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Saya NORAZILAH BTE MOHD YUSOF

(HURUF BESAR)

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Alamat Tetap: 156, BUKIT APING BARAT, 81900 KOTA TINGGI, JOHOR

EN. SHAIKH NASIR B. SHAIKH ABD. RAHMAN

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STUDY ON LOUDSPEAKER SYSTEM

NORAZILAH BTE MOHD YUSOF

A PROJECT REPORT SUBMITTED IN PART FULFILLMENT OF THE REQUIREMENT FOR THE DEGREE OF BACHELOR OF ELECTRICAL ENGINEERING

FACULTY OF ELECTRICAL ENGINEERING

UNIVERSITI TEKNOLOGI MALAYSIA

1997

DECLARATION

I declare that this thesis entitled ' Study on Loudspeaker System' is the result of my own research except for works that been cited in the reference.

Signature :

Name of Candidature : Norazilah bte Mohd Yusof

Date : 10 April 1997

SPECIAL DEDICATION TO MY PARENTS,

BROTHERS AND FRIENDS

WITH LOVE

ACKNOWLEDGEMENT

In the name of Allah, Most gracious, Most merciful. First and foremost, praise be to god who gave us free will and strength to help us learn new things and overcome ignorance in our lives and also to help me complete my project report.

It is a great pleasure to express my heartiest gratitude to my supervisor, Encik Shaikh Nasir bin Shaikh Abdul Rahman for their guidance's, critism, encouragement and providing me a great deal of information concerning the project.

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Last, but certainly not least, I am indebted to my parents for their understanding, help and giving me moral support throughout my project development period.

ABSTRACT

The loudspeaker is one of the device that exist in electronic equipment that normally use in radio receiver set, television, etc that changes electrical impulses into audible sounds.

Although all the speaker do the same things which reproduce sound from electrical signals, indeed, not all the speakers are designed alike. There are different sizes, types and functions of speaker and made with a variety of materials. The designs also depend on the cost and the requirement nowadays.

This project purposely to study and briefly explaining the operating of loudspeaker system. Through this project, it will concern about how speaker work according to its types and the elements required in building loudspeaker system such as the drivers and enclosures. In addition, it will shortly mention the using of computer program to design loudspeaker.

ABSTRAK

Pembesar suara merupakan salah satu peranti yang terdapat di dalam alat-alat elektronik iaitu biasanya digunakan pada set penerima radio, televisyen dan sebagainya yang mana ianya berfungsi untuk menukarkan dedenyut elektrik kepada gelombang bunyi.

Walaupun pada amnya setiap pembesar suara mempunyai fungsi yang sama iaitu menghasilkan bunyi daripada isyarat elektrik, sebenarnya, setiap satunya adalah berbeza. Perbezaan ini dapat dilihat di segi saiz, jenis dan fungsi juga dibuat daripada pelbagai jenis material. Rekabentuknya juga bergantung kepada kos dan permintaan masa kini.

Projek ini bertujuan untuk menjalankan kajian dan menerangkan secara ringkas tentang pengoperasian sesebuah sistem pembesar suara. Melalui kajian ini akan diterangkan bagaimana sesebuah pembesar suara bekerja mengikut jenisnya juga elemen- elemen yang perlu ada untuk merekabentuk pembesar suara ini seperti pemicu dan penutup pembesar suara. Sebagai tambahan, projek ini juga akan merangkan secara ringkas penggunaan pengaturcaraan komputer untuk merekabentuk sesebuah pembesar suara.

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DESCRIPTION

PREFACE

The objectives of this thesis is to study and design loudspeaker system. The thesis begins with an introduction to the principles sound which obvious the characteristic of sound. It continues with the transformation of sound and its performance.

Then the discussion on loudspeaker system begins, and we will familiarise ourselves with the principal types of loudspeakers. Also, speaker specifications is introduced as a starting point for loudspeaker system design as discussed in chapter 6.

Moreover we will look into the unit loading in the speaker. These consists of the several types of driving systems which is used in different types of loudspeaker. The discussion is followed with the application of multiple drivers in loudspeaker system, which will be described in more details in chapter 4.

Speaker performance is useless without an enclosure to make it a quality sound system. There are many types of enclosures being used. However, only three types will be discussed in chapter 5.

Chapter 6 then deals with the designing of loudspeaker using QBASIC programming. It continues with some brief explanations about choosing a loudspeaker.

Lastly, we will look at some recommendation and also the conclusion of this project on chapter 7.

CHAPTER 1

THE SOUND CIRCUIT

Sound would be of little interest if we could not hear. It is through the production and perception of sounds that it is possible to communicate and monitor events in our surroundings.

This chapter deals with sound in its various forms beginning with a description of what it is and its characteristic, followed by it's transformation and finally we will look at the sound performance in room.

1.1 PHYSICAL NATURE OF SOUND

Sound is a physical disturbance in the medium through which it is propagated. Although the most common medium is air, sound can travel in any solid, liquid, or gas. In air, sound consists of localized variations in pressure above and below normal atmospheric pressure (compressions and rarefactions)

1.2 SIMPLE SOUND SOURCE

The simplest source of sound as shown in figure 1.1, expands and contracts equally in all directions as if a perfectly round ballon were rapidly inflated and deflated. The expansion and contraction of the source result in three-dimensional sound ripples which spread out unimpeded in all directions as everexpanding spheres of compression and rarefaction at the velocity of sound which is :

$$c = f\lambda$$

Where c = velocity of sound, ft/s (m/s)

f = frequency, Hz

λ = wavelength, ft/Hz (m/Hz)

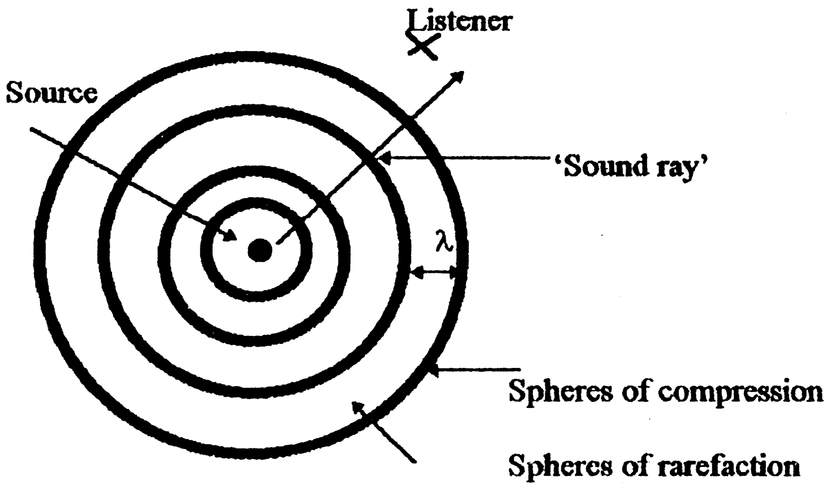


Figure1.1: Simple sound source

The speed of sound in air is approximately 1130 ft/s at normal room temperature. For quick estimates, this may be rounded off to 1000 ft/s. For design surveys, it may be more convenient to use a simplification that sound travels about 1 ft/0.001s.

