 feet from the loudspeaker, the reflected sound adds 3 db to the direct radiation at 500 c/s under steady state conditions, and 1 db when the sound only continues for 0.02 second as for speech (Ref. 14, Fig. 13.26). At higher frequencies the reflected sound becomes rapidly smaller, and may be neglected. At 200 c/s the reflected sound adds 5.5 db under steady state conditions or 2 db when the sound continues for 0.02 second. At 100 c/s the reflected sound adds 9.5 db under steady state conditions, or 6 db when the sound continues for 0.02 second. The effect of the reflected sound becomes more important as the listener moves further away from the loudspeaker.

References 13A, 14, 15, 50, 55, 175, 178.

The **acoustical power** in watts to produce an intensity level of 80 db (0 db = 0.0002 dyne/cm^2) is plotted in Fig. 20.30 against the volume of the enclosure in cubic feet (after Olson, Ref. 13A). For living rooms the following information may be derived—

Volume	1000	2000	3000	4000	cu. ft.
Intensity level	Acoustical power in watts				
55 db	0.000 0051	.000 0098	.000 014	.000 019	watt
60 db	0.000 016	.000 031	.000 045	.000 059	watt
65 db	0.000 051	.000 098	.000 14	.000 19	watt
70 db	0.000 16	.000 31	.000 45	.000 59	watt
75 db	0.000 51	.000 98	.001 4	.001 9	watt
80 db	0.001 6	.003 1	.004 5	.005 9	watt
90 db	0.016	031	.045	.059	watt
100 db	0.16	.31	.45	.59	watt
110 db	1.6	3.1	4.5	5.9	watts

These values apply to enclosures with optimum reverberation times. Living rooms may have reverberation times which are lower than the optimum values, and the acoustical power required will then be greater than indicated above.

The intensity levels for home listening under various conditions are set out in Chapter 14 Sect. 7(iii).

(iii) Acoustics of rooms

When a musical item is produced in the studio and reproduced by a loudspeaker in a living room, there are two reverberation times—that of the studio and that of the room. Good listening conditions are usually achieved by controlling the reverberation time of the studio so that the music will sound well in the average medium or large living room. Large concert halls with long reverberation times are not ideal, but there is nothing which can be done in the living room to improve matters.

The desirable and usual rise of reverberation time at low frequencies with rigid walls may be replaced by a general fall, due to the **vibration** of walls, floor and ceiling. This is one prevalent cause of loss of bass, below 150 c/s ; if an attempt is made to counteract it by bass boosting, the acoustical effect is not quite natural (Ref. 53).

It is preferable to mount the loudspeaker on sound absorbing pads or thick carpet to prevent the direct transmission of vibration to the floor.

At low frequencies it is doubtful whether the reverberation time of a small or even medium sized room means much, because of **room resonances** producing standing waves. In a typical case with a room 16 ft. 9 in. × 11 ft. 6 in. × 8 ft. 6 in. the first nine resonances are 36.8, 51.1, 62.9, 68.6, 73.6, 77.6, 85.5, 102.3, 137.2 (Ref. 53).

The most desirable shape for a listening room is approximately in the ratio height : width : length = 1 : 1.27 : 1.62, so as to distribute the resonances fairly uniformly.

The peaks due to these resonances are of the order of 20 db and completely mask less pronounced effects. The bad effects may be reduced by

(1) increased acoustical damping at low frequencies, especially near the corners of the room where it is twice as effective as elsewhere (Ref. 175), and
(2) an open door or window, or preferably both.

In addition to these room resonances, which occur over the whole room, there are other peaks of response due to focusing or interference between different paths, at middle and high frequencies, which vary from point to point. Thanks to our two ears, these sharp peaks and valleys are much less evident than they otherwise would be.

References 13A, 14, 53, 175, 192, 208, 220, 221.

(iv) Loudspeaker placement

The ideal position for a loudspeaker is in one corner of the room, either low down near the floor or in the corner between two walls and the ceiling. In either case the boundaries of the room form a triangular conical horn, and the angle of radiation is $\pi/2$ steradians. The axis of the loudspeaker may point directly at the head of one listener, say seated with his head 42 inches above the floor, but every other listener will then lose something of the high frequencies. If a ceiling-corner position is used, the maximum angle of elevation between a listener and the loudspeaker should not normally exceed 30° ; this position is therefore limited to large rooms or those with low ceilings. If a special loudspeaker is used having wide angle radiation at all frequencies, the problem disappears. In other cases the trouble may be minimized by

(1) accentuating the high frequencies and pointing the axis of the loudspeaker away from the listeners, who should be seated at approximately the same angle to the axis, or
(2) keeping the listeners within an angle of 15° from the axis.

Another possible position for the loudspeaker is in the corner of a room, half-way between floor and ceiling—this gives an angle of radiation of π steradians.

