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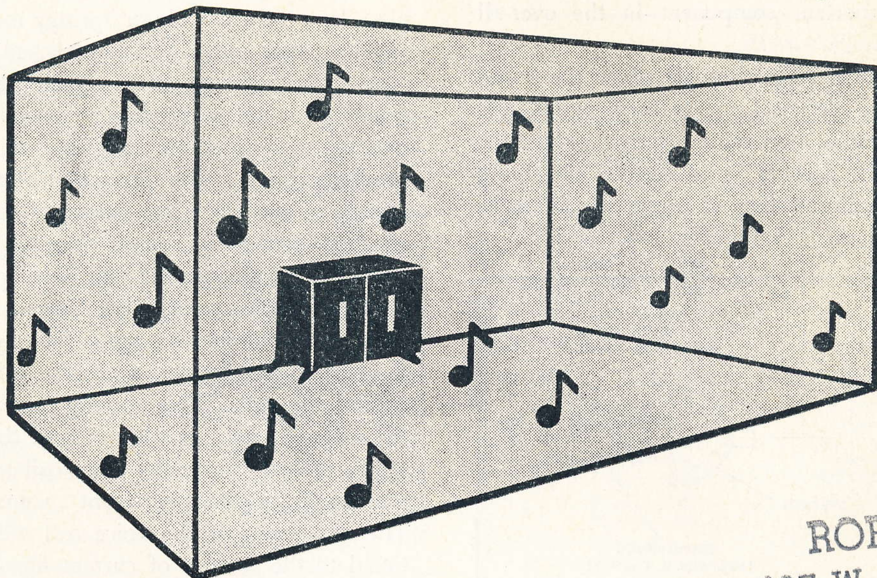
RADIO HOME STUDY



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HIGH-FIDELITY

AND

LOUDSPEAKER SYSTEMS

Prepared By

THE ELECTRONIC EDUCATIONAL DEPT.

SERVICE DIVISION, PHILCO CORPORATION

The audio system of a radio or television receiver is basically composed of three major sections, the audio amplifier, the speaker and the speaker enclosure. Each of these units must be properly designed in order to provide a well balanced system with the desired fidelity and tonal qualities.

Two of these units, the speaker and the speaker enclosure, have been the subjects of many engineering advances in recent years, and these advancements are indicated by considerable improvement in both fidelity and tonal quality. These two units are the subjects of this Home Study Course.

The loudspeaker employed in a radio or television receiver is a very important component in the over-all amplifier system.

The speaker is the final stage through which the sound intelligence in its audio electrical energy form passes, and is then converted to audio mechanical energy. Thus, the primary purpose of the speaker is to convert the audio electrical energy into audio mechanical energy which can be detected by the human ear. Figure No. 1.

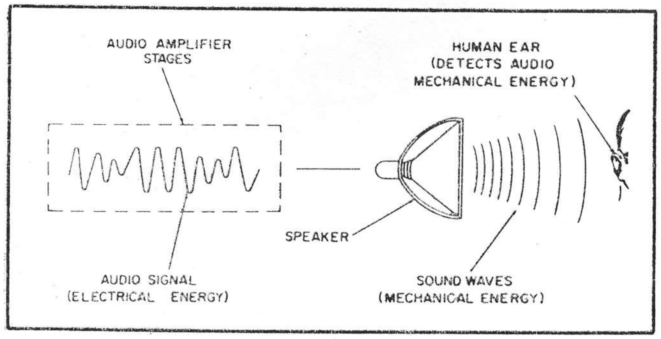

Figure One

There are two basic types of speakers employed in today's receiver or phonograph, the dynamic speaker and the electrostatic speaker. The former has been in use for some time generally, while the later is a type which has been perfected more recently. The later type and its use will be covered further on in this Home Study Course.

The dynamic speaker is subdivided into two general classes, the permanent magnet dynamic speaker and the electro-magnetic dynamic speaker more often referred to as the PM and EM speaker respectively.

The very early type speakers consisted of a paper cone fastened to a diaphragm. This speaker did not prove very satisfactory since under strong current signals being fed to the speaker coil, the diaphragm was pulled against the magnet and a speaker rattle with accompanying sound distortion would result. Figure 2.

To overcome this and other problems presented by this construction, the dynamic speaker was developed, which incorporated a voice coil. The voice coil is a small coil wound on a tube of bakelite or fiber material, and was so mounted that the tube could slide back and forth on the permanent magnet. The voice coil tube is held in

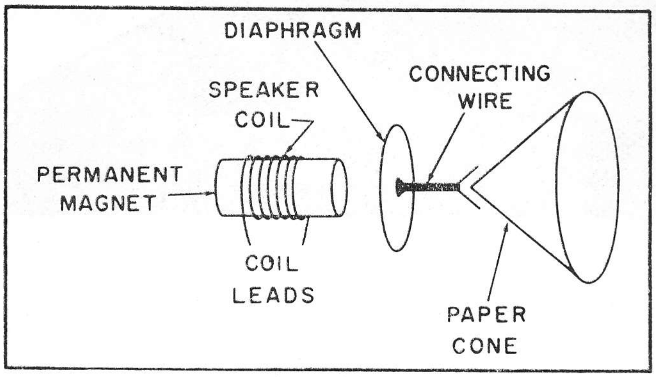

Figure Two

place by a very flexible or springy material called a spider and the voice coil is further attached directly to the paper cone. Variations of current passing through the voice coil would produce a varying magnetic field which would interact with a stable magnetic field produced by the speaker magnet. The interaction of magnetic fields causes motion of the voice coil form and the cone to which it is attached producing sound waves. Thus the permanent magnet dynamic speaker was developed. Figure 3.

When the fluctuating Plate Current (Audio-Modulation) from the audio output stage is fed to the voice coil through the output transformer, a varying magnetic field is set up around the coil. This varying magnetic field reacts with the magnetic field existing around the permanent magnet, which in turn causes the voice coil to move up and down the length of the permanent magnet within limitations. The movement of the voice coil will be directly proportional to the amount of current flowing through its windings, a greater amount of current flow producing a greater movement of the coil and vice versa. Due to the action of the spider and its flexible properties, the voice coil will be in motion on the magnet during current flow, but returns to its original position when the current drops off or ceases.

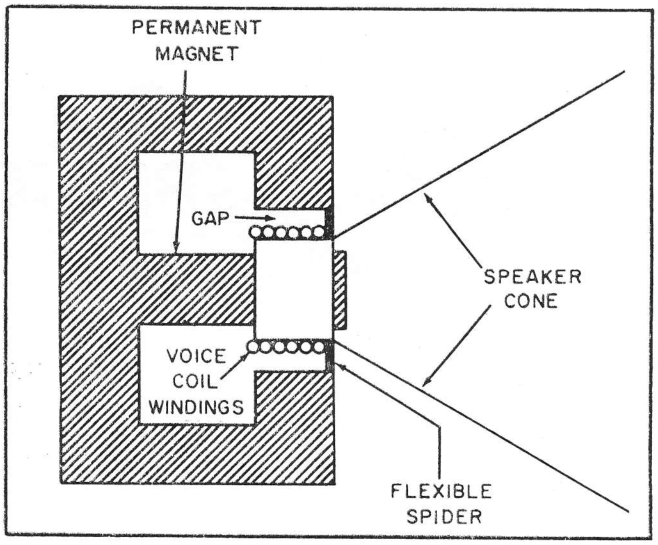

Figure Three

Since the speaker cone is directly connected to the voice coil, its motion corresponds to the motion of the voice coil. Further, the cone's movement will correspond with the audio modulation signal. This movement or vibration will disturb the air and produce sound waves directly related to the variations of the audio signal. This, therefore, is the basic operation of the Permanent Magnet Dynamic Speaker, to convert the audio electrical energy into audio mechanical energy or accoustical energy.

The basic design of the electromagnetic dynamic speaker is very similar with respect to the voice coil, cone, etc. The electromagnetic type differs, however, in that an electromagnet, formed by a field coil, replaces the permanent magnet. Figure 4.

In the electromagnetic dynamic speaker a direct current is passed through the field coil, which consists of a considerable number of turns of wire around a soft iron core. The voice coil is positioned on the projecting end of the iron core and is held in place by the spider or inner suspension. Current through the field coil causes the iron core to become magnetized, while the current applied to the voice coil produces the same action as occurred with the permanent magnet speaker and the audio electrical energy is converted to accoustical energy.

It may be noted that the electromagnet employs a soft iron core while the permanent magnet employs a hard metal core. From the basic study of magnetism, it may be remembered that soft iron will lose its magnetism as soon as the current through the field coil is removed, while the hard steel employed in the permanent magnet speaker will retain its magnetic properties for an indefinite period. The choice of speaker magnet types is primarily governed by cost factors and space requirements since, within certain speaker size limitations, the difference in performance between the two types is negligible. The electromagnetic type speaker requires extra power to activate the field coil, while the permanent magnet type requires only the power applied to the voice coil. This problem of course, is taken into consideration, in the modern design of receivers, by designing the field coil to act as a choke in the power supply filter system. In Figure No. 5 the circuitry of the two types of speakers is illustrated. In the electromagnetic speaker circuit the field coil is employed as a filter choke while the permanent magnet speaker circuit utilizes a resistor in the filter circuit. In some receivers employing permanent magnet speakers, a separate choke may be used to improve filtering action.

Filtering of the B plus supply in an electromagnet speaker circuit must be given careful consideration in view of the possibility of AC hum problems, however, this means of supplying the power to the field coil is commonly used.

The design engineer, therefore, has two factors in mind, in designing the electromagnetic type speaker, a field coil sufficiently heavy enough to handle the power requirements

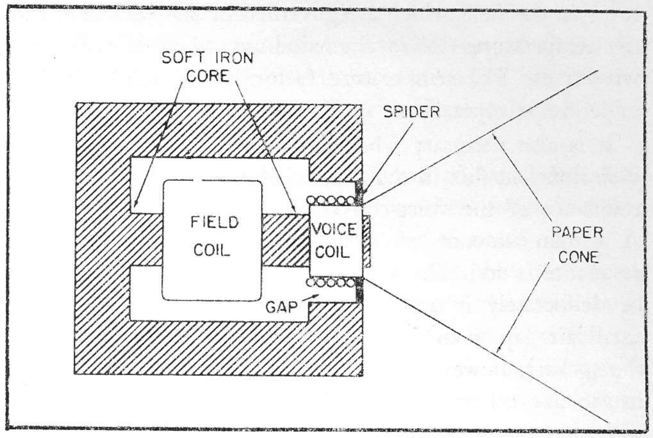

Figure Four

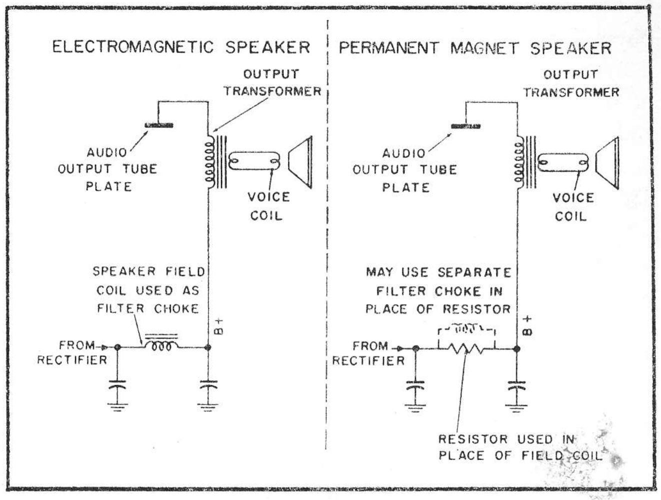

Figure Five

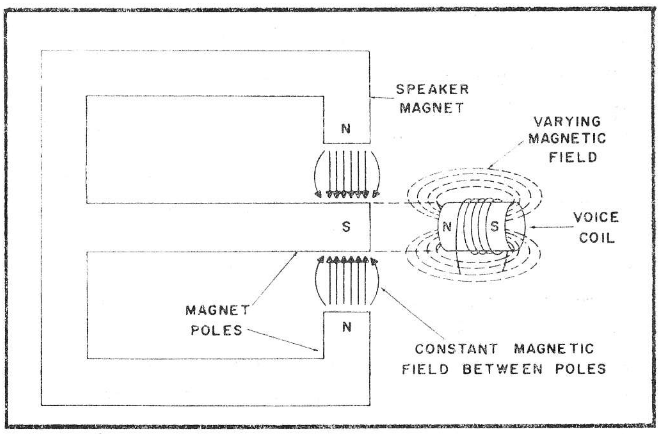

Figure Six

of the receiver and also providing sufficient magnetism for the required speaker performance.