Sound wave of all frequencies, whether from a low-frequency *woofer* or a high-frequency *tweeter*, travel at the same speed. An international standard (International Organization for Standardization, Recommendation

R226, 1961) sets middle A (so-called tuning A) at 440 Hz. This tone has a wavelength of 2.59 ft.

1.2.1 Sound spectrum

The audible spectrum of sound ranges from 20 Hz to 20 kHz. The fundamental tone or pitch of musical instruments ranges from piano at the lowest end of human hearing to about 4 kHz. However, every instrument also develops harmonics that are frequencies many time higher than the fundamental pitch. These harmonics are important in our ability to identify types of musical instruments.

1.3 SOUND PROPAGATION

Sound wave travel away from a simple source in spheres of ever-increasing diameter. The sound pressure is reduced in amplitude by a factor of 4 each time that the radius is doubled, since the sound energy is distributed over the sphere's surface, which has become 4 times larger. In decibel terms, the new sound level is decreased by 20 log (ratio of distance). Thus, when the radius or the distance that a sound wave travels has doubled, the sound level is reduced by 20 log (2), or 6 dB. Conversely, each time that a listener's distance from the source is halved, the sound level increases by 6 dB. This is not true once a listener is close to the source.

Most speaker cabinets have dimensions of less than 1 m; this is typically the minimum distance at which the rule of 6 dB per distance doubling can be applied. At a distance of less than 1 m the sound level increases asymptotically to a maximum value at the vibrating surface.

1.3.1 Sound power

Sound power level (Lw) applies only to the source, whereas sound pressure level

(Lp) is also dependent, on the environment and the distance from the source but this two sound expressed in decibels, a logarithmic ratio.

Sound power level cannot be measured directly but is calculated from measurement of sound pressure level made with a sound-level meter. Sound power is calculated from ;

$$L_w = L_p - 10 \log (Q/4\pi r^2) - 10.2$$

Where r = radius, ft

Q = directivity factor

And,

$$L_w = L_p - 10 \log (Q/4\pi r^2)$$

Where r = radius, m

1.4 SOUND DIRECTIVITY

The directivity characteristic can be specified by means of a directivity factor. If an omnidirectional source is placed against a large reflection surface such as a floor, the sound will radiate only into a hemisphere, or half of the previous solid angle. The directivity factor Q of this source increases from 1 to 2. If the solid angle is again halved by another large plane, such as by placing the source on a floor next to a wall, the directivity factor now increses to 4. When a source is placed in the corner of a rectangular room, the sound can radiate only into one-eight of a sphere; so the directivity factor is now 8.

1.5 THE TRANSFORMATION OF SOUND

Although the loudspeaker is the prime source of sound in any reproducing system, the sound we actually hear when we listen to a radio or phonograph is not entirely the result of the loudspeaker performance. We hear the result of many interacting factors, which constitute the subject matter of applied and practical acoustics. Not until the electrical signal (the counterpart of the original sound) acts upon the loudspeaker mechanism is the signal transformed into sound waves. This transformation is not the end of the road to fidelity. The loudspeaker is only the beginning of one chain among others in the high fidelity circuit, but this chain consisting of the acoustic circuit extending from the vibrating system of the speaker diaphragm to the nerve endings in the listener's brain, is obviously of more than passing interest. This is in fact the circuit most intimately and critically involved in the subjective or personal factor in the listener's high fidelity equation.

1.6 THE SOUND PERFORMANCE IN ROOM

The loudspeaker baffle or enclosure is the one determining factor for the final performance of the loudspeaker itself. The size of the baffle, its construction, and its actual shape will determine how well the loudspeaker will reproduce the low frequencies, how well high frequencies will be dispersed in the room, and even where in the room it should be placed for optimum performance.

It will be recognized that a room is an enclosure. There is no basic difference between the room in which we listen and the enclosure in which the loudspeaker is mounted. There are, of course, differences in degree.

However, the sound has yet to reach our ear, and our consciousness. The diaphragm is vibrating, as is the air, with the baffle molding these vibrations in intensity and in direction. Some of the sound so transformed breaks away from the speaker-baffle combination, traveling straight to the listener's ear. More of the sound, travels as quickly to other parts of the room. In fact, only a small part of the total sound produced actually moves directly to the listener's ear, for his ear occupies only a tiny portion of the total physical volume of the room. The overall sound fills the room, although in different degrees. It tranverses the room, bouncing from wall to wall, and later, some of it reaches the listener's ears. When it finally gets to the ear some of the sounds that were generated at the same time have already arrived. Thus the ear receive an "echo" of the original sound.

The room condition then is the next link in the acoustic chain. The "liveness" of the room determines not only the apparent spaciousness of the sound but also what sounds will be re-inforced and what sounds will be absorbed. The amount and texture of draperies in the room, the wall textures, the amount of large untreated plaster surfaces, the thickness of pile on the rugs, and other structural and decorative factors affect the sound before the ear hears it. The listening room modifies the sound in intensity, in tonal characteritics, and in directivity.

CHAPTER 2

LOUDSPEAKER SYSTEM

The term loudspeaker is commonly used to describe both the loudspeaker unit and the loudspeaker system. A loudspeaker system consists of a cabinet or enclosure into which the loudspeaker units operate. The idea is to isolate the rear from the front of the speaker cone to avoid cancellation of the sound waves at low frequencies. A speaker system can be derived by mounting the unit over a hole cut in a large panel, called the baffle board which acts as the isolating medium between the two sides of the cone.

Moreover, a loudspeaker system may contain either a single unit, or two or more of them, depending on the design, cost and requirements. It is the job of the loudspeaker system to convert electrical impulses into sound waves, and to do this as evenly as possible over the entire frequency spectrum, from deep bass to high treble.