In all cases where a loudspeaker is mounted in the corner of a room, the adjacent walls and floor or ceiling should have reflecting surfaces so as to act as a horn. The rest of the room should have a considerable amount of damping material such as upholstered furniture, heavy drapings, carpets, books or acoustical tiles.

In all cases, no listener should be closer than 4 feet from the loudspeaker, with a greater distance for dual or triple systems and considerably greater distance for horns.

Loudspeakers with open-backed cabinets are very difficult to place satisfactorily. They should be about 9 inches out from the wall, and the wall behind them should preferably be damped by a heavy curtain or sheet of acoustical absorbent material. A corner position is desirable.

If a loudspeaker must be placed along one of the walls, it should preferably be along the shorter side, about the centre. This gives an angle of radiation of 2π steradians.

If two separate loudspeakers are used, each handling the full frequency range, they may, if desired, be placed in positions wide apart, such as the two ends of a room. In a fairly large living room some approach to a third dimensional effect may be obtained with the listener slightly nearer to one loudspeaker—this is the pseudo-stereophonic effect, see (v) below.

(v) Stereophonic reproduction

Reproduced sound normally comes from a single source—the loudspeaker—so that there can be no indication of the direction from which the sound originally came. In stereophonic reproduction with three or more channels there can be a 3 dimensional effect which enables the listener to determine the position of the source. Stereophonic reproduction in the strict sense can only be applied in large concert halls or cinemas where the audience is situated further from the loudspeakers than the distance between any two loudspeakers. An illusion which gives a sense of spatial reproduction can be applied to a room by means of two loudspeakers each fed from a separate microphone through separate amplifiers. In radio broadcasting at least two separate transmitters and receivers are required—three channels are desirable. Some methods of recording are capable of carrying two or three channels, such as sound on film and magnetic tape (Refs. 86, 101, 226).

There are methods of obtaining an approach to stereophonic reproduction with a single channel (Ref 87)

When there are two loudspeakers so placed with respect to the listener as to give a transit time difference from 1 to 30 milliseconds, the impression is that of more " liveliness " and " body " (Ref. 125). This is the **pseudo-stereophonic effect,** and to be fully effective requires the listener to be at least 4 feet—and preferably a greater distance—nearer to one loudspeaker than the other. With suitable positioning and relative volumes, the effect is quite surprising.

A more elaborate method for heightening sound perspective utilizes volume expansion in one of the two channels (Ref. 49).

References 6, 13, 14, 36, 49, 68, 86, 87, 97, 101, 112, 160, 161, 162, 163, 169, 195, 203, 216 (giving additional references), 226, 235, 236, 237, 238, also Supplement.

(vi) Sound-reinforcing systems

With this system there are two sources of sound, the original sound and the augmented sound from the loudspeakers. Usually the intensity of the original sound issuing from the stage will be sufficient over a limited area, but the sound energy from the loudspeakers must progressively increase towards the rear of the auditorium. The problem is to select a loudspeaker or loudspeakers with suitable directional characteristics, and to adjust the power output and orientation so that the total intensity level is constant for all parts of the listening area.

The acoustical power required to produce an intensity level of at least 80 db for ordinary speech and music is as follows (Ref. 13A) :

Volume (cu. ft.)	10 000	50 000	100 000	500 000
Acoustical power (watts)	0.014	0.06	0.11	0.5

For full orchestral dynamic range, a maximum intensity level of 100 db is required, and the acoustical powers should be multiplied by 100. In noisy situations the sound level should be 10 to 16 db above steady noise or 20 to 30 db above the noise in relatively quiet intervals.

One of the problems in design is the avoidance of acoustical feedback. It is usual to adopt directional loudspeakers and sometimes also directional microphones. The microphone is mounted as close to the sound source as possible and in a position of low radiated sound intensity.

For the best illusion to the listener the loudspeaker(s) should be mounted so that the sound appears to come from the direction of the original source. The difference in path length between the original sound and the sound from the loudspeaker should not exceed 60 to 80 feet.

The overall response of the whole system should be free from peaks. When feedback occurs, it does so at the frequency at which the system has a peak.

If at all possible, low frequencies should be attenuated so as to reduce the size of the loudspeaker and of the amplifier, and to reduce acoustical feedback. A minimum frequency range for speech is 400 to 4000 c/s (Ref. 196). If orchestral music is to be amplified the whole frequency range should be provided for, although here also some compromise may be necessary.

In small or medium halls a single loudspeaker or bunch of loudspeakers may be mounted 20 to 30 feet above the microphone and somewhat more forward than the microphone. The loudspeakers should be pointed downwards to the audience (Ref. 196).

In larger halls one or more **line-source loudspeakers** may be used. A line-source consists of a number of loudspeakers mounted close together in a straight line. A vertical line-source gives directionality in the vertical plane but not in the horizontal plane. If the line-source is " tapered " in strength so that the sound from each element varies linearly from a maximum at the centre to zero at either end, the directionality is improved. The line-source should be tilted slightly forwards so that the central loudspeaker points directly to the centre of the audience. Better results may be obtained by using two line-sources, a long one for the low frequencies and a shorter one (one quarter the length of the long one) for the high frequencies, with a cross-over network for $f_c = 1000$ c/s (Ref. 197).