In general, the field coil is designed for a dissipation approximately equal to the electrical power output of the receiver. The design engineer must consider the ampereturns in the field which are governed by the wire size, and the temperature rise in the windings when selecting the wire size. The temperature factor is ordinarily checked under actual operation.

It is also necessary when designing the speaker to provide sufficient flux in the gap, otherwise the low frequency resonance of the voice-coil cone may be overpronounced. A certain amount of over-emphasis in low frequency resonance is desirable in some cases and, consequently, may be deliberately incorporated into the speaker design. A small air gap would of course improve the sensitivity of the speaker, however, there are limitations to the decrease in gap size, otherwise mechanical problems are encountered which tend to introduce distortion.

The voice coil must provide a field which will interact with the field of the magnet and thus cause the voice coil to move a desired distance on the magnet. Therefore, the physical design of the voice coil dictates that it be small both in wire size and in overall size. Due to the diameter of the magnet pole piece employed in the average radio and television speakers which varies from one-half inch to one-inch, the voice coil impedance is usually 3.2 ohms. This impedance will, of course, vary as the speaker is designed for heavy output wattage, however, a 3.2 ohm voice coil has been employed in speakers capable of 12 to 15 watts.

The actual operation of the voice coil, as previously described, depends upon the variations in the voice coil magnetic field caused by the varying electron flow through the coil and the varying magnetic field interacting with the constant magnetic field of the speaker magnet. Figure 6.

From basic physics and the study of magnetism we learned that like poles repel and opposite poles attract. Remember that the position of the voice coil is directly over one of the poles on the speaker magnet. Figure 7. We know further that by applying the left hand rule to the current flow in the voice coil, we can determine the pole positions of the magnetic field formed by the current passing through the coil. Regardless of which direction the voice coil current flows through the coil, two magnetic lines of force will exist, that produced by the coil and that produced by the speaker magnet. If the voice coil field is such that its north pole is in the same direction as the north pole of the magnet, the magnetic lines of force would have a tendency to cause a repelling action, and attempt to push the voice coil off the magnet entirely, which in this case would be in a forward direction on the speaker's axis. Figure 8.

If either the poles of the magnet or the voice coil were reversed, so that the opposite poles were adjacent, north to south, the voice coil direction would be backward and the cone would naturally move in the same direction. The reason for this action, of course, would be due to the tendency of the magnetic lines of force to cause the magnets to align or center the poles as closely as possible.

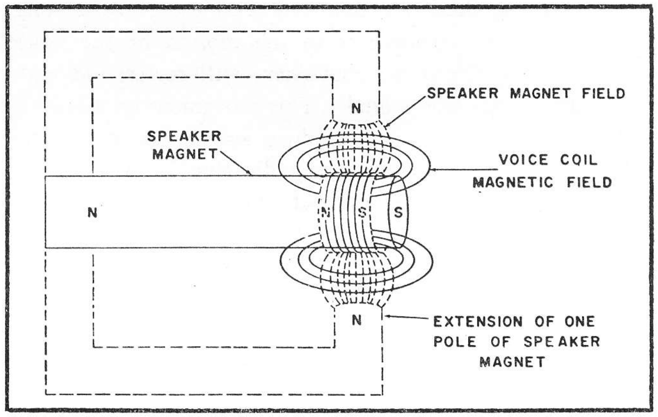

Figure Seven

Movement of the voice coil in either direction will naturally cause a corresponding movement of the cone which moves or disturbs the air mass and creates the necessary sound waves.

It may be well to note at this point, that the ability of the cone to move the surrounding air mass in proportion to the amount of electrical power applied to the voice coil will determine the efficiency of the speaker. Efficiency is defined therefore, in the case of a loudspeaker, as the amount of accoustical output obtained for a given amount of audio signal input. As such it is measured in decibels.

To obtain good efficiency, the design engineer must properly match the various speaker components, the speaker magnet, the voice coil and the cone. The speaker magnet and the voice coil must both be of sufficient size, weight and power to move the cone mass in the desired proportion to the applied audio signal power. At the same time, the voice coil size and weight must not be too great or distortion of the accoustical output may result.

This brings us then to the design of the cone. The proper design of the cone is of great importance in achieving proper tonal balance with respect to fidelity and range. It may be well to note that the design engineer's goal is tonal balance in the speaker with respect to the overall audio system. A loudspeaker may have excellent low frequency response but, due to the inability of the speaker to reproduce sufficient high frequencies, effect a very

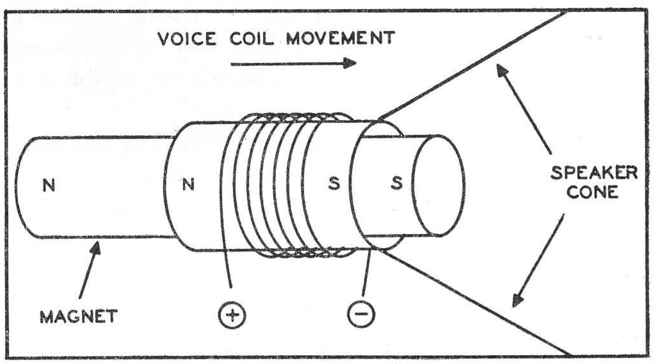

Figure Eight

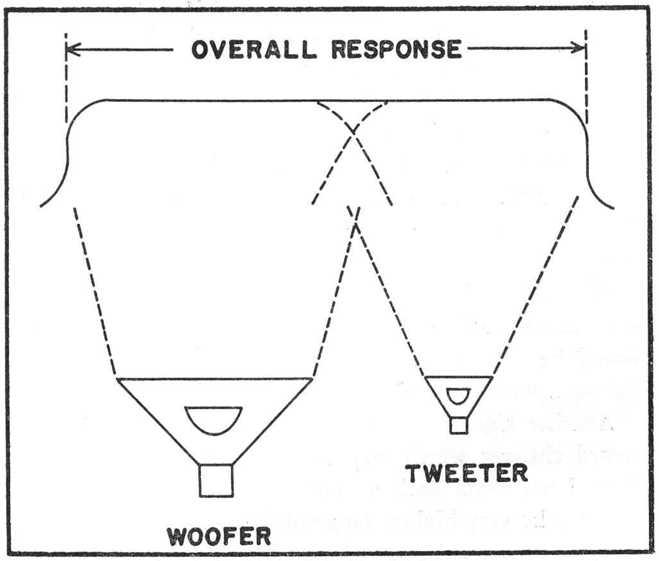

Figure Nine

"booming" sound. In other words, the reproduced sound would contain too much bass characteristic or response. Then again, the speaker may have excellent high frequency response yet, due to the absence of sufficient low frequencies, sound very thin or "tinny". However, if a balance is achieved between the amount of low frequency response and high frequency response even though both may be somewhat limited, the loudspeaker can provide an overall sound range with a great deal of naturalness which is satisfactory to the ear.

The factors which effect the frequency response of the cone are the size, material, shape and the construction or design. The low frequency range of the cone can be extnded by increasing the cone size, however, this results in a sacrifice of some of the high frequency range. Primarily, this is due to a necessary increase of the voice coil mass which has a direct effect on the high frequency range. By going in the other direction and decreasing the size of the cone, the high frequency response can be increased at a sacrifice of the low frequency range. It is therefore obvious that we must select a cone size which in conjunction with the amplifier system and cabinet enclosure will give the desired result.

A partial solution is to use two speakers, one containing a large cone to provide the low frequencies and the other containing a small cone to provide the high frequencies. The small speaker is designed to cover a specific range of high frequencies and is properly matched with the large speaker, designed for a specific range of low frequencies. In this manner the overall range of frequencies is extended to provide the desired response. Figure 9.

This arrangement, however, has its limitations. In view of the space required for two speakers of this type, the cabinet enclosure, into which the speakers will be installed, must be relatively large and this would eliminate many small console and practically all table model receivers. A further consideration might be the cost of the two speakers as compared to the cost of the receiver into which they will be installed, which in some cases, would be prohibitive.

Space limitations within the cabinet enclosure resulted in the development of the co-axial speaker, which consists of a speaker with a small cone area mounted within a speaker with a large cone area and on the same axis. Figure No. 10. The name co-axial means having coincident axes, and consequently was given to this type speaker in view of its design. In some instances the term "whoofer-tweeter" is given to these dual speaker arrangements whether they are two separate speakers or the co-axial type. The term "whoofer" is applied to the large speaker and the small speaker is referred to as the "tweeter".

The type and size of the receiver model is still a factor which will determine the use of the co-axial speaker since the problems of both manufacturing cost and space are not entirely solved by this arrangement.

A different method of treating the problem of cone size is taken in some cases, particularly in the auto radio and small table model radios, where space is critical, in order to extend the frequency response. Ordinarily, because of the space limitations, a speaker with a considerably small cone area would have to be employed. However, by using an oval shaped cone, instead of the conventional round shape, an improvement is obtained. Figure 11. Since a small speaker has a naturally poor low frequency response, an increase in the cone area in this manner improved the low frequency response, the tonal balance and to some degree the efficiency of the speaker.