This chapter discussed a principal types of loudspeaker. It also will give a briefly explanation of some fundamental needed about loudspeaker system. In addition, it will introduce some of the speaker specification which is only supplied by the manufacturer whereby, the specification is needed in designing loudspeaker.

2.1 LOUDSPEAKER

A loudspeaker is an electromagnetic transducer for converting electrical signals into sound. There are two principal types of loudspeakers :

1. The vibrating surface, called the diaphragm radiates sound directly into the air (direct radiator type).
2. Those in which a horn is interposed between the diaphragm and the air (indirect radiator).

This two principal types of loudspeakers will be discuss further. In addition, for more detail, there are several types of loudspeakers that using all these principal which will be mention later in chapter 3.

The ideal loudspeaker would be pulsating sphere that radiates sound in phase equally in all direction and all audio frequencies.

The ***direct-radiator*** type is used in most home radio receiving sets, in phonographs, and in small public-address systems. It is designed that the driver mechanism of voice coil and diaphragm is in direct contact with the air mass of the surrounding environment. In other words, the driver directly radiates its energy into the listening area. But, of course, the driver is quite small compared to the large amount of air that must be set in motion. In terms of impedance, the high mechanical impedance of the driver is directly coupled to the low acoustic impedance of the air, resulting in an inefficient transfer power. In order to increase the physical size of the moving system, and thus move more air, the diaphragm of a direct radiator is usually surrounded by a speaker cone as shown in figure 2.1.

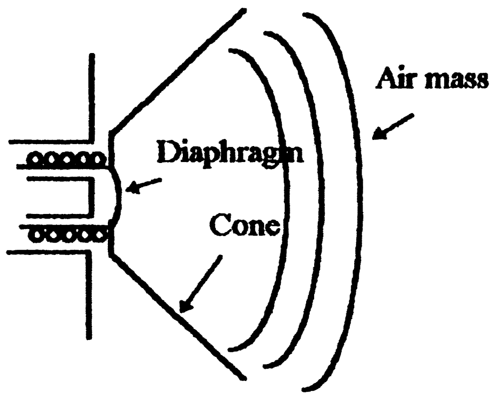


Figure 2.1 The direct radiator type

The horn type or ***indirect radiator*** is used in high-fidelity reproducing systems, in large sound systems in theaters and auditoriums, and in music and outdoor-announcing systems. The complete indirect radiator system as shown in figure 2.2 consist of the driver, compression chamber, and a horn with an expanding cross-sectional area. The driver and compression chamber are usually as a single unit known as a compression driver. The horn indirectly couples the small diaphragm to the large air mass.

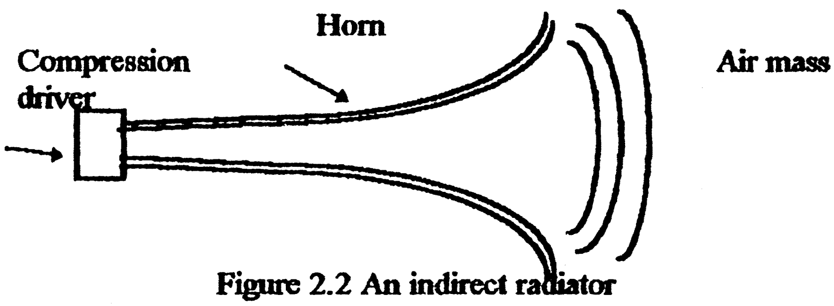


Figure 2.2 An indirect radiator

2.1.1 Doppler effect

A high frequency produced by a cones that is also moving backwards and forwards with large bass for excursions exhibits Doppler effect. The pitch rises as the cone moves forward and falls as it recedes, thus frequency modulating it. The modulation is proportional to the amplitude of the low frequncy excursions and, as small cones move further than large ones to produce the same power, they generate more f.m distortion. The distortation increase is proportional to the square of the decrease in cone diameter and is at a maximum on axis, decreasing as the off-axis angle increase.

2.1.2 Phasing

To reproduce the original sound waves accurately, the phasing of individual units must be such as to preserve the original phase coherence when their outputs are acoustically combined.

When mounted on a flat surface the bass-driver cone, being deeper, is further from the listener than that of the treble unit. The delayed sound results in a phase difference, and when the spacing is half a wavelength, cancellation of sound waves will occur. At normal spacings, this can occur from 3-4 kHz, which is within the overlap region of the two drivers in many speakers. Some models have the tweeter connected in opposite phase to partly compensate; others have the units in line. These solutions are only effective on axis, as the delay reduces with an increase of off-axis angle.

2.1.3 Controlled flexure

Cone buckling, normally undesirable, is used and controlled in some cases to produce full-range speakers. The cone has curved sides, and the central area responds to high frequencies independent of the rest, owing to flexure at points governed by the cone contour, inertia of the outer area and the applied frequency. As the frequency decreases, larger areas are brought into play until the whole cone moves at the lowest frequencies. This effect enables a single unit to effectively reproduce a wide frequency range, 40 Hz-17Hz being typical with some extending up to 20 kHz.

2.1.4 Cone materials

The necessary characteristics are : rigidity, low mass and self-damping to minimise resonances. The most common is paper which is light and has excellent self-damping, but is not very rigid or consistent. The paper pulp stock consists of wood and rag with various additions. Among these are *kapok:* hollow, oily fibres from the silk-cotton tree chosen for their lightness: and waxes, fungicides and resins. The stock is beaten for precise periods in vats. Long periods produce short-fibre, thin, hard paper which results in sensitive cones, though prone to strong resonances. Short periods give long fibres that are more flexible and are suited for bass cones and controlled flexure full-range cones.

Polystyrene reinforced with aluminium foil is very light and rigid but has poor damping. Honeycombed aluminiuim is about 1000 times more rigid than paper and has been used in disc form giving a more piston-like action than cones. Polypropylene is light, has good self-damping properties and is more rigid than paper.

2.1.5 Cone suspension

At the edge the cone is terminated by cloth, foam, rubber or simply by corrugations in the cone materials. Its purpose is to support the cone, preventing sideways movement but without unduly restricting normal forward and backward motion. In addition it should absorb waves that travel out from the cone centre, which are otherwise reflected back and form standing waves. Cone material corrugations can be of *two sine rolls*, a *single sine rolls*, or a deeper *accordian pleat*. Those made of foam or rubber can be of a *half-roll out*, having the roll facing the front of the speaker, or a *half-roll in*, with it facing the rear. Units designed for sealed-box mounting rely on the enclosed air partly restrain cone movement and so are designated as having *acoustic suspension*. These should not be used in non-sealed enclosures.