In large installations, time delay may be incorporated (Ref. 145, 197).

Multiple loudspeakers

Haas (Ref. 125) has investigated the effects of a single echo on the audibility of speech. He has shown that when the time delay of the echo is from 5 to 30 milliseconds, the echo is not detectable as a separate source if the level of the echo is less than 10 db above that of the direct sound, and that the listener only has the sensation of hearing from the nearer loudspeaker. At greater time delays he determined how loud such an echo could be before interfering with speech (e.g. echo 0 db, 50 msecs, path difference 56 ft.). For echoes at lower levels see Ref. 308. See also Refs. 308, 326, 327.

A typical reflex horn, with an input of 1 watt to the voice coil, gives the following sound levels on the axis (Ref. 73):

Distance	10	20	40	80	160	320	feet
Sound level	100	94	88	82	76	70	db

With a single horn, the axis should point directly to the farthest part of the audience. Horns may be arranged radially to cover a larger audience, allowing 30° for each horn.

For speech reproduction only, a 10 watt amplifier with two horn loudspeakers is sufficient for a crowd of 5000 people in quiet surroundings. For musical reproduction a power of 40 watts with four loudspeakers will be required. It is usual to allow from 5 to 10 watts for each horn loudspeaker. Alternatively a large number of any convenient type of loudspeaker may be used, with the spacing between loudspeakers not greater than 70 feet. When a wider frequency range is required, some dual horn system is frequently used, particularly with open-air orchestral sound reinforcing.

For large crowds, several line-source loudspeakers—see (vi) above—may be used at the centre of the crowd.

References 13A (pp. 292-296), 14 (pp. 406-409), 73, 104, 125, 176.

(viii) Intercommunicating systems

Only speech is to be reproduced, and the usual requirement is merely to have sufficient articulation to be understood. A reduced frequency range is almost universal, while a very restricted range is used in noisy surroundings (see Chapter 14 Sect. 11).

References 1, 13 (pp. 297, 299), 13A (p. 426).

(ix) Background music in factories

If the noise level is comparatively low, the system may be quite conventional, with a frequency range from say 100 to at least 6000 c/s. Loudspeakers should be placed fairly close together to give good coverage of high frequencies. The spacing should be adjusted so that, with the prevailing noise level, the range of each is slightly over half the distance between them.

If the noise level is high, the highest noise intensity is often limited in frequency range at both extremes. In such a case, as an alternative to over-riding the noise, the full frequency range up to 8000 c/s may be used for the music, which may be at the same level as the mid-frequency noise. The low frequency music range may either be used or attenuated as desired. Careful choice of the source of music is required to give the full frequency range without distortion. A conventional A-M receiver is unsuitable, owing to sideband cutting. Many shellac discs, especially the older ones, are unsuitable on account of limited frequency range.

In all cases some form of volume compressor or limiter is required to reduce the volume range, and the music should be selected to avoid sudden large changes in volume.

SECTION 7: DISTORTION IN LOUDSPEAKERS

(i) Non-linearity (ii) Frequency-modulation distortion in loudspeakers (iii) Transient distortion (iv) Sub-harmonics and sub-frequencies (v) Intermodulation distortion.

(i) Non-linearity

Non-linearity in a cone occurs when the force versus displacement characteristic deviates from a straight line. The principal causes of non-linearity are—

1. Insufficient rigidity in the cone,
2. Non-linear suspension, and
3. Non-uniform flux density.

Lack of rigidity in the cone is usually the result of reduction in the mass of the cone to achieve high sensitivity. In addition to the other defects—ragged response and poor reproduction of transients—this has a marked effect in increasing the harmonic distortion over the whole frequency range (Ref. 15 Fig. 3).

All cone or diaphragm suspensions are non-linear to a greater or less extent—their stiffness usually increases with larger amplitudes. The total harmonic distortion of any such loudspeaker is fairly low (of the order of 1%) at frequencies of about 300 c/s and above, and is not appreciably affected by non-linearity in the suspension. As the frequency is decreased, however, the distortion rapidly rises in loudspeakers having non-linear suspension. For example, one 10 inch dynamic loudspeaker with a non-linear suspension gives 10% total harmonic distortion with an input of 2 watts at 60 c/s, and 30% distortion with an input of 10 watts at the same frequency (Ref. 13A Fig. 6.34). On the other hand a good quality loudspeaker is capable of handling an input of 10 watts at 60 c/s with only 3% total harmonic distortion, or an input of 2 watts at 60 c/s with less than 1% distortion (Ref. 36 Fig. 25).

When a loudspeaker is used in a well designed vented baffle, or in a suitable horn, the maximum amplitude of the cone will be decreased and the distortion arising from a non-linear suspension will also be decreased.