At this point the limitations of the input signal to the audio system should be considered since they will directly

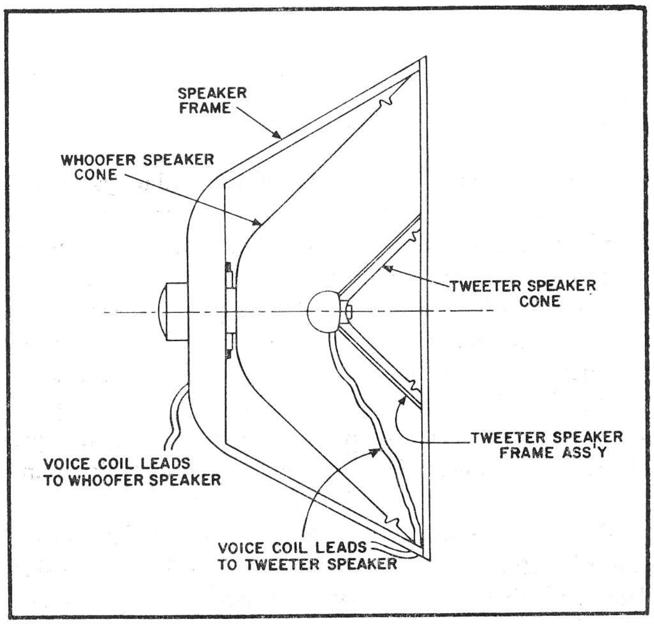

Figure Ten

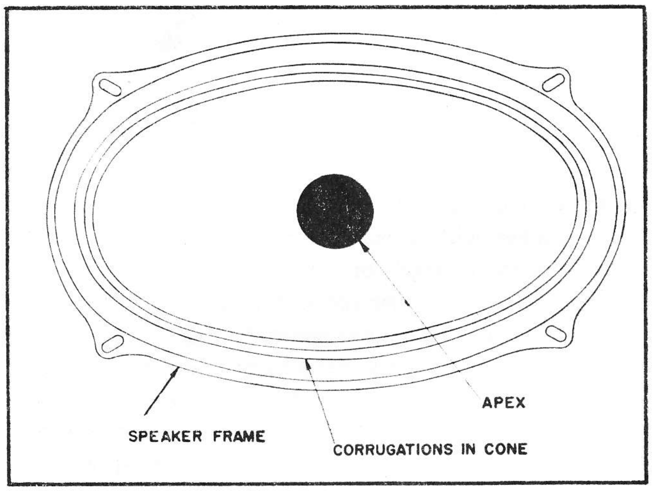

Figure Eleven

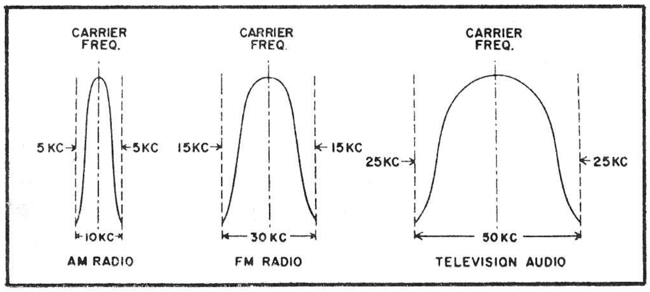

Figure Twelve

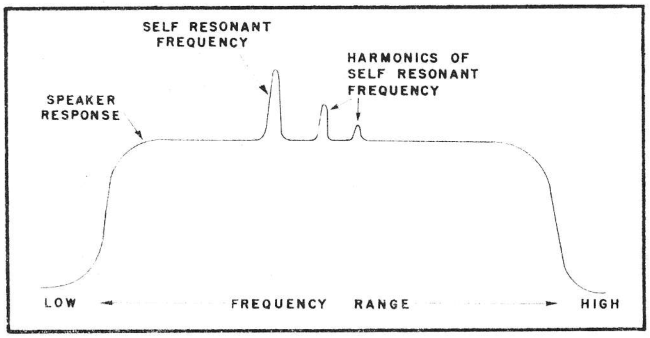

Figure Thirteen

effect the design considerations of the audio system and its components including the speaker. For example, the standard AM broadcast radio station, with the possible exception of some clear channel stations, is limited to a band-width of 10 kilocycles, 5 kilocycles on either side of the carrier frequency for the audio modulation. Clear channel stations are those which are located in a position on the broadcast band of frequencies such that they can extend the side-band frequencies beyond the 5 kilocycle limit, without interfering with adjacent radio station transmission, and are authorized to do so.

The major number of AM broadcast stations in the United States are of the type limited to a 5 kilocycle side-band and, therefore, the maximum possible audio modulation signal transmitted would be 5 KC or less.

The FM radio and television stations, however, are capable of transmitting audio signals which have considerably greater audio frequency range. Figure 12. In many cases, due to technical problems, the stations are unable to utilize their full audio modulation capabilities. Network programs may also suffer a loss of frequency response due to limitations of the cable transmission equipment. These factors are mentioned only to point out the possible sources of loss of response.

The maximum audio modulated signal, therefore, that a standard AM Broadcast radio would ordinarily receive would be 5 thousand cycles or less, provided that it does not incorporate an FM radio section or a phonograph.

Another signal source of course is the phonograph or record changer which may be incorporated in the model. The signal from such a source may contain frequencies up into the very highest range of human hearing.

These then are some of the factors which govern the design of the audio system and consequently, the speaker size and design.

The speaker should exhibit certain characteristics such as uniformity of response, good transient response, negligible amplitude distortion characteristics and self-resonant effects. Very often, when one of these problems is present in the speaker, one or more of the others may also be present, since they are somewhat inter-related.

The frequency response of the speaker should be relatively uniform, whether its range is extended or limited, so that each frequency reproduced by the speaker has approximately the same intensity, otherwise, the naturalness of the reproduced sound will be affected. This problem occurs due to self resonant characteristics of the speaker or cone and consequently, as we mentioned previously, are inter-related. Figure 13.

The design engineer must compensate for any self-resonant characteristics, otherwise a note from a certain musical instrument, reproduced by the speaker and occurring at this resonant frequency, may be many decibles higher in intensity than a note of a higher or lower octave. Harmonics of the resonant frequency may also exist, which have a similar effect.

Similarly, the speaker must have good transient response or the naturalness of the reproduced sound may be affected. To better understand this characteristic, we can employ an audio signal of a given frequency and apply a short burst of this signal to the speaker. In figure 14, the normal burst of this audio frequency signal is illustrated in addition to the resultant audio output being reproduced by speakers having good and poor transient response. Note the reverberations in the wave both before and after the tone burst in figure 14A. The reverberations before the tone bursts are caused by a preceding burst. A speaker exhibiting such characteristics can cause an orchestral instrument such as a drum to sound like a rumble, and if the situation is severe, the reverberations may run into the audio signals which are following, thus causing considerable distortion of the signal. In other words, the clarity of the sound is distorted through mechanical imperfections of the speaker cone. They occur when the cone, because of its mass, material or design, does not immediately stop its movement when the audio signal current applied to the voice coil ceases. Thus, the vibrating motion of the cone continues until other forces dampen and subdue the movement. The spider, to some degree, helps to dampen such vibrations. The spider, as we have mentioned earlier, is actually an inner suspension and its purpose is to position the voice coil with respect to the speaker magnet. A properly designed inner suspension will prevent the side motion of the voice coil against the magnet while permitting freedom of movement of the voice coil backwards and forwards over the magnet and returning the voice coil to its original position after current through the voice coil ceases.

Poor transient response may produce an effect different from the continued vibration problem just discussed. The speaker cone may resist the natural motion stimulated by the audio signal current through the voice coil, so that the cone does not begin to move as soon as current is applied to the coil. Figue 15B. Note how the forepart of the burst is suppressed and must build up gradually. The resultant sound lacks crispness and it is very possible for low intensity audio signals to be lost entirely.

It is very often the case that a speaker having poor transient response will also have poor response uniformity, which is another indication of the inter-relation of these undesirable characteristics.

Amplitude distortion may be present due to the design characteristics of the cone material, cone mass, or general design of the speaker.

To understand what amplitude distortion is, let us apply an audio signal to a speaker exhibiting such distortion.

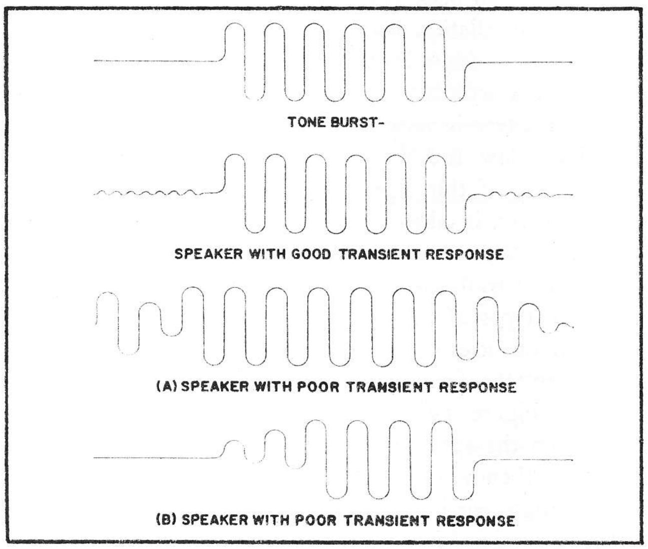

Figure Fourteen

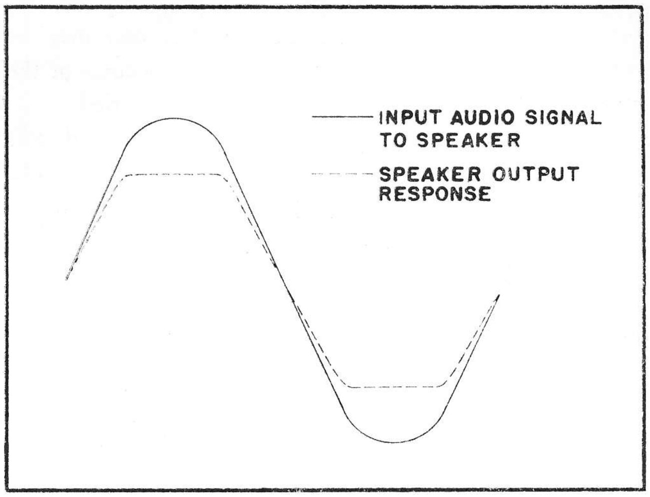

Figure Fifteen

Figure 15. The movement of the speaker cone is ordinarily in proportion to the audio signal applied to the voice coil, consequently, the cone will move a proportionate fraction of an inch for each volt of signal applied to the voice coil, within limitations. For example, let us assume that the speaker cone should move 1/32 inch when 1 volt of audio signal is applied to the voice coil. When 2 volts are applied the cone should move 1/16 inch and, with 3 volts of signal, a movement of 3/32 inch would be expected. However, due to the characteristics of the cone, the cone movement may be less than 3/32 inch at 3 volts. Further since the movement of the cone is greatest at low audio frequencies such distortion is ordinarily more pronounced at low frequencies, than in the middle or high frequency range.

The movement of the cone varies throughout the frequency range. At the low audio frequencies the entire cone is in action as a single unit, all parts in motion in unison, and the action therefore, is somewhat like a piston. However, as the frequency increases, the movement of the cone is in sections, with the greatest cone activity centered around the apex.

Therefore, care must be taken to select the proper material for the apex part of the cone, the apex shape, and the dust cover over the apex to achieve the desired high frequency response.

In the preceding paragraphs we have discussed the major problems which may be encountered in designing the loudspeaker. As we mentioned previously proper selection of the magnet and voice coil is important to the design and also in reducing undesirable characteristics.