2.1.6 Cone resonance

Output drops at 12 dB/octave below cone resonance so it should be made as low as possible to achieve good bass response. The free air resonant frequency is proportional to the square root of the reciprocal of the mass of the cone and the compliance of the suspension. Compliance should be as large as possible commensurate with a stable suspension. Cone mass should be large and can readily be increased. The resonant frequency can be obtained from :

$$f_r = \frac{1}{2\pi\sqrt{MC}}$$

Where ; M = cone mass in grams

C = Compliance in meter/Newton. The reciprocal of the suspension stiffness.

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

The air in a sealed enclosure adds to the cone stiffness so reducing the compliance and thereby raising the resonant frequency.

2.1.7 Damping

A peak appears at the resonant frequency which is Q times the normal level. The peak is thus self-dampening, but the dampening depends on the effectiveness of the e.m.f. generation (flux density times coil length) and inversely on coil resistance, cone mass and resonant frequency. The value of Q is therefore :

$$Q = \frac{2\pi f_r MR}{(Bl)^2}$$

Where, R = Coil resistance

M = Mass in grams

f_r = Resonant frequency

B = Flux density

l = Coil length

A Q of 1 gives optimum damping at resonant frequency, but a lift of 3 dB just above it. A Q of 0.7 gives a flat response and an earlier but gentler roll off and is the preferred value. As enclosure air-volume affect the resonant frequency, it also affects the damping. There is thus a critical volume for a given bass unit to achieve the preferred damping level.

2.1.8 Cone velocity/radiation resistance

As the frequency increases, the cone inertia progressively reduces the amplitude of it excursions, so that at the high frequencies it is small compared to the low, for the same electrical input. However, at low frequencies the cone is an inefficient radiator, much of the air being pushed aside instead of being compressed. As the frequency rises the air does not move aside fast enough so it offers resistance to the cone and is compressed.

The radiation resistance of a loudspeaker determines its power transmitting capabilities and it is not a constant factor. It changes with the frequency of the power being transmitted and with the dimension of the device that does the transmitting.

The increasing radiation resistance exactly offsets the falling cone velocity so maintaining constant acoustic output at rising frequencies. The unit is said to be working in its piston region over this frequency range.

2.2 SPEAKER SPECIFICATIONS

The speaker specifications are useful to speaker designer. By knowing to analyze the specifications, they will know how the specifications will affect the sound of assembled speaker system. Note that these specifications are not given for every speaker type, nor do all speaker manufactures list them. The following part describes some specifications and how to interpret them. Furthermore, the specification list would be used in designing loudspeaker which will be mention in chapter 6.

2.2.1 Impedance

Speaker impedance is usually either four or eight ohms, and this value is normally given by the speaker's manufactures for all speaker types.

2.2.2 Frequency response

The wider the frequency response the better is the performance of the speaker. However, one should balance the frequency range with the flatness of the response. Most specification sheet include a response graph, showing the ability to deliver the same level of loudness over a wide sonic spectrum. Other specification sheets may just give a frequency range and response tolerance like "100 Hz to 5 kHz 3dB".The normal frequency response graph is almost like a hilly mountain. However, the flatter the response would be the better.

2.2.3 Free-air resonance

Free-air resonance is the most common way to express cone resonance. The frequency is often given as a range, but for computational purpuse, the figure in the middle of the range will be choosen. It also used to compute the best size of enclosure for the particular speaker. Free-air resonance is typically given for woofers only

2.2.4 Q(ts)

The Q of a speaker denotes its resonance magnification, which takes into account the degree of damping of a speaker, and the tendency of the speaker reach its maximum sound output level when operating at the free-air resonance frequency. The Q and the frequency of resonance are increased when the speaker is placed in a sealed box. The free-air Q of a speaker can range from about 0.2 to 1.5, with 0.4 or 0.5 being common. The Q(ts) specification being used when designing speaker enclosures and the value usually reported only for woofers

2.2.5 V(as)

It is difficult to measure the compliance of the speaker other than to say that it is "high" or "low". The V(as) figure is a way to show speaker compliance by comparison. V(as) is the volume of air in cubic feet or liters which has the same compliance as the speaker suspension. The larger the speaker, the larger the V(as).

The V(as) specification also used in designing speaker enclosure and it is given only for woofers.

CHAPTER 3

THE LOUDSPEAKERS UNIT

It is shown in chapter 2 that a loudspeaker system commonly consists of a 'box' enclosing at least one loudspeaker unit. The majority of loudspeaker units are of the electromagnetic type, using a moving coil in a magnetic field.

However, this chapter will consider the driving system of loudspeaker unit and elaborate several types of loudspeaker which is an additional part of chapter 2 above. Furthermore, it will touch on multiple drivers uses in the speaker system.

3.1 THE DRIVING SYSTEM

3.1.1 Moving coil

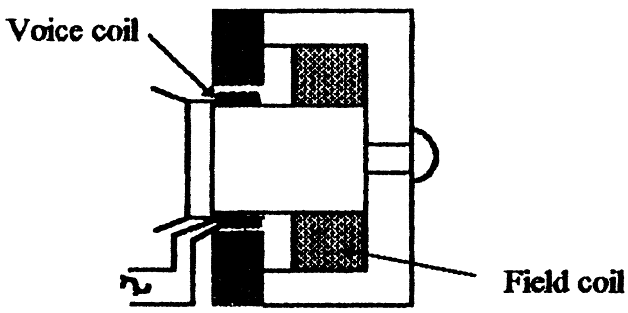


Figure 3.1 Basic construction of moving-coil

The operating principles and basic construction (figure 3.1) of the almost universal moving-coil loudspeaker have not changed since it was first produced by Rice and Kellog in 1927. Its success is due to the fact that it is fundamentally a linear transducer capable of giving reasonably efficient energy conversion with a compact and relatively cheap unit.

In moving-coil loudspeakers, the speech coil alone is subjected to axial magnetic forces in a radial magnet gap. The coil is attached to the cone or diaphragm which forms the sound radiating surface. Conflicting requirement of the cone are rigidity and low mass. Rigidity is necessary for the cone to behave as a piston , moving the adjacent air without buckling and so adding spurious air motion. Low mass is required to minimise inertia and ensure the cone responds accurately to rapid signal changes. A large area is desirable to propagate low frequency sound, but this also conflicts with the low mass requirement.

Because of the need for rigidity and low weight, the diaphragm has to be relatively small, 18 in being the maximum size for bass unit, average wide range unit

being about 8-12 in in diameter.