Non-uniform flux density up to the maximum amplitude of operation is another source of harmonic distortion. This distortion is usually less than 1% so long as the amplitude of movement is small, consisting of odd harmonics only unless the field is not symmetrical about the voice coil. However at high input levels the distortion is usually severe. In one case total distortion at 50 c/s with an input of 5 watts, was reduced from 9% to less than 5%, by careful design (Ref. 84).

At frequencies above 3000 c/s, the harmonic distortion with single large cone loudspeakers is negligibly small compared with the frequency-modulation distortion—see (ii) below.

With dual loudspeakers, or a dual cone loudspeaker, the total harmonic distortion tends to reach a peak in the vicinity of the cross-over frequency—see Sect 5(iii), also Refs. 36, 116. In one example the total harmonic distortion is over 3% from 500 to 1500 c/s for a 15 inch duo-cone loudspeaker with an input of 5 watts.

The cross-over distortion, and the other forms of distortion described above, all increase rapidly as the input power is increased.

The harmonic distortion below the bass resonant frequency is selective (Fig. 13.54). When the load imposes a greater impedance to the harmonic than to the fundamental, the measured harmonic distortion increases. At a frequency equal to one-half of the bass resonant frequency the second harmonic rises to a peak since the second harmonic frequency is equal to that of the bass resonance. Similarly at a frequency equal to one-third the frequency of the bass resonance, the third harmonic rises to a peak, and so with higher harmonics.

At frequencies above about 1000 c/s all harmonics tend to increase since the impedance of the load to the harmonics is greater than the impedance to the fundamental. This is offset to some extent by the fact that with a triode valve or with most of the commonly used negative feedback circuits, as the load impedance is increased, so the distortion decreases. The nett effect is found by the combination of these separate effects.

A system has been developed (Ref. 152) for automatically recording the total harmonic distortion of a loudspeaker over the a-f range. This is particularly helpful in detecting narrow frequency bands where the distortion is high.

One method of recording the performance of a loudspeaker with regard to non-linear distortion is to measure on the axis the maximum sound level before a certain amount of harmonic distortion is produced. This may be plotted as a curve of maximum undistorted sound pressure as a function of frequency (Ref. 153). Alternatively the distortion may be plotted as a function of frequency for constant electrical input. In all cases the loudspeaker must be tested in the enclosure for which it has been designed.

References 13A (pp. 163-173), 20, 21, 23, 36, 61, 106, 108, 109, 133, 152.

(ii) Frequency-modulation distortion in loudspeakers

The origin of this form of distortion is the **Doppler effect** whereby the pitch rises when the source of sound is advancing towards the listener and falls when it is receding (Refs. 3, 13A p. 18). The effect in loudspeakers is entirely independent of non-linearity. If a loudspeaker has two input frequencies, say 50 and 5000 c/s, the acoustical output can be resolved, like a frequency-modulated wave, into a carrier frequency (the 5000 c/s input) and sidebands (intermodulation frequencies) plus the 50 c/s input. The distortion may be measured in terms of the distortion factor, and so made comparable with the total harmonic distortion when a single input frequency is applied. The distortion factor on the axis is given by

$$d.f. = 0.33\ A_1 f_2 \text{ per cent} \tag{1}$$

where A_1 = amplitude of cone motion (each side of the mean position) in inches at the modulating frequency—say 50 or 100 c/s

f_2 = modulated frequency (variable frequency),

and $d.f.$ = distortion factor in per cent, defined as the square root of the ratio of the power in the sidebands to the total power in the wave.

The distortion factor is proportional to the variable modulated frequency f_2, and is quite small below 1000 c/s. As a typical example take a 12 inch loudspeaker with an equivalent single frequency power input of 0.125 watt to the voice coil—

f_2	1000	2000	5000	10 000	c/s
$d.f.$	0.65	1.3	3.2	6.4	%

This condition applies to an input of 4.2 watts with a loudspeaker efficiency of 3%.

Tests have indicated that the greater part of the distortion above 3000 c/s with single loudspeakers is due to frequency modulation distortion ; this is reduced very considerably by the use of separate high and low frequency loudspeakers.

References 3, 7, 13A.

(iii) Transient distortion

A transient is a waveform, usually with a steep wavefront, which does not repeat at periodic intervals. Any sudden commencement or cessation of a periodic wave has a transient component. For reproduction without distortion the acoustical waveform must be the same as the input waveform.

It is known from the theory of linear dynamic systems of minimum phase shift type that the amplitude response, the phase characteristic and the transient response to various applied waveforms are merely equivalent ways of observing the same inherent performance of the circuit (Ref. 131). Poor transient response leads to fuzzy reproduction with poor definition. Some requirements for good reproduction of transients are

(1) the loudspeaker must respond to the highest audible frequencies,
(2) the loudspeaker frequency response characteristic must be smooth and uniform, and free from sharp peaks and dips,
(3) the loudspeaker must be sufficiently damped, particularly near the bass resonant frequency,
(4) the bass resonant frequency should be as low as possible, and
(5) the phase-shift characteristics of the loudspeaker should be good.