Various methods are also employed in the design of the cone to help overcome these problems. Alterations can be made to the cone to help overcome the problems of tonal balance, response uniformity, self-resonant characteristics, transient response, amplitude distortion and efficiency. Since the cone movement varies at different frequencies, the material used in the cone can be changed to either accentuate or suppress the particular frequency or frequencies. The material selected for the cone may be either soft or hard, fiber or paper and the thickness of the material, which effects the cone mass, may be varied.

In some instances, a combination of hard and soft material may be employed in the cone construction. The hard or rigid material is employed in the center cone section, while the outer section is made of the soft or non-rigid material. The purpose of employing two materials of varying rigidity is to reduce or eliminate cone resonance effects.

A very effective and widely used method of improving the response uniformity which, of course, helps to reduce cone resonance effects, is to corrugate or wrinkle the cone material at specific points. Figure 16. These concentric corrugations act, in effect, as flexible hinges for the cone mass on either side of the corrugation. The corrugations serve to break up the resonant wave effect that may be present due either to the cone mass or the rigidity of the material.

The shallowness or angle between the sides of the cone will to some degree effect the rigidity, and in turn the efficiency and response. The shallowness therefore, should be in proportion to the speaker size.

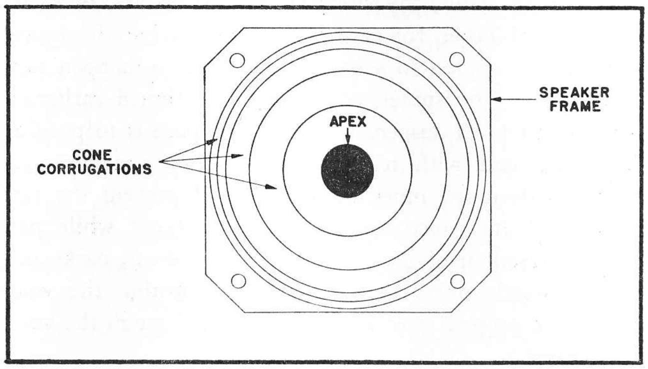

Figure Sixteen

DESIGN OF THE SPEAKER ENCLOSURE

The design of the cabinet enclosure in which the speaker will be installed is of equal importance to the overall audio system design. The cabinet can be designed to help suppress certain undesirable characteristics of the speaker system which, for various reasons, could not be entirely handled by the basic design of the speaker. More often, however, the cabinet enclosure is designed to supplement and add to the desirable characteristics of the speaker and thus improve the response of the overall system. The enclosure is ordinarily considered as a baffle. One of the characteristics of a loudspeaker is that the sound emanating from the rear of the speaker is 180 degrees out of phase with the sound emanating from the front and, consequently, causing a cancellation. Figure 17. This effect is most pronounced in the lower frequencies in view of their longer wavelengths. The length of the sound path or air path of the waves is also important since a shorter air path for the front and back waves to travel and meet will cause a greater cancellation. Such a situation would exist with a speaker without any baffling.

If we increase the length of the air path, from front to back of the speaker, the cancellation effect is reduced. The simplest method of accomplishing this, of course, is to add to the surface area at the top, bottom and sides of the speaker frame or by mounting the speaker on a flat baffle board. A flat baffle of this type, however, must have considerable size in order to satisfactorily accomplish its purpose.

The problem, therefore, is to reduce the size of the baffle, but at the same time achieve the desired results. One method of reducing the comparative size is to fold back the edges to form a box shape, with one end open. Figure 18. Actually, this design forms what might be considered a resonant open ended chamber. The resonance of an enclosure or baffle of this design causes a sharp rise in the response. The frequency, at which this will occur, is dependent upon the construction of the baffle enclosure, but is usually in the low frequency range. This causes a very booming effect in the reproduced sound which of course, further produces an undesirable effect on the naturalness. In Figure 18, the transition from a flat baffle type to the open ended baffle enclosure is illustrated. The open end baffle enclosure is very commonly employed because of its ease of manufacture.

The resonant effect can be reduced by the design of the speaker. Placement of the speaker to one side or off the center axis of the cabinet is commonly employed as a means of reducing the cancellation effect further.

Placing a back on the open end type enclosure will reduce entirely all back radiation from the speaker and prevent any cancellation with the front sound waves. This is known as an "infinite baffle" type of enclosure. An undesirable characteristic of this type of cabinet is that the resonance effect is raised in frequency. Further the frequencies below the resonant point usually decline. In an enclosure of this type, the undesirable effect of resonance, however, is taken into consideration by designing the speaker to compensate for this effect. Acoustic padding on the inner walls also is employed in the "infinite baffle".

Certain types of cabinet enclosures are designed so that at low frequencies the back wave from the speaker is used to augment the front wave, as in the bass-reflex type enclosure. Figure 19. In the bass-reflex baffle, the back wave from the speaker, at low frequencies, is inverted in phase and then added to the forward sound wave in phase. This arrangement is effectively an increase in the efficiency of the speaker at low frequencies. Such an enclosure must be carefully designed to match the speaker in order to

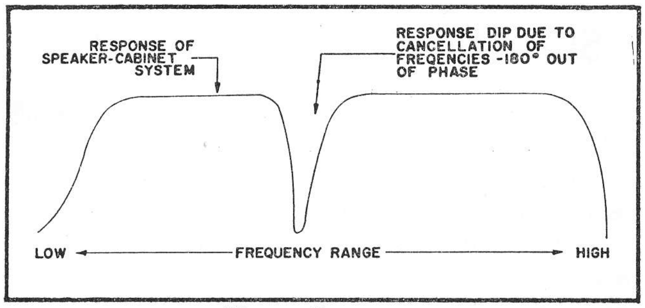

Figure Seventeen

function properly — a properly designed system of speaker and bass-reflex enclosure provides a relatively smooth and extended low frequency response. There is some distortion of the mid-frequency response. However, this can usually be corrected by proper use of damping or sound absorbing material.

There are variations of this type enclosure wherein the back wave of the loudspeaker is inverted in phase and used to augment the forward sound wave.

One type is the labyrinth enclosure, Figure 20, which contains a resonant tube, whose length is one-quarter wavelength long at the resonant frequency of the speaker. Essentially, the quarter wave tube, which is folded in this design in order to conserve space, presents a high impedance to the rear of the speaker at its resonant frequency and a low impedance to the sound at the open end or cabinet vent. Therefore, the resonant frequency is dampened.

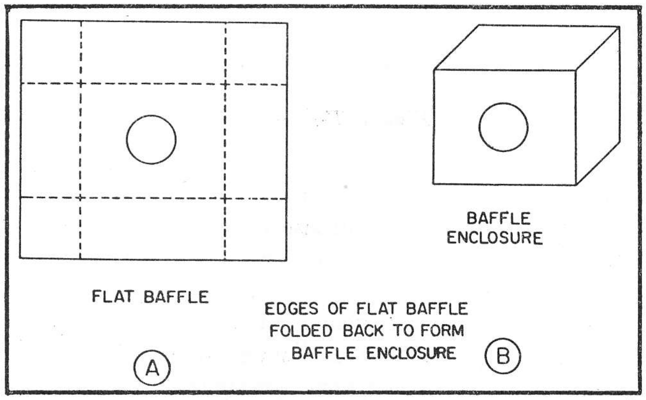

Figure Eighteen

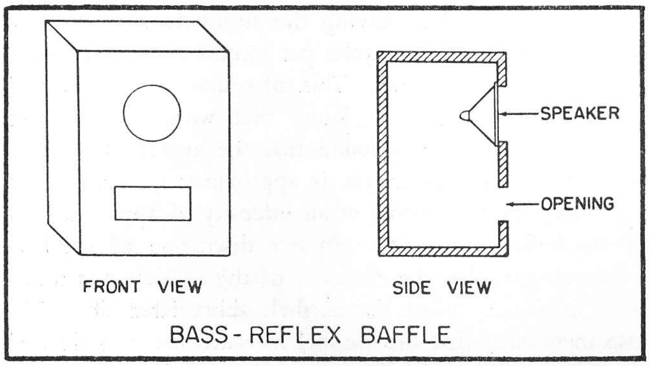

Figure Nineteen

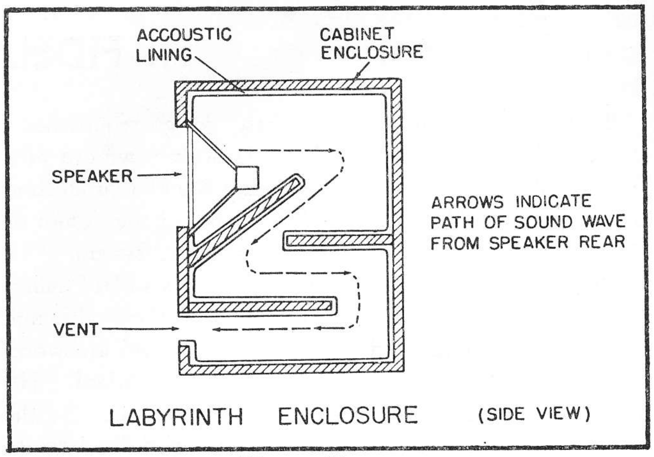

Figure Twenty

The response is increased however at twice the resonant frequency since the tube is effectively a half-wave length long, which causes a phase inversion of the sound to the cabinet opening which then adds to the forward speaker wave. An acoustic lining or sound absorbent lining used on the tube suppresses any resonant effects at the higher frequencies.

Another type of baffle enclosure is the folded horn baffle. Figure 21. In this arrangement, high frequency sound is radiated from the front of the speaker, while the low frequencies are radiated by the horn shaped arrangement of the enclosure. Actually, the addition of a horn to the speaker effectively increases its efficiency or ability to move the air mass at the lower frequencies. Variations of this arrangement may be found where the general shape or design of the horn is altered, however, the foregoing paragraphs deal with some of the basic types of baffle enclosures.

In summarizing, therefore, the primary purpose of the baffle is to reduce or eliminate the cancellation of sound waves which occurs due to their out-of-phase relationship from the front and rear of the speaker. The baffle, however, can be designed to serve a further purpose of effectively extending and smoothing the frequency response and thus improve the overall response which is important to the tonal balance of the system.

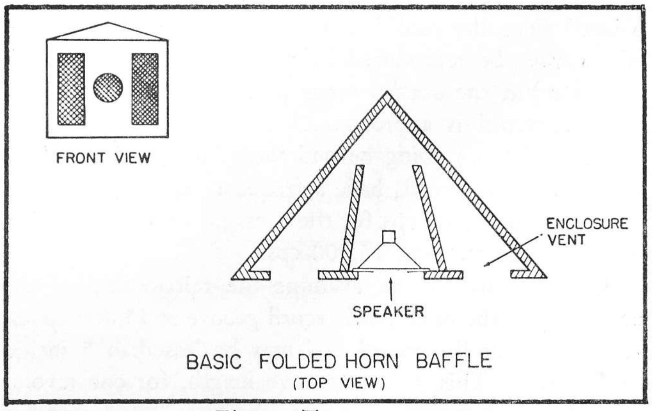

Figure Twenty-one

HI-FIDELITY SOUND

With an understanding of the basic design requirements of both the speaker and the baffle enclosure, we can now discuss the requirements of Hi-fidelity sound reproduction as it pertains to the home audio system and the design of the Philco Hi-fidelity speaker and enclosure system.