3.1.2 Moving Iron

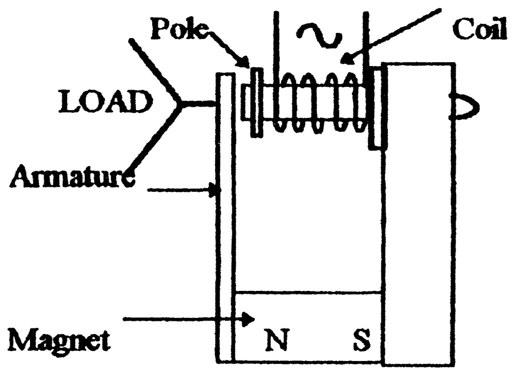


Figure 3.2 Basic construction of moving iron

This consisted of a permanent magnet operating directly upon a ferrous armature, clamped at one end, with a diaphragm, usually horn loaded, attached to the free end of the reed (refer figure 3.2). Most of the earliest loudspeakers were based on this system. Reproduction was marred by reed resonance and absence of bass, plus distortion from non-linearity in the single-sided operation of the reed in the magnet system.

3.1.3 Balanced Armature

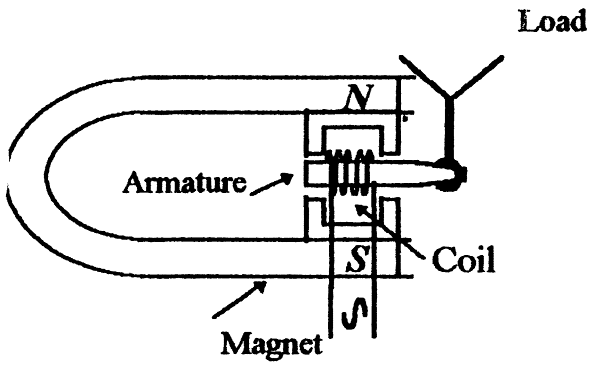


Figure 3.3 Basic construction of balanced armature

This design was due to the push-pull operation of the reed and a definite improvement on the previous type. There was still resonance from the reed, and it was not found possible to reduce the stiffness of the armature sufficiently to give bass response below 120 cycles.

3.1.4 Crystal

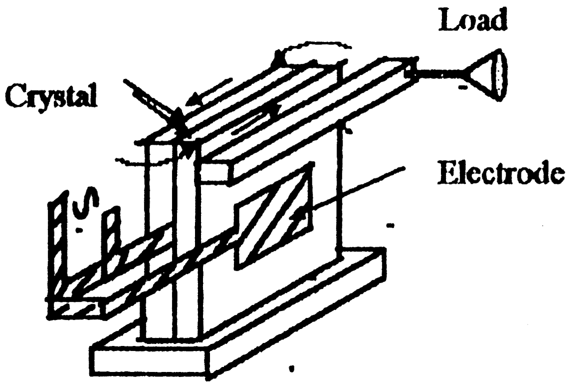


Figure 3.4 Basic construction of crystal

It is usually used in microphones and pickups which is consist the bending and twisting properties of certain crystal assemblies with applied EMFs.. The system has been applied to loudspeaker for high note reproduction.The crystal is not easily adapted for bass reproduction on account of the large movements which are necessary in a loudspeaker at low frequencies.

3.1.5 Ribbon

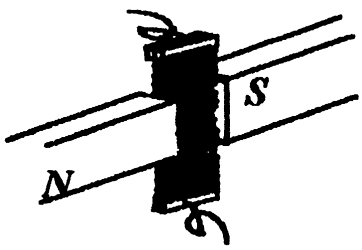


Figure 3.5 Basic construction of ribbon

The ribbon loudspeaker operates on the same principle as the moving coil except that the diaphragm and voice coil are one and the same thing. The original ribbon design includes all the essentials of present-day products, down to the corrugations in the aluminium strip. Although capable of giving excellent results at high frequencies, it has never really succeeded. This is because of the wide gap involves either low sensitivity or very expensive magnets. Moreover, the ribbon excursion is limited, so the system is not well suited to low frequencies and even high frequency reproduction is inadequate without horn loading.

In addition, the ribbon is fragile and is easily displaced mechanically. It is also extremely vulnerable to electrical overload caused by loud clicks or bangs.

3.1.6 Ionophone

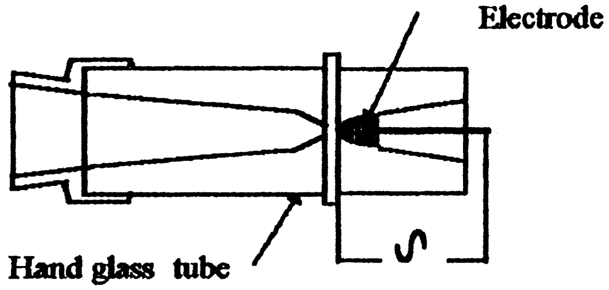


Figure 3.6 Basic construction of Ionophone

The ionophone does not employ moving parts. The sound source is a radio frequency corona discharge which takes places inside a small quartz or hard glass tube opening to the air through and exponential horn. The input signal produces amplitude modulation of the RF oscillator which causes corresponding variations in the discharge. Theoretically the ionophone is capable of perfect performance, limited only by the directivity and frequency response of the horn and the linearity of the electrical generator which maintains the discharge.

3.1.7 Single-sided Electrostatic

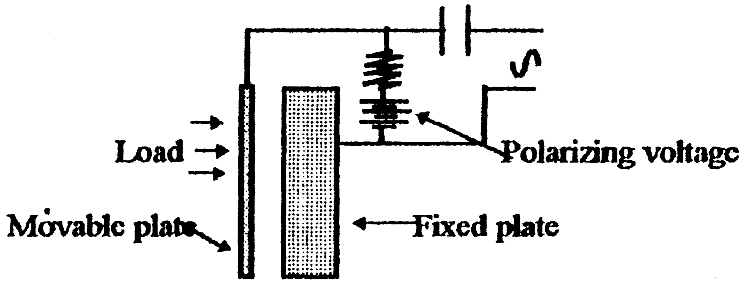


Figure 3.7 Basic construction of single-sided electrostatic

The electrostatic loudspeaker uses the principle of a condensor. The model were then expected to cover a wide frequency range and severe distortation was produced by the single plate system. Recent German design, used the system only for the HF ranges of 7000-20,000 c/s,thus avoiding the frequency doubling which sets in at lower frequencies, low sensitivity and increasing the polarizing voltage beyond about 250 produced little improvememnt. The device had therefore no application in efficient, high class speaker systems, but has been used successfully in commercial sets where small magnets are normally used in the main speaker.