Reasonably good response to transients is obtainable with loudspeakers having a **frequency response** to 10 000 c/s, but better response is achieved with an increase to 15 000 c/s. Generally speaking it is safe to say that a loudspeaker with a smooth response characteristic has a better transient response than one having sharp resonances, but considerable skill is required for an accurate interpretation of the response characteristic (see Ref. 131 for assistance in this direction).

Loudspeaker damping at the bass resonant frequency has been dealt with in Sect. 2(x). With a suitable choice of loudspeaker and enclosure, and with a sufficiently low amplifier output resistance, critical damping or any desired lesser degree is obtainable. There are differences of opinion as to the amount of damping to be used for good fidelity. Some prefer critical damping without any overshoot, others prefer a slight overshoot with its more rapid uptake (e.g. Ref. 174).

At frequencies above 400 c/s, the cone of a direct radiator loudspeaker ceases to act as a piston, and the effective damping at any point on the cone, at any frequency within this range, is likely to be less, and may be very much less than that with a true piston cone.

Horns provide acoustical damping over a wide frequency range, and well designed horn loudspeakers have better transient response than direct radiators.

The **phase-shift characteristics** of loudspeakers are not readily evaluated. Above 1000 c/s the smoothness of the response characteristic appears to be the best available guide to good phase-shift characteristics ; below 1000 c/s the "envelope delay" may be used to supplement the information which can be derived from the response characteristic (Ref. 138).

When a tone burst* is applied, some of the frequencies generated are not harmonically related to the applied signal, and they cause an annoying type of distortion somewhat similar to intermodulation. The ear is very sensitive to this type of inharmonic distortion (Ref. 131).

Ringing at the frequency of the bass resonance becomes progressively less pronounced as the frequency of the applied tone burst is increased away from the resonant frequency.

Testing loudspeakers for transients

There are four methods of testing loudspeakers for transient response—

1. Applied unit impulse,
2. Suddenly applied square wave,
3. Suddenly applied tone burst, and
4. Shorter's method.

Applied unit impulse gives all the information at one test, but is hard to interpret because it causes all the peaks to ring simultaneously (Ref. 131).

The suddenly applied square wave is more selective, since it emphasizes the ringing of the peaks of nearly the same frequency as the applied wave (Refs. 13A, 15, 36, 126, 131).

The suddenly applied tone burst method is capable of giving much valuable information. Ref. 131 gives a theory of ringing and interpretation of results. It seems that those loudspeakers that sound best generally reproduce tone bursts well, although this is better substantiated for the high frequencies than for the low (Refs. 131, 138, 166).

References to tone burst methods : 44, 88, 121, 131, 133, 138, 158, 166, 213.

It has been demonstrated (Refs. 44, 88) that a smooth frequency response curve means a rapid build-up of a transient. The decay characteristic, after the input voltage has ceased, requires further consideration. Each lightly-damped resonant element, when shock-excited by the sudden cessation of the applied voltage, will dissipate its stored energy by radiating sound at its own natural frequency. The method of testing by Shorter (Refs. 44, 117, 121) is to measure, at all frequencies, the response at time intervals $t = 0$, 10, 20, 30 and 40 milliseconds after the cessation of the applied voltage. Good transient response appears to be indicated by

*A tone burst is a wave-train pulse which contains a number of waves of a certain frequency.

1. Attenuation of 35 db at 1000 c/s, increasing steadily to 50 db at 3000 c/s and higher frequencies, for $t = 10$ milliseconds.
2. Further attenuation of about 10 db for $t = 20$ milliseconds.
3. Further substantial attenuations at $t = 30$ and $t = 40$ milliseconds. These should never be negative, even over narrow frequency bands. A negative value may indicate a spurious ripple frequency.

In double-unit loudspeakers, the important frequency region from the standpoint of transient response is near or below the overlap frequency band (Ref. 36).

References 5 (Part 1), 13A (pp. 159-163, 375), 15 (pp. 138-141, 202-212, 332-339), 36, 40, 44, 54 (Part 4), 88 (pp. 59-64), 117, 121, 131, 132, 133, 135, 138, 150, 155, 158, 166, 172, 174, 179.

(iv) Sub-harmonics and sub-frequencies

In addition to harmonic frequencies, a direct radiator loudspeaker produces sub-harmonics, that is frequencies one half, one third, one quarter, etc., of the applied frequency. Of these, the one half frequency is the only one of consequence. It occurs only in limited frequency regions, and it does not occur below a moderate, critical, level. However when this level is reached, it increases rapidly at first, and is accompanied by frequencies of 3/2, 5/2, 7/2, and 9/2 of the applied frequency (Ref. 188). Sub-harmonics are more objectionable to the listener than harmonics of the same percentage, but they require a relatively long time to " build up " and are generally assumed to be not very obvious in ordinary sound reproduction. However one authority states that a good correlation has been found between the number of sub-harmonics produced by the speaker and the quality rating of the speaker as determined by listening tests (Ref. 153).