Unfortunately, we do not have a concise standard definition of hi-fidelity sound, at the present time, although various groups interested in the field of sound are working toward the establishment of such a standard. Hi-fidelity has different meanings to various people. To the engineer or technician, it probably means specialized equipment, designed specifically for extremely wide-band, low-distortion, audio reproduction. To the average layman, it usually means a system of superior musical reproduction. Perhaps a reasonable definition, which to some degree incorporates both of these ideas might be that Hi-fidelity is audio reproduction of sufficient quality to completely satisfy the individual with its naturalness. This, of course, is by no means a technical definition, but considering the various opinions, even from relatively authoritative sources, a concise definition is not intended.

The current growth of interest in high fidelity stems from several factors. Music and its appreciation is a basic part of the culture of most peoples. In recent years a great wealth of music, from many lands and for every musical taste, has been made available, through the medium of radio transmission and recordings. Engineering advancements in both of these mediums along with the available reproducing systems, which in the past contained technical limitations, made it possible to provide life-like musical entertainment to the public. Also the introduction of the 33-1/3 RPM microgroove records provided another great advancement in recording techniques and also increased the possibility of the average person to obtain high quality audio entertainment.

We have already examined the frequency response limitations of various program sources in our discussion of the design of the loudspeaker. The frequency response limitations of the available recordings should also be considered since the record is a source of audio frequencies which must be reproduced by the audio amplifier system. For example, the normal range of quality standard groove 78 rpm record is approximately 50 to 8,000 cps with a few special types going beyond these limits.

The new LP records have a frequency range of approximately 50 to 10,000 cps for the average, with the extended range records reaching 15,000 cps.

At this point, let us examine the relationship of the needle tip to the modulated record groove at 15,000 cycles. A groove near the record seal may be less than 5 inches in diameter. This gives a groove length, for one revolution of approximately 15 inches which at 33-1/3 rpm has a speed of 8-1/4 inches per second. Thus, one complete cycle of our 15 KC signal would require only .55 thousandths of an inch. Since this one cycle represents excursions in two different directions, we have a diameter, for the curve of one half cycle, of only .275 thousandths. We can readily see that LP needles, having a tip radius of one thousandths will have considerable difficulty tracking and reproducing this 15 KC modulation. Even at the outside groove of a 12 inch LP record, the groove curvature at 15,000 cycles is still too tight for really good pickup action.

Surface noise, which is predominantly composed of high frequencies, provides another limitation of frequency response. It is true, that the newer LP records are remarkably quiet, or free from surface imperfections. However, the noise, small as it may be, begins to assert itself as the high audio frequencies are approached.

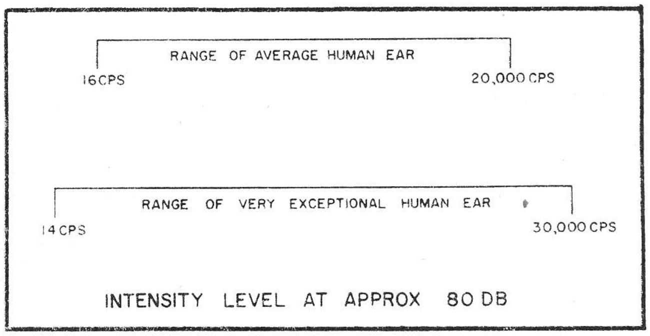

Figure Twenty-two

Dust, dirt and scratches on the record surface, which accumulate through use will also be a source of noise and become more noticeable as the frequency response of the system is raised.

External to the equipment itself, the limitations of the human ear must be considered since this is the organ for which the entire audio system is designed.

Considerable variations are found in the range and sensitivity of human hearing. The exceptional human ear may be considered as having the ability to hear a range of frequencies from 14 cycles per second to approximately 30,000 cycles per second. This, of course, is very unusual and few people can be found that would reach these extremes. It has been found that the average frequency response of the human ear is approximately from 16 to 20,000 cycles per second, at an intensity of approximately 80 decibels. Figure 22. In our discussion of speakers we mentioned that the efficiency of the speaker was measured in a unit called the decibel, abbreviated db.. We also mentioned that this decibel measurement was derived from a comparison of the power input and power output of the speaker, or the ratio of the two power factors. This can also be used to express the ratio of other factors such as sound intensity with respect to its audibility to the human ear. Although the mental impression of loudness of sound is dependent upon the intensity of pressure of the sound wave, the two are not directly proportional as might be expected. Sound intensity is measured by units called microwatts/CM2. Like all physical sensations, there is a degree of stimulation below which there is no reaction. In the field of hearing this level or point is called the "threshold of hearing" and is a sound intensity of one ten-thousandth of a microwatt/CM2. If we should double the sound intensity level of the above example we would expect the mental impression to be that of hearing a sound twice as loud, however, such is not the case. The difference as far as the ear is concerned, would be hardly perceptible. Actually, to make the sound seem twice as loud to the average human ear, the intensity must be increased ten times above the threshold of hearing. If we again wish to double the loudness of the sound we must raise the intensity one hundred times above the threshold of hearing. It can be seen therefore that it is not the difference in intensity levels which produces the sensation of increased loudness but the multiplication factor of 10. This relationship may be recognized as being approximately logarithmic.

The point of the threshold of hearing is therefore the reference zero level for our decibel measurements, and the formula for calculating sound levels in described units is so set up to give a level of 0 db. at this point. An intensity of ten times greater than the zero db reference level would be then 20 db. It may also be well to note that a person with exceptional hearing may be able to perceive a change in loudness of 1 db, where as to the large group of people with average hearing, a change of several db would hardly be noticeable.

In studying the frequency response of the human ear we find that the response curve is very non-linear. For example, if we run a frequency curve graph at the threshold of audibility we find that the necessary intensity for the sound to be heard, below the 1000 cycle reference point, must be steadily increased as the frequency decreases. At 250 cycles an audio level of 20 db is required, while near the lower limit of hearing, 25 cycles, a level of 70 db. As the frequency rises above the reference point of 1000 cycles, the average ear becomes more snsitive to the sound waves until a point in the vicinity of 2500 cycles is reached where the level required is between approximately minus 8 db and minus 10 db. Going beyond this frequency, the sensitivity of the ear rapidly falls off until at 10,000 cycles the sound level must be raised to approximately 10 db and above 17,500 to 20,000, only the most exceptional ears are capable of perceiving any sound. Figure 23.

We find therefore, that the necessary sound intensity varies greatly in order for the average human ear to perceive the sound at various frequencies. However, if we raise the intensity reference level at 1000 cycles to approximately 80 db we find that the response curve is much more

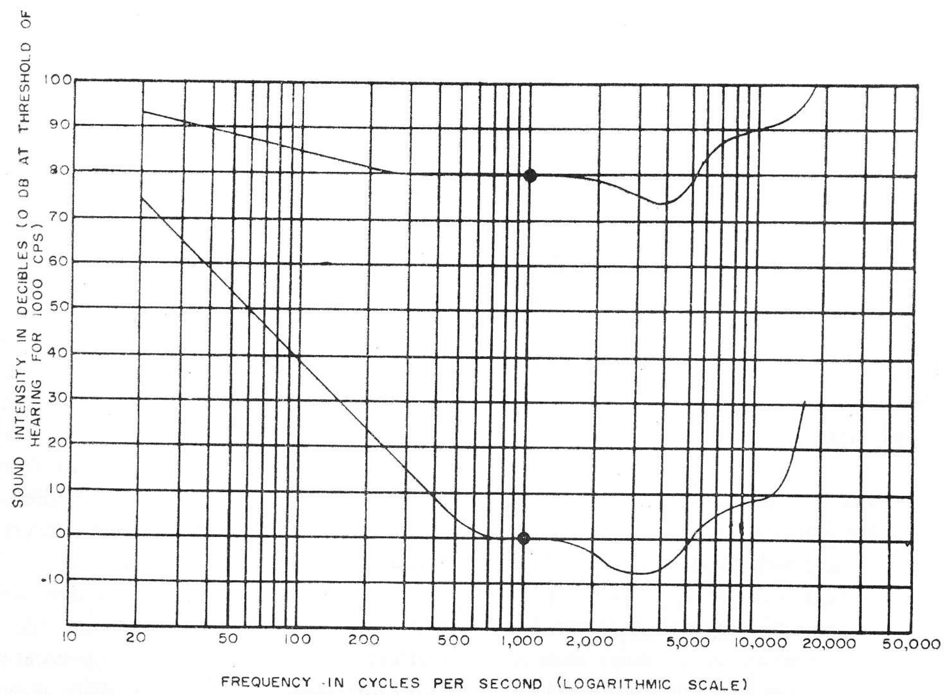

Figure Twenty-three

linear. With a level of 80 db at 1000 cps we find that we require an approximate intensity of 90 db at 30 cps which compared against the "threshold" curve at the same frequency shows a difference of only 25 db.

From the above information therefore we can realize the peculiarities of human hearing and how important this factor is in the reproduction of sound and in the requirements of our high-fidelity equipment, particularly at various frequencies.

There is another characteristic of human hearing which is equally important and that is the tendency of the ear to "mask" or suppress a sound of lower intensity when a sound of greater intensity is present. For example, in the presence of heavy machinery in a factory, we must shout to make ourselves heard over the noise level of the machinery. Yet at the same time the sound waves from our voice would reach the ears of the listeners at a normal speaking level. Due to the masking action of the ear however, the sound waves from our voice would be suppressed in favor of the higher intensity level of sound emanating from the machinery. It has been found by experience and research that this masking can be eliminated by keeping the desired signal, or wanted sound, at an intensity of 40 db above the noise level. Sound and noise surveys have shown us that the noise level of an average city home lies between the limits of 30 to 44 db. This explains why the usual radio-phonograph, as played in a city residence, is operated at a level of approximately 66 to 75 db. When this same radio-phonograph is operated at an average level of 50 db in a room with a noise level of 45 db, all sounds lower in intensity than the noise level are masked. How this affects the perceived frequency range can be easily illustrated.

In Figure 23 we have already seen that the ears sensitivity falls off at both the upper and lower ends. Consequently, in a room with a noise level of 45 db, the frequencies below approximately 300 cycles and above approximately 6500 cycles fall below the room noise level, and therefore would not be clearly apparent to the ear. Figure 24.

It should be noted that in Figure 23 the intensity was varied in order to bring the sound level up to the threshold of hearing. In Figure 24 the sound intensity is maintained at a level of 50 decibels at all frequencies. For this reason the response curves in the two graphs appear to be reversed.

Now, if we raise our intensity level of the radio-phonograph to 78 db, we find that the entire audible response range lies above the masking level or noise level of 45 db in the room, and we can then enjoy the entire range. Figure 25. This then can be considered as a good level for the faithful reproduction of sound or as one of the criteria for the high fidelity system.