3.1.8 Push-pull Electrostatic

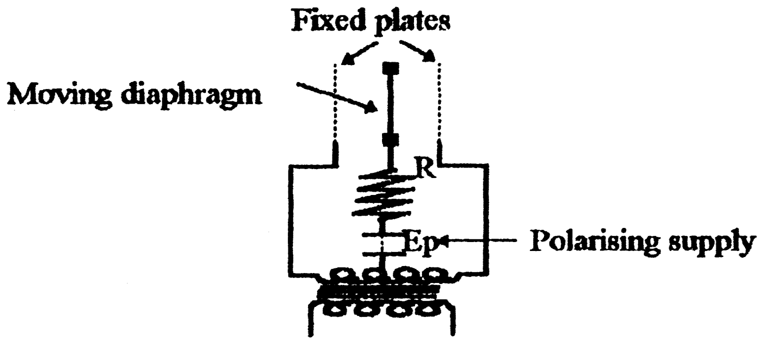


Figure 3.8 Basic construction of push-pull electrostatic

The development of push-pull working has had an even greater effect on electrostatic speaker than the balanced armature had on the old original moving iron. Therefore, further information will be discuss in electrostatic speaker's type.

3.2 SEVERAL TYPES OF LOUDSPEAKERS

As mention previously, there are many types of loudspeaker that uses the two basic principal of loudspeaker, direct and indirect radiator. In the following section, most of the speakers using the direct radiator, except for horn loudspeaker, is discussed

3.2.1 The dynamic loudspeaker

The dynamic loudspeaker has found acceptance in all kinds of reproducing systems. It is found in the smallest pocket radios and is a major component of the most elaborate theater systems.

One of the major factors contributing to the popularity of the permanent magnet dynamic loudspeaker is the fact that it has its own powerhouse. It requires no external source of power other than the signal power to make it operate. This powerhouse (the magnet) has a virtually eternal life. The permanent magnet of the dynamic loudspeaker, once it has been charged to full capacity by the manufacturer of the loudspeaker, retains its magnetomotive power almost unaltered forever, and unless it is subjected to sever mechanical shock, it cannot be drained or run down.

Basically the dynamic loudspeaker is made up of the following components which is the voice coil, voice coil former, centering spider, magnet, magnetic circuit, diaphragm, apex radiator and the basket or housing. These components are the basic elements of the simplest type. Figure 3.9 shows a full cutaway of a typical loudspeaker in which all these elements are clearly marked and arranged in a somewhat exploded view.

The dynamic loudspeaker is a device that function with other electronic equipment, but in itself it has little that is of an electrical

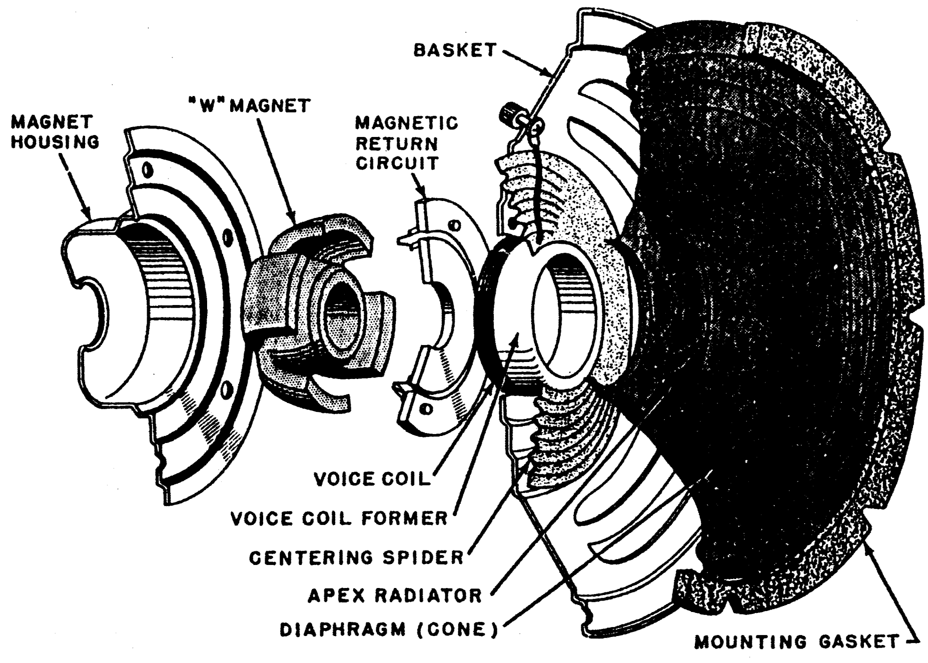


Figure 3.9 A cutaway view of dynamic loudspeaker

3.2.2 The electrodynamic loudspeaker

The electrodynamic speaker and the permanent-magnet dynamic speaker function in exactly the same manner.

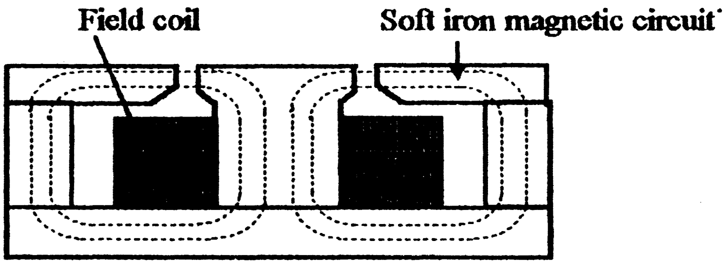


Figure 3.10 Permanent-magnet dynamic speaker

In place of the permanent magnet found in the permanent magnet dynamic speaker, there is instead a completely "soft" iron circuit. Around the center leg of this iron circuit is wound a large multi-layer coil, as shown in figure 3.10. When a direct current is made to flow through this "field" coil, it sets up a magnetic field about itself, which magnetizes the iron circuit within the coil. A field of magnetic flux is thus set up across the air gap of the return keeper. The strength of this field is a direct function of the strength of the current that flows through the coil, and the design of the iron circuit. With proper magnetic circuit design it is possible to get large values of gap flux, which means a more powerful speaker.

The manner of connecting an electrodynamic loudspeaker is shown in figure 3.11. It should be realized that the field coil has absolutely nothing to do with the voice coil. The field coil produces the steady magnetic field whereas the voice coil produces the varying signal field. Connections from the field coil are made directly to the system power supply, while the voice coil connects to the amplifier signal output terminals exactly as in the permanent magnet dynamic loudspeaker.