References 13A (pp. 167-168), 19, 21, 133, 153, 171, 188 (pp. 751-752).

(v) Intermodulation distortion

Tests on the intermodulation distortion of loudspeakers have been described but no general conclusions can yet be derived (Refs. 108, 109, 171).

SECTION 8 : SUMMARY OF ACOUSTICAL DATA

(i) Definitions in acoustics (ii) Electrical, mechanical and acoustical equivalent (iii) Velocity and wavelength of sound (iv) Musical scales.

(i) Definitions in acoustics

Sound energy density is the sound energy per unit volume. The unit is the erg per cubic centimetre.

Sound energy flux is the average rate of flow of sound energy through any specified area. The unit is the erg per second.

The sound intensity (or sound energy flux density) in a specified direction at a point, is the sound energy transmitted per second in the specified direction through unit area normal to this direction at the point. It may be expressed either in ergs per second per square centimetre or in watts per square centimetre.

Sound pressure is exerted by sound waves on any surface area. It is measured in dynes per square centimetre as the r.m.s. value over one cycle. The sound pressure is proportional to the square root of the sound energy density.

The pressure level, in decibels, of a sound is 20 times the logarithm to the base 10 of the ratio of the pressure P of this sound to the reference pressure P_0. Unless otherwise specified, the reference pressure is understood to be 0.0002 dyne per square centimetre.

The velocity level, in decibels, of a sound is 20 times the logarithm to the base 10 of the ratio of the particle velocity of the sound to the reference particle velocity. Unless otherwise specified, the reference particle velocity is understood to be 5×10^{-6} centimetre per second effective value.

The intensity level, in decibels, of a sound is 10 times the logarithm to the base 10 of the ratio of the intensity I of this sound to the reference intensity I_0. Unless otherwise specified, the reference intensity I_0 shall be 10^{-16} watt per square centimetre.

Compliance is ease in bending. It is the reciprocal of stiffness.

See Chapter 14 Sect. 7 for volume range and peak dynamic range.

See Chapter 19 Sect. 5 for loudness and Sect. 7 for sound level and noise.

See also I.R.E. Standards on Acoustics : Definitions of Terms, 1951, Proc. I.R.E. 39.5 (May 1951) 509; "American Standard of Acoustical Terminology", A.S.A. Z24.1—1951.

(ii) Electrical, mechanical and acoustical equivalents

Electrical	Acoustical	Acoustical unit
Capacitance	acoustical capacitance	cm^5/dyne
Inductance	inertance	grams/cm^2/cm^2
Resistance	acoustical resistance	acoustical ohms
E.M.F.	pressure	dynes/cm^2
Impedance	acoustical impedance	acoustical ohms

Electrical	Mechanical	Mechanical unit
Capacitance	Compliance	cm/dyne

(iii) Velocity and wavelength of sound

The velocity of sound in a medium varies according to the relation $V = \sqrt{E/\rho}$ where E is the elasticity and ρ the density of the medium. For any particular medium, the velocity depends also on the temperature (because of its effect on the density of the medium) and the intensity of the sound. In the audible range, velocity decreases with decreasing intensity ; but in the ultrasonic range, velocity increases with decreasing intensity.

Ref. Kaye, G. W. C., and T. H. Laby, " Physical and Chemical Constants " (ninth edition, Longmans, Green and Co. London).

Velocity of Sound in Some Common Media

Medium	metres/sec.	Medium	metres/sec.
Air (dry)	342 (at 18° C)	Steel	4700—5200
Hydrogen	130	Nickel	4970
Water vapour (sat'd)	413	Glass	4000—5300
Water (sea)	1540	Brass	3650
Alcohol	1260	Wood	3300—5000
Aluminium	5100	Rubber	50—70
Copper	3970		

Sound transmission in air at 20°C and 760 m.m.

Frequency	30	50	100	200	400	1000	4000	c/s
Wavelength	452	271	136	67.7	33.9	13.6	3.39	inches
	37.7	22.6	11.3	5.65	2.82	1.13	0.282	feet

Velocity of sound in air = 13 550 inches per second
= 1129 feet per second
= 344 metres per second.

$f \times \lambda = 1129$
where f = frequency in cycles per second
and λ = wavelength in feet.

(iv) Musical scales

Every musical tone has a frequency which is measured in cycles per second.

A **scale** is a series of tones ascending or descending in frequency by definite intervals.

The **octave** is the most important interval ; two tones are separated by one octave when the frequencies are in the ratio 2 : 1.