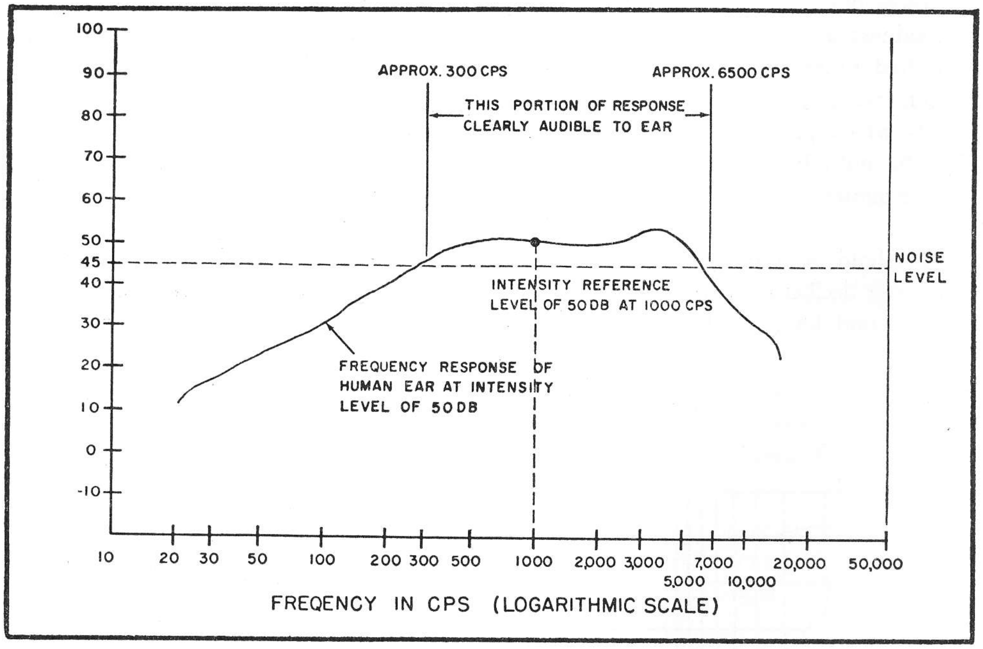

Figure Twenty-four

This brings us to the dynamic range of sound. The dynamic range is the difference in intensity between the lowest sound and the loudest sound of any program or musical selection. Immediately in front of a symphony orchestra, the dynamic range is in the order of 98 to 100 db. To the average listener in a concert hall, however, the dynamic range is about 86 db. A good FM broadcast of the same musical selection in a live concert would reduce the dynamic range to about 76 db. An LP recording may further reduce the dynamic range to approximately 65 db. This reduction of sound intensity range is done deliberately. Large intensities are reduced to prevent overloading of the audio system and also to prevent intergroove

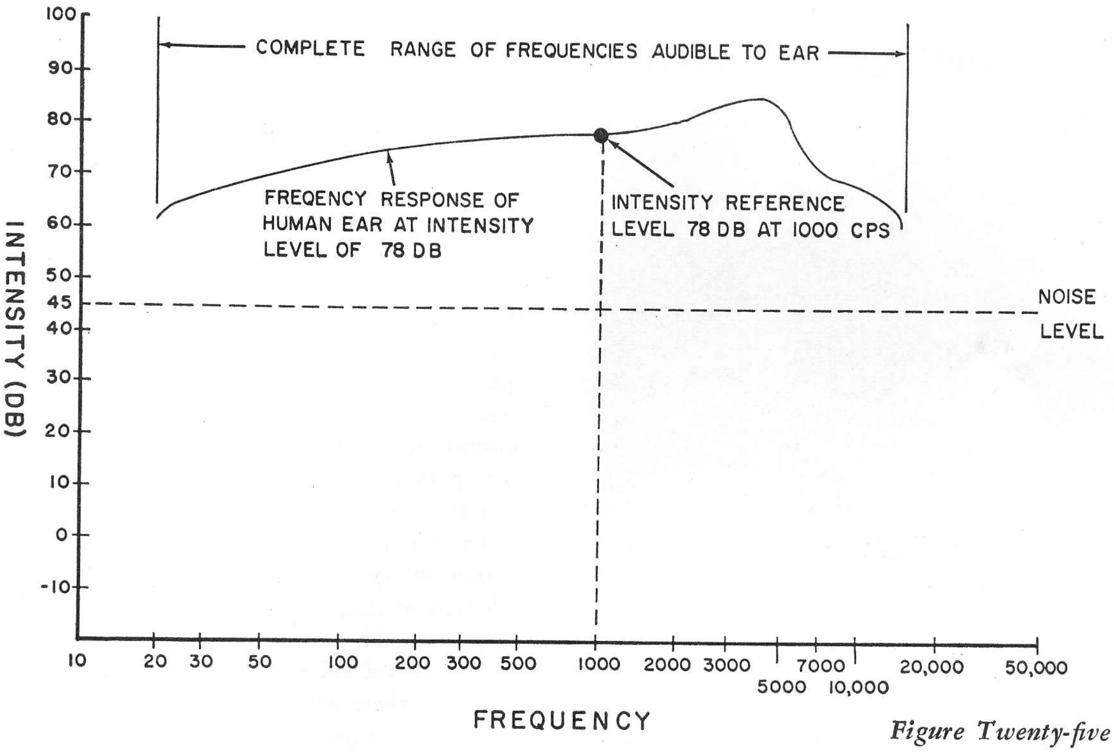

Figure Twenty-five

modulation in the manufacture of the record. Soft passages are increased so as to over-ride the record's noise level. These two actions will naturally reduce the difference in intensity between the loudest and softest passages, thereby reducing the dynamic range. This reduction of the dynamic range has a definite effect on the realism of the final reproduction. For example, listen to and compare two recordings, one containing music performed by a full symphony orchestra and the other containing an instrumental solo. The difference then becomes apparent in that the solo, having less natural dynamic range, would require little or no compression of the intensity and therefore, would have a much more live sound than the full orchestra which may require compression of the volume.

The dynamic range, which has been reduced by compression, can be increased by a system known as volume expansion. The amount of expansion, in order to be effective, should be the exact equal of the amount of compression used when the selection was recorded. It should also act instantaneously without introducing distortion. Unfortunately none of the present methods accomplish this without causing serious distortion.

The dynamic range is also effected by the noise level of the room in which the musical selection is heard. Let us assume for a moment that we are listening to a symphony orchestra whose sound intensity varies from 20 db for the softest passage to 96 db for the loudest sounds. The dynamic range therefore is the difference between the two limits, or 76 db. When this is reproduced in our listening room, all sounds below the room noise level of 45 db are effected by the masking action of the ear as we have previously discussed. This leaves us with a fully perceptible range of approximately from 45 db to 96 db, or a dynamic range of only 51 db. Obviously by reducing the ambient noise level of the room the perceivable dynamic range can be extended.

One more very important item must be considered and that is the usable range of sound. Of all of the various orchestral instruments the lowest frequency that would be reproduced would be approximately 17 cycles such as can be produced by a pipe organ. However, a frequency this low is not particularly pleasing to the ear since it is very harsh or guttural and consequently would rarely be used. The lowest note of the piano is approximately 27 cycles.

The highest frequencies that may be reproduced would be those of the violin, oboe and a few other instruments which may reach frequencies of approximately 16,000 to 18,000 cycles. Human voice frequencies, both male and female, fall between these lower and higher limits. We may therefore consider that a range of between 20 to 25 cycles to approximately 18,000 cycles in our response system should be sufficient to handle all of the various frequencies that would be reproduced and therefore may be considered as our usable range.

We have now discussed the characteristics of the speaker, the speaker enclosure, the characteristics and limitations of the originating sources of our reproduced sound, and the characteristics and limitations of the human ear. Consequently, we now have a realization of the requirements for a high-fidelity system. This brings us to the discussion of the speaker and enclosure system in the Philco "High-Fidelity "Phonorama" model radio-phonograph. Figure 26.

PHILCO PHONORAMA – MODEL 1758

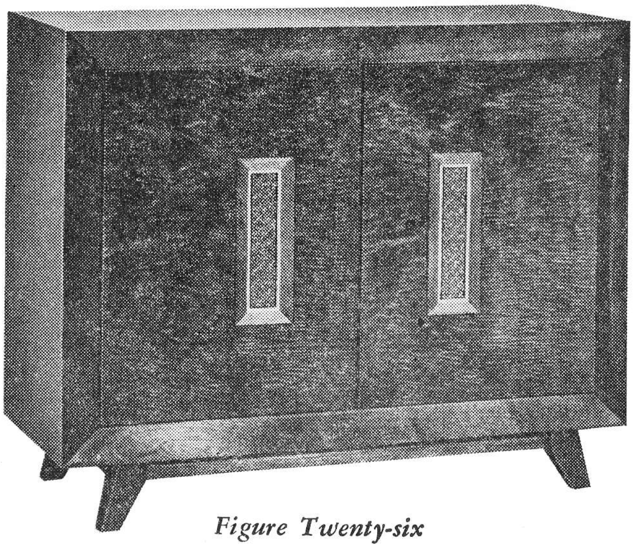

Figure Twenty-six

In the beginning of our discussion of high-fidelity we mentioned that the many engineering advancements in recent years in the sources of the sound, the medium for transmitting it and the means of reproducing it have made high-fidelity possible. However, two factors still remained which prevented wide acceptance to high-fidelity, the elaborateness of the reproducing equipment and the directly related high cost. Until recently the equipment consisted of a number of pieces, with perhaps one section containing the radio portion, another the amplifier system, another the phonograph and still another the speaker and the speaker enclosure. Various attempts were made to house several units together to reduce the problem of space somewhat, however the equipment still remained bulky and costly. Philco's engineering aims therefore, in the design of the Phonorama, were to provide a true high-fidelity unit, with an audio frequency range that extended to the usable limits of hearing and which provided a completely high-quality, life-like, sound reproduction. Most important, the entire unit was to be housed in a single cabinet enclosure with physical dimensions comparable to the radio-phonograph combinations with which we had all been familiar for many years and which fitted in nicely with the decor of the average home. Such a design would of course reduce the cost factor considerably.

Previous Philco publications have covered the phonograph and the r-f and audio amplifier sections of the Philco Phonorama. We are interested primarily in the design of the acoustical system comprising the speakers and the cabinet enclosure. In passing, however, it is well to note that since the Phonorama was built as a complete unit, that is the receiver, the speakers, the record changer, the acoustical lens and the cabinet were all designed as a complete audio reproducing system and not as separate units. Each individual unit had to receive careful design consideration in order that it would be matched perfectly to the other sections and thus contribute to the overall audio response.

The whoofer or bass speaker employed in the Phonorama is basically a 10 inch speaker with a voice coil impendance of 3.2 ohms. Figure 27. In order to achieve proper tonal balance over the desired wide range of frequencies, the size of the magnet and voice coil were increased to maximum size for a 10 inch speaker. The magnet generally employed in a regular permanent magnet speaker is considerably less. The magnet employed is a 10 ounce alnico 5 magnet while the size of the voice coil is 1-1/4 inches in diameter. In our discussion of the design of speakers earlier we pointed out that proper selection of magnet size and voice coil were important in relation to the response and the efficiency. If either the magnet or the voice coil were increased beyond this optimum size there would be a severe power loss, particularly in the low frequency response.