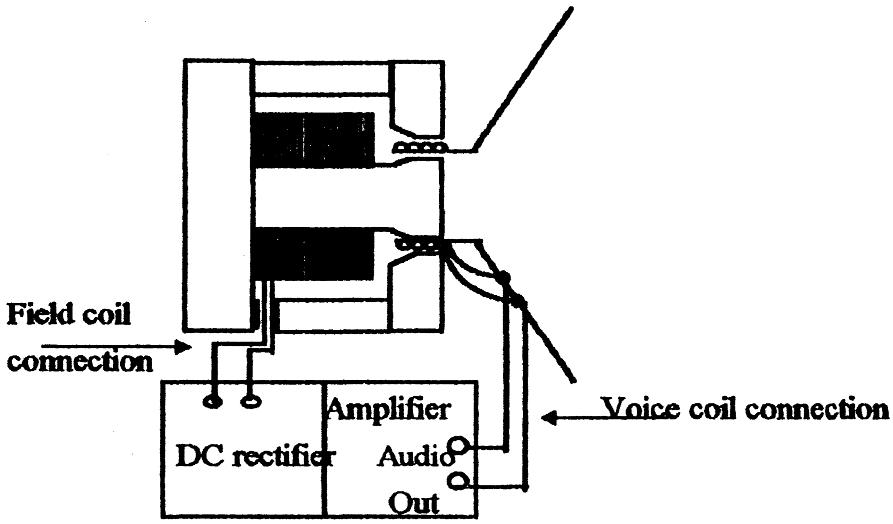


Figure 3.11 Connection of electrodynamic speaker to amplifier

3.2.3 The electrostatic loudspeaker

The electrostatic speaker uses a thin conductive diaphragm driven over its whole surface by an electrostatic force. As rigidity is not a requirement the mass can be extremely small, so the response to transients and high frequencies is almost immediate. With all parts of the surface being driven, there are no break-ups or spurious movements such as experienced with a cone.

When a voltage is applied across two conductors having a large surface area in close proximity they are mutually attracted. If one is flexible it will move toward the other.

The construction of an electrostatic speaker as shown in figure 3.12, includes a single and a push-pull type. In the single type a fixed electrode consisting of a perforated metal plate, for example, is set against a diaphragm in which metal is deposited on a light, rugged polymeric film at intervals of approximately 0.3 to 0.5 mm. This type has the disadvantage that the operation becomes unstable and distortion increases for a large amplitude. Therefore, it is used mainly for a high frequency range speaker.

In the push-pull type the diaphragm is arranged between two fixed electrodes. Even when this type is vibrated at a large amplitude, operation is stable and distortion is significantly low. Therefore, the push-pull type is also used as a midrange and low-range speaker.

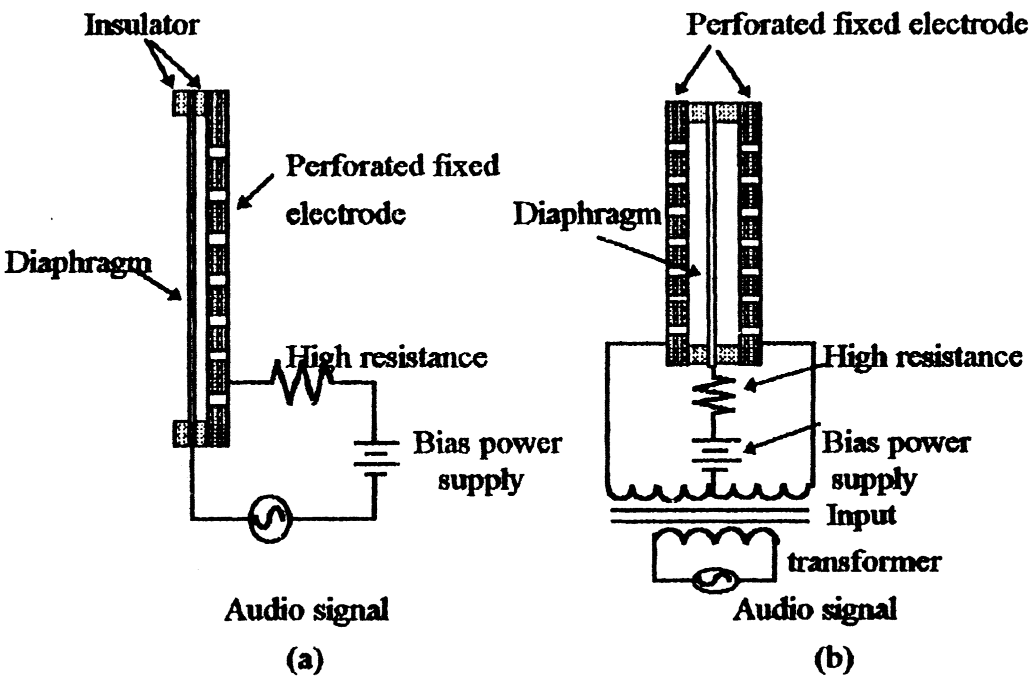


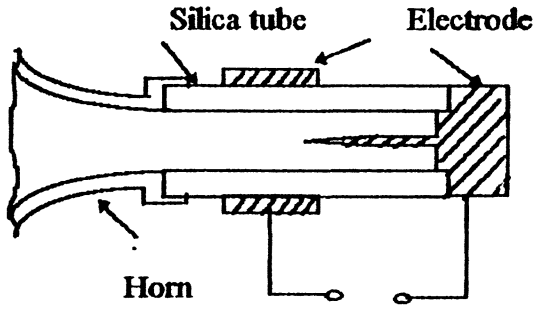
Figure 3.12 Electrostatic speaker configuration. (a) Single-ended drive. (b) Push-pull drive

3.2.4 The crystal/piezoelectric loudspeaker

The principle behind the crystal loudspeaker is one of contraction and expansion of a certain crystal material under the influence of an alternating electric field applied to the surfaces of the crystal. No polarizing voltages are necessary for the crystal loudspeaker as the flexure motion of the crystal follow directly in step with the polarity of the applied voltage. However, crystals are rather fragile devices, and it is not possible to drive them sufficiently hard to obtain useful output power for loudspeaker performance, especially on the low frequency end. Therefore, they are at present used only for earphones and for pillow speakers.