Each octave is subdivided into a number of smaller intervals. In the **equally tempered scale** the octave is divided into twelve equal intervals (1.0595 : 1) to allow a change of key without retuning. Thus in the equally tempered scale, if any frequency is multiplied by 1.0595 it is raised by one semi-tone ; if multiplied by 1.1225 it is raised by one tone. All normal musical instruments follow the equally tempered scale.

In addition, there are several systems of what is called the **natural scale, or just intonation,** in which for example C♯ has a different frequency from D♭ (Ref. 90 Part 1).

Table 1 gives one version of " just intonation."

TABLE 1

Tone	Interval	Frequency ratio: Natural (just) scale	Frequency ratio: Tempered scale
C	Unison	1 = 1.000	1.000
C♯	Semitone	25/24 = 1.042	
			1.0595
D♭	Minor second	27/25 = 1.080	
D	Major second (= tone)	9/8 = 1.125	1.1225
D♯	Augmented second	75/64 = 1.172	
			1.189
E♭	Minor third	6/5 = 1.200	
E	Major third	5/4 = 1.250	
			1.260
F♭	Diminished fourth	32/25 = 1.280	
E♯	Augmented third	125/96 = 1.302	
			1.335
F	Perfect fourth	4/3 = 1.333	
F♯	Augmented fourth	25/18 = 1.389	
			1.414
G♭	Diminished fifth	36/25 = 1.440	
G	Perfect fifth	3/2 = 1.500	1.498
G♯	Augmented fifth	25/16 = 1.563	
			1.587
A♭	Minor sixth	8/5 = 1.600	
A	Major sixth	5/3 = 1.667	1.682
A♯	Augmented sixth	125/72 = 1.736	
			1.782
B♭	Minor seventh	9/5 = 1.800	
B	Major seventh	15/8 = 1.875	
			1.888
C♭	Diminished octave	48/25 = 1.920	
B♯	Augmented seventh	125/64 = 1.953	2.000
C′	Octave	2 = 2.000	2.000

The frequency of all tones is determined by the "pitch." The international standard pitch is a frequency of 440 c/s for tone A (equivalent to 261.63 for C) whereas the "physical pitch" is a frequency of 256 for C.

The octaves of C in the two examples above are :

CCCC	CCC	CC	C	c^{i}	c^{ii}	c^{iii}	c^{iv}	c^{v}	
16.4	32.7	65.4	130.8	261.6	523.3	1046.5	2093	4186	c/s
16	32	64	128	256	512	1024	2048	4096	c/s

The frequency of any tone may be calculated by multiplying the value of C next below it by the frequency ratio of the tempered scale as given by the appropriate row in Table 1.

References 13 (pp. 334-335), 89, 90, 91, 92, 93, 127.

SECTION 9 : STANDARDS FOR LOUDSPEAKERS

(i) Voice coil impedance for radio receivers (ii) Loudspeaker standard ratings for sound equipment.

(i) Voice coil impedance for radio receivers

The American R.M.A. Standard REC-104 (Jan. 1947) "Moving coil loudspeakers for radio receivers" specifies a voice coil impedance of 3.2 ohms $\pm$ 10% measured at the first frequency above resonance giving a minimum value tested without a baffle. This impedance differs from that specified for sound equipment (see below).

This standard (REC-104) is applied to loudspeakers having a maximum pole piece diameter not over 1 inch.

(ii) Loudspeaker standard ratings for sound equipment

The American R.M.A. Standard SE-103 "Speakers for sound equipment" (Ref. 95) lays down certain definitions and ratings which are summarized below.

The **loudspeaker impedance** (Z_S) is the complex value of the electrical impedance given as a function of frequency, and is measured at the accessible signal terminals of the speaker.

Published data to give magnitude and phase angle as function of frequency.

The **loudspeaker rating impedance** (R_{SR}) is the value of a pure resistance, specified by the manufacturer, in which the electrical power available to the speaker is measured. The value shall be 4, 8 or 16 ohms.

Loudspeakers may also be rated in terms of **power drawn from standard distribution lines.** In this case it is necessary to specify the power and the line voltage.

The **loudspeaker measurement source impedance** (R_{SG}) is the value of a pure resistance to be connected in series with the speaker and a constant voltage source to measure the speaker performance. The value of R_{SG} shall be 40% of the value of the loudspeaker rating impedance R_{SR}.* This is equivalent to operating the loudspeaker from a source having voltage regulation of 3 db.

The **loudspeaker pressure rating** (or pressure "efficiency") G_{SP} is the difference between the axial sound pressure level (referred to a distance of 30 feet) and the available input power level, and is expressed in db. For further details see Chapter 19 Sect. 1(viii).

Standard Test Signal No. 1, as indicated by G_{SP1}, is a loudness weighted signal covering the frequency band from 300 to 3300 c/s. (Refs. 95, 50).

*Some manufacturers (e.g. R.C.A.) apply constant voltage to the loudspeaker, so that R_{SG} is made zero.