Careful consideration was given to the design of the cone. The cone of a speaker, as we pointed out earlier, must be shaped so that it has maximum ability to move the surrounding air mass while at the same time providing the desired tonal response. The curvature of the cone therefore is of a specific design. The apex has a sharp angle while the outer edge of the cone is flared. Figure 28. The purpose of this design is to smooth out the mid-range frequencies, while the large size of the voice coil is intended to smooth out the low frequency response.

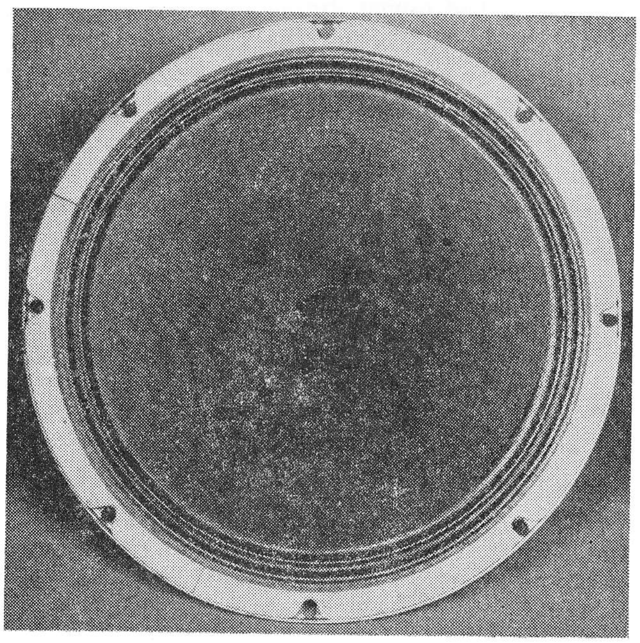

Figure Twenty-seven

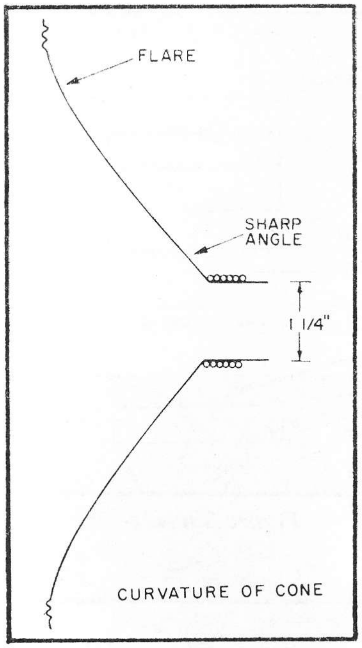

Figure Twenty-eight

The outer edge of the cone is corrugated to permit the cone to vibrate more easily at low frequencies. In order to prevent over emphasis of the low frequencies due to these corrugations and the large voice coil, a visceloid coating is applied to the corrugations. This coating adds to the cone mass and thus slightly retards the movement of the cone so that the low frequency response is not accentuated to a greater degree than the mid range or the high range of the response.

A slotted baffle or Acoustic Lens is employed in the Phonorama. The size of the opening in the lens is extremely critical and therefore must be of an exact size in relation to the size of the speaker, otherwise the purpose of the acoustical lens would be defeated. When used with a 10 inch speaker the acoustic lens must be 8-1/2 inches long and 1-3/4 inches wide. Perhaps the greatest advantage in using an acoustical lens is because of its spatial distribution when used with a shallow enclosure. In Figure 29A the acoustic lens is compared to a pin-hole camera to illustrate the action of the lens. For light to strike the extremities of the photographic plate it must enter the camera at a relatively wide angle. Conversely, the sound waves passing the acoustic lens are distributed over a very wide angle similar to the paths of the light waves entering the camera. Figure 29B.

To further illustrate, if a person were to stand at the axis of a speaker mounted on a plain baffle, he would, of course, hear most of the frequencies being reproduced with negligible loss of the high frequencies. At a position off the axis of the speaker some of the highs would be lost to the listener, Figure 30. To clarify this, it must be remembered that at various frequencies the cone vibrates in a different manner as we mentioned earlier in the discussion of the design of the cone. At very high frequencies the cone vibrates in sections with the portion of the cone nearest to the apex vibrating with the greatest intensity. Thus at higher frequencies, the large speaker will have a tendency to beam these frequencies along the axis of the speaker and, therefore, decrease the spatial distribution at the higher range of the speaker.

Looking at the response curves in Figure 31, it can be seen how the acoustic lens smooths out the frequency response over a wider range and thus aids in overcoming the loss of the highs. When the listener is on the axis of a speaker without an acoustic lens, some of the high frequencies may actually be over accentuated. When the listener is off the axis of a speaker without an acoustical lens, some of these frequencies are lost as we have already stated. However, note how the acoustical lens levels the response over a wider range so that there is little or no change at the higher frequencies.

Further, the spatial distribution achieved by the acoustic lens is at a greater angle than a regular baffle and consequently, there is no beaming effect of the high frequencies.

The cabinet enclosure presented a design problem to the engineering department since the enclosure had to house the receiver, phonograph, the speaker and the acoustic lens. It may be remembered that in our discussion of the baffle enclosure we noted that it is a particularly important factor in smoothing and extending the frequency response in addition to the overall response of the system. It may be well to note that the response of the Phonorama, at 10 watts output, has only a 1% distortion factor.

The depth of the cabinet enclosure affects the overall response and consequently the speaker enclosure had to be shallow in order to obtain optimum performance from the acoustical lens. The cabinet front was designed as a smooth flat surface to prevent response distortion. The thickness of the cabinet or baffle affects the response and it was found that a thickness of one-half inch was the maximum for optimum results.

In our earlier discussion of baffles we noted that one of the methods employed to reduce the cancellation effect caused by the sound waves from the speaker rear was to mount the speaker off the center axis of the enclosure. For this reason, the speaker and acoustical lens are mounted off the center axis of the cabinet in the Phonorama. In this manner the cancellation that would occur in the mid-frequency range is considerably reduced. Figure 32.

The design features so far provide a response range which is fairly broad and extends from approximately 20 cycles to over 12,000 cycles. However, as we have already noted the usable audio range of frequencies extends beyond 12,000 cycles to approximately 17,000 cycles. A tweeter or high frequency speaker is therefore necessary to reproduce the frequencies this high in the response.

A new type of tweeter speaker is employed with the Phonorama high-fidelity model radio-phonograph. It is known as an electro-static speaker and its construction is similar to that of a condenser. Consequently, it is sometimes known as a condenser speaker. The high frequency range of the speaker is outstanding and extended considerably above the previous limitations of most high-fidelity systems. Figure 33.

As was mentioned previously its construction is not unlike that of a condenser since it consists of two metallic plates separated by a thin insulating film acting as a dielectric. One plate is made of aluminum and is rigid. The shape of this plate is semi-circular. The surface, however, is not smooth but is formed into 16 small flat segments, while the entire surface area is perforated. The perforations prevent back pressure on the diaphragm which would result from the vibrating movement of the front plate and which, of course, would introduce mechanical distortion problems.

The front plate consists of a pure gold vapor deposit upon a polyester plastic film which forms the dielectric. The characteristics of the plastic film are that it will not shrink or stretch and has almost no mass. It is fitted across the segmented rigid plate and held in place by a spring loaded metal bar located at the back of the rigid plate. Figure 34.

The result of this mechanical construction is equivalent to 16 tweeter speakers arranged to form a semi-circle thus providing almost 180 degrees of spatial distribution of the sound.

Due to the light mass of the diaphragm material or gold vapor deposit and plastic film, the transient response is excellent. The frequency range of this speaker extends upwards from a frequency of approximately 7,000 cycles. Consequently, when this speaker is incorporated into the Phonorama, the high frequency range is extended to over 20,000 cps. Figure 35.

The actual operation of the speaker is by the varying of electrostatic fields. A polarizing voltage is applied across the two plates of the speaker which sets up an electrostatic charge between them.

In the study of basic electronics we may recall that when a DC voltage was applied across the plates of a condenser, one plate assumed a positive charge and the other, a negative charge depending upon the current flow. The charges set up a stress between the two plates so that they were attracted to one another. Figure 36. A familiar demonstration of this effect, it may be remembered, was to suspend two small metal plates on wire conductors in a glass jar. A battery was connected across the wire conductors which fed through the lid of the jar. As soon as the battery voltage was applied, the two pieces of metal would immediately swing together and adhere. Removal of the battery voltage and shorting the wire conductors would cause them to drop apart. The application of voltage to