The **loudspeaker pressure-frequency response** (L_S) is the variation of the 30 foot axial free-space pressure level as a function of frequency, and is expressed in db. The pressure level L_d at distance d feet may be used to compute the pressure level L_S at 30 feet by the relation

$$L_S = (L_d + 20 \log_{10} d) - 29.5 \text{ db} \tag{1}$$

The **loudspeaker directivity index** (K_S) is the ratio, expressed in db, of the power which would be radiated if the free-space axial sound pressure were constant over a sphere, to the actual power radiated. It is expressed in the form :

$$K_S = 10 \log_{10} \left[4\pi \Big/ \int_0^{\pi} \int_0^{2\pi} (p/p_a)^2 \sin\theta \; d\theta \; d\phi \right] \tag{2}$$

where p_a = axial free space sound pressure, dynes/cm²

p = general free-space sound pressure at the same distance, dynes/cm²

and θ and ϕ are the angular polar co-ordinates of the system, and the speaker axis is at $\theta = 0$.

(When standard test signal No. 1 is used, K_{S1} is the speaker loudness directivity index).

The value of the directivity index (K_S) may be calculated, over the frequency range which determines loudness, by using the following table for a rigid piston vibrating axially in an infinite baffle (based on curves in Ref. 50) :

df =	500	2000	5000	10 000	15 000	25 000	35 000	45 000
K_S =	3.0	3.0	4.0	7.0	10.5	15.0	18.2	20 db

where f = frequency in cycles per second

and d = piston diameter in inches.

When using standard test signal No. 1, the value of the **loudness directivity index** K_{S1} is tabulated for similar conditions to those above :

d =	0	4	8	12	16	20	inches
K_{S1} =	3	4.4	5.7	7.0	8.0	9.0	db

Values of K_{S1} for various types of baffles and horns are given by Ref. 50.

The loudness directivity index K_{S1} is used to evaluate the **loudness efficiency rating,** but the Standard refers to the article by Hopkins and Stryker (Ref. 50) for detailed information. Loudness efficiency rating is defined as the ratio of the total "effective" acoustical power produced by the loudspeaker to the available electrical power. The loudness efficiency rating is equal to

$$LR = 100e = 100 \times 10^{L_e/10} \tag{3}$$

where e = electroacoustical efficiency

$L_e = 20 \log_{10} p_{ax} - 16 - k - K_{S1}$

p_{ax} = effective sound pressure on axis of loudspeaker at distance 30 ft., in dynes/cm²,

$k = 10 \log_{10} W_R$

W_R = maximum available electrical power in watts

and K_{S1} = loudness directivity index (see above).

See Refs. 50, 121 (pp. 697-706).

Loudspeaker efficiency in terms of acoustical power

(Based on SE-103—Ref. 95).

The loudspeaker efficiency (G_{SW}) in terms of acoustical power is the difference between the output acoustical power level and the available input electrical power level, and is expressed in db.

The symbol G_{SW1} indicates the loudspeaker efficiency when standard test signal No. 1 is used.

$$G_{SW} = 10 \log_{10} (W_S/W_0)/(W_{AS}/W_0) \tag{4}$$

$$= 10 \log_{10} (W_S/W_{AS}) \tag{5}$$

$$= 20 \log_{10} p_S - K_S - 20 \log_{10} E_G + 10 \log_{10} R_{SR} + 20 \log_{10} (1 + R_{SG}/R_{SR}) - 16 \tag{6}$$

where G_{SW} = loudspeaker efficiency in db

W_S = total radiated acoustical power, in watts

W_{AS} = electrical power available to the speaker, in watts

W_0 = reference power = 0.001 watt
E_G = r.m.s. value of constant voltage of source, in volts
R_{SR} = loudspeaker rating impedance, in ohms
R_{SG} = loudspeaker measurement source impedance, in ohms
p_S = 30 ft. axial free space sound pressure, in dynes/cm²
and K_S = loudspeaker directivity index, in db.

It is obvious that $(W_S/W_{AS}) \times 100$ is the loudspeaker percentage efficiency, and hence

Efficiency	2	4	6	10	20	30	100	%
G_{SW}	−17	−14	−12.2	−10	−7	−5.2	0	db

The total radiated acoustical power in dbm is given by the electrical power available to the speaker in dbm plus the speaker efficiency in db. For example, if the available electrical power is 10 watts = 40 dbm, and the speaker efficiency is −13 db, then the total radiated acoustical power will be (40 − 13) dbm = 27 dbm = 0.5 watt.

See also " American Recommended Practice for Loudness Testing " C16.4—1942 (Ref. 70).

For loudspeaker testing see also Refs. 13A (pp. 353-376), 121 (pp. 607-609, 661-706), 175 (pp. 292-295), 188 (pp. 768-773).

For Standards for multiple loudspeakers in sound systems see Chapter 21 Sect. 2(ii).

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Additional references will be found in the Supplement commencing on page 1475.