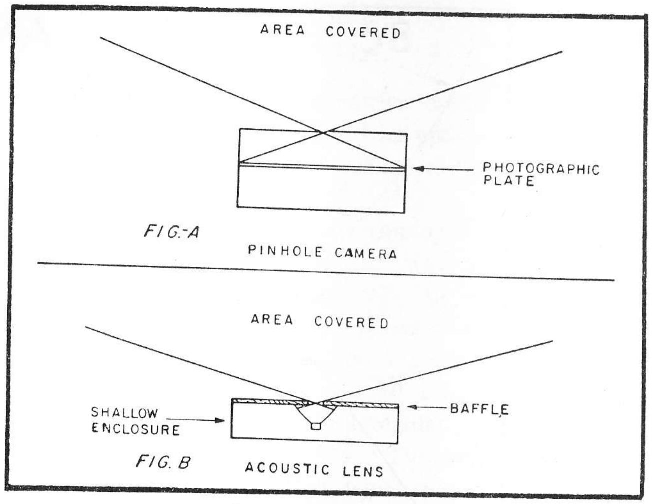

Figure Twenty-nine

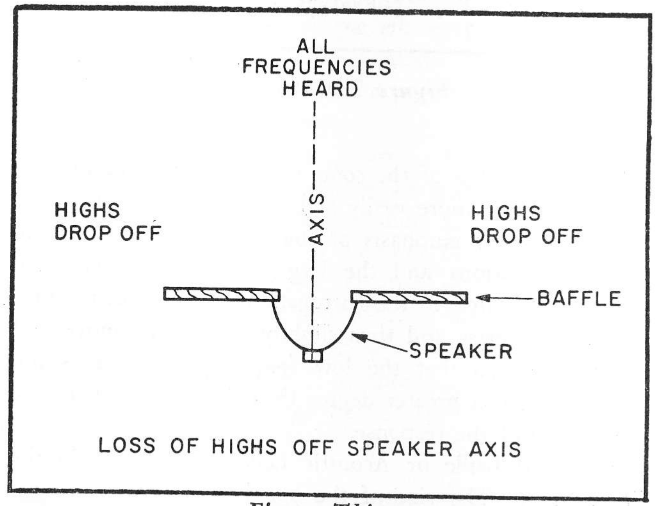

Figure Thirty

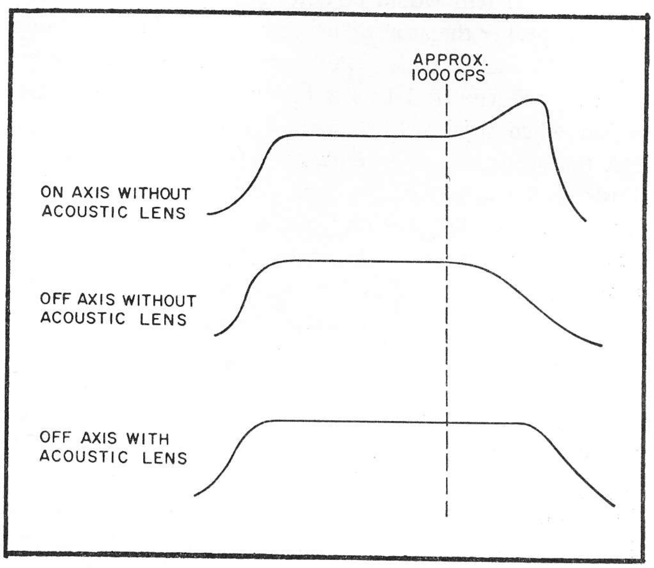

Figure Thirty-one

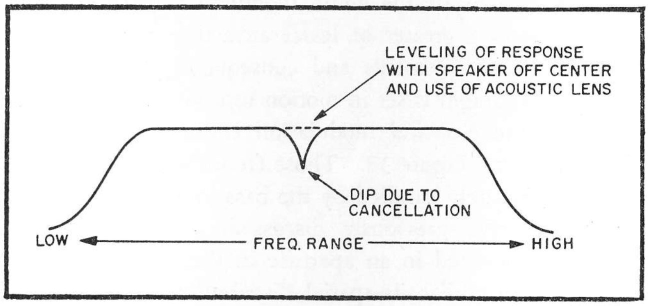

Figure Thirty-two

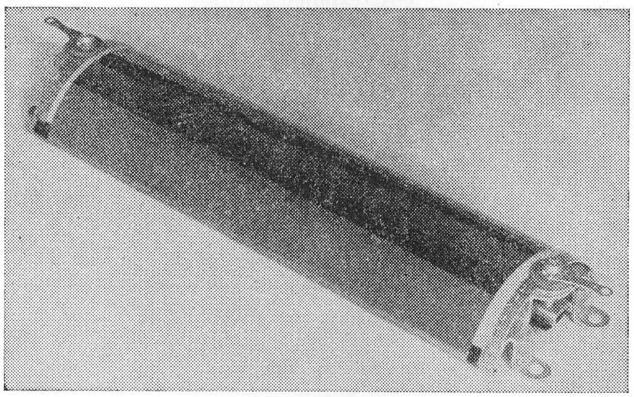

Figure Thirty-three

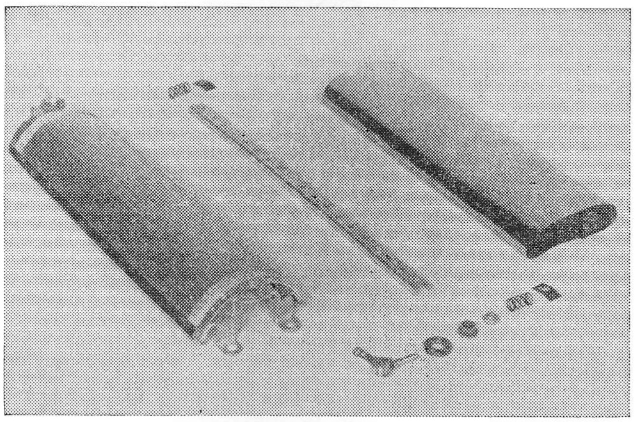

Figure Thirty-four

the two plates in this manner actually formed a condenser with the air between the plates as the dialectric, and since neither plate was fixed in position they were able to adhere to one another. This is the same principle by which the electrostatic speaker functions except that one of the plates is in fixed position.

A polarizing voltage for the speaker is obtained from the plate of one of the tubes in the push-pull audio output stage. This voltage passes through a 4.7 megohm resistor to the terminal on the speaker attached to the diaphragm. The rigid frame of the speaker is connected to ground, and since capacity exists between the two plates they form a condenser. The resistor and the capacity of the speaker form an integration network that filters the fluctuations in the plate voltage, causing it to appear across

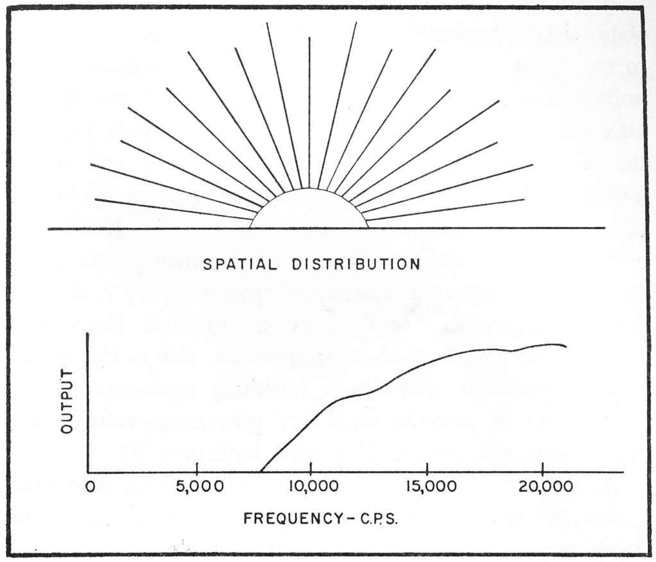

Figure Thirty-five

the speaker as a constant value DC voltage. This polarizing voltage sets up the required stress or tension between the two speaker elements so that a change in the electrostatic field will cause the diaphragm to move back and forth.

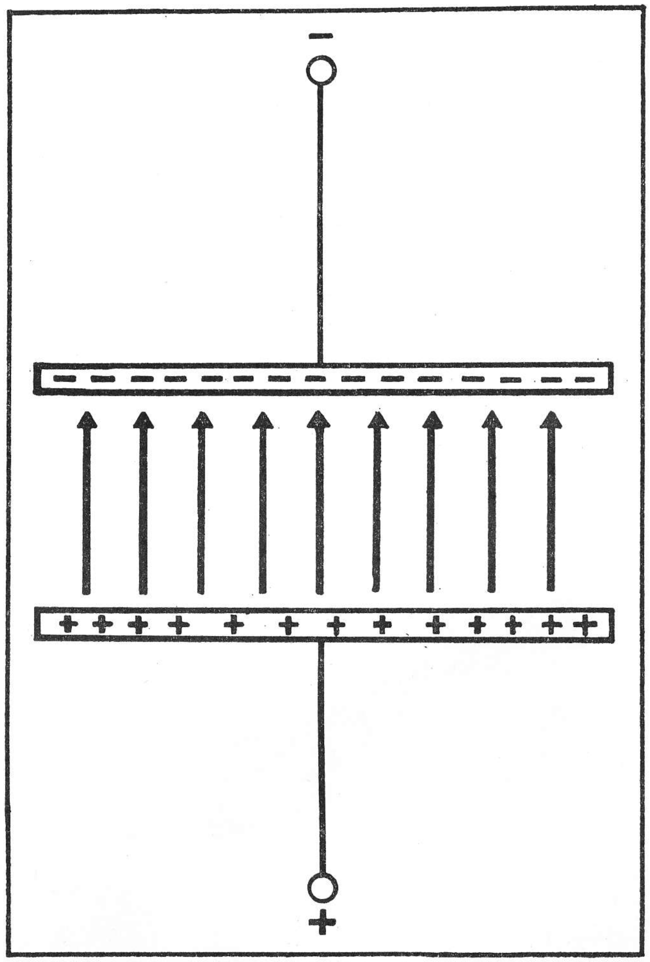

Figure Thirty-six

A voltage is then needed which will vary the electrostatic charge between the two plates in direct proportion to the audio modulation which is to be reproduced. This voltage is obtained through a resistance-capacitance-inductance network from the same audio output tube plate to the same speaker terminal (the diaphragm) and is the means by which the voltage variations in the output tube plate circuit are coupled to the speaker. This R-C-L network actually forms a high pass filter which passes information only above a frequency of approximately 7,000 cps.

The purpose of the filter is to eliminate frequencies below 7,000 cps since these frequencies, due to the design of this speaker, would not be faithfully reproduced. The circuits which provide both the polarizing voltage and modulating voltage are illustrated in Figure 37.

As the varying audio modulation voltage is superimposed across the speaker which already has applied to it the polarizing voltage, the strength of the electrostatic force between the two plates also changes in direct proportion to the variations of the audio modulation voltage. This in turn causes a greater or lesser attracting force between the two speaker elements and consequently, the movable plate or diaphragm is set in motion reproducing in audable vibrations the electrical modulation energy being applied to the speaker. Figure 38. Those frequencies below 7,000 cps are adequately handled by the bass or whoofer speaker which we have previously discussed. The electrostatic speaker is mounted in an aperture in the cabinet front in order to fully utilize its spatial distribution characteristics.

Thus the design of the Phonorama with the bass speaker, acoustical lens, cabinet enclosure and high frequency electrostatic speaker, provides a response range of approximately from 20 to 20,000 cps which is relatively smooth and which well covers the range of usable sound. High fidelity sound reproduction is therefore available in a price range comparable to that of the quality console models or combination radio-phonograph models that have been on the market for some years.

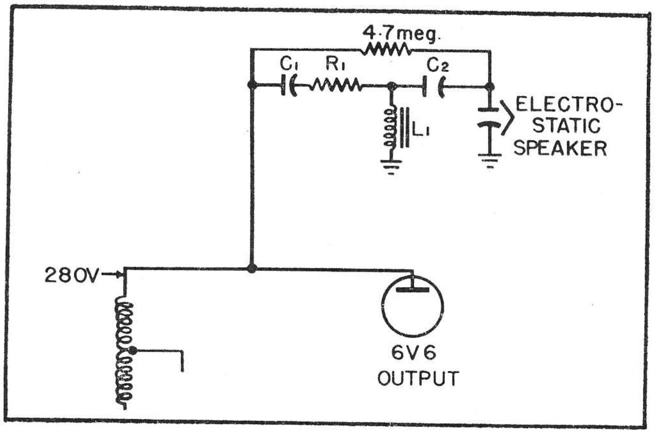

Figure Thirty-seven

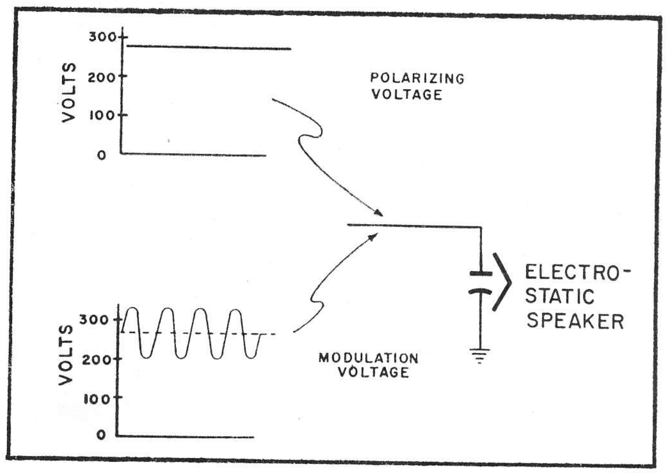

Figure Thirty-eight

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