3.2.5 The ionic loudspeaker (Discharging-Type speaker)

As is apparent from a roll of thunder, it has been well known for many years that sound is produced by electrical discharges in air. However, this system cannot be put to practical use owing to the many noises caused by high-voltage arc discharges. In the 1950s, a system was developed wherein the arc is generated by an alternating current at a high frequency (eg: 20 MHz). Thus the arc is started and stopped continually at the high-frequency rate. The sound signal is generated by an amplitude modulation of the high-frequency carrier signal.



Figurre 3.13 Ionic speaker configuration

One electrode is a tube of a dielectric such as silica to prevent corona arc discharge (Fig. 3.13). This system was used in a commercial product called the ionophone as a speaker with a wide frequency range. The ionophone has no mechanical system such as diaphragm. However, it consists of a mechanism by which air particles are given an electric charge. After the air molecules have been charged, they form an "atmosphere" of ions, the ions in this case being , of course, charged air particles. A modulating voltage is then applied to this atmosphere, making it pulsate in accordance with the signal-modulated voltage. This pulsing gives rise to an actual sound wave (in the air) , which is then propagated in the usual manner. This mode of operation makes the ionophone truly a "diaphragmless" loudspeaker. The air molecules are actually agitated electrically without anything physical pushing them.

Because of the lack of mechanical action, transient characteristics are excellant. However, this type has the disadvantage of electrode wear and the need for a high-frequency oscillator. For this reasons, this speaker device presently is only used for very specialized applications.

3.2.6 The horn loudspeaker

Horn loudspeaker have been used for many years for reproducing sound efficiently within a specified sector. A horn speaker, as indicated by its name, is composed of horn and speaker component. A horn is defined as a "sound pipe whose cross section changes gradually in the longitudinal direction".

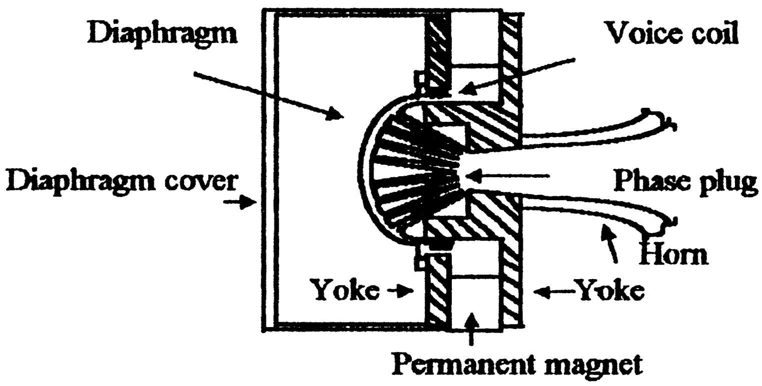


Figure 3.14 The construction of a horn speaker

The construction of a horn speaker are shown in figure 3.14 which consist of a driver and a horn. For diaphragm in drivers, for which lightness and a high resonance frequency of vibration of the normal mode are required and dome-formed shapes generally are used. The phase equalizer is installed at the junction between driver and horn. It enlarges the section of the route of the sound gradually by placing a rigid body closely in front of the diaphragm, for the cross section of the horn throat is smaller than the area of the diaphragm. The air room between the phase equalizer and the diaphragm is designed to be as small as possible for the reproduction of a high compass, and the distance between the phase equalizer and the route of sound is kept constant to prevent sound from being diminished by distance.

3.3 THE MULTIPLE DRIVERS

Most speakers have more than one driver. The reason for having multiple drivers is simple : the high frequencies unit, called the *tweeter* is small and light, thereby able to respond quickly to the rapidly changing high frequencies. The bass unit, termed the *woofer,* is large enough to propagate the longer wavelength. A mid-range unit of intermediate size, called a squawker, is sometimes used as well (medium frequency).

3.3.1 Tweeteer : Designed for max. efficiency at high frequency

For the high end of the acoustic spectrum there is the tweeter. Again, because the tweeter is restricted in range, it may be more specialized in design in favor of improved high frequency efficiency, with sacrifices being made in areas where good performance is not expected of the speaker. The tweeter must be attuned, or resonated, in the high frequency region since it is used to reproduce the high frequencies. High resonant frequeny is obtained with structures that are light in weight and stiffly supported, as indicated in figure 3.15

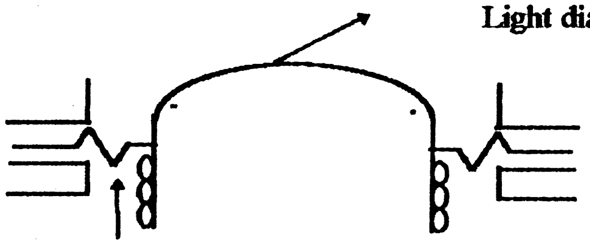


Figure 3.15 Tweeter sructures with have small, light diaphragm and tightly suspended to obtain high resonant frequency.

Naturally, to make the diaphragm light in weight, it must become small in size. Therefore, in contrast to the large 15-inch woofers, cone type tweeters are in the 3-inch class. By reducing the diaphragm both in size and in weight, it becomes necessary to proportionately reduce the voice coil to get optimum drive of the diaphragm by the coil. As a result of the entire moving structure of the cone type tweeter being so diminutive, it becomes almost a scaled down model of the larger woofer, but with a high resonance frequency.

3.3.2 Mid range : Determined by woofer and Tweeter limits

The midrange speaker begins to operate in that frequency region where the upper range of the woofer section becomes attenuated either by itself or through the intermediate devices of networks. Thus the expected low frequency response of the midrange speaker is usually fairly high in the acoustic spectrum. The upper range of the midrange speaker is governed by the lower range coverage of the tweeter, for the midrange unit generally need to go up in frequency only far enough to provide a smooth overlap with the tweeter range.

This midrange unit may be either a direct radiator cone speaker or a compression driver horn loaded system. The choice will be governed by the design of the system as a whole. If the woofer and tweeter sections of the system are direct radiator cone types, a midrange speaker of the cone type will match them in efficiency and provide a balanced system. If the woofer and the tweeter are high efficiency horn loaded sections, the midrange unit should preferably be a horn type. If it is a cone type, the midrange speaker may preferably be of the wide range type so that it may be operated low enough in frequency to match the woofer and at the same time give good high frequency reproduction up to the area where the tweeter takes over.

3.3.3 Woofer : Designed for max. efficiency at low frequency

A woofer is an integral part of a good high fidelity wide range speaker system, but it is not in itself a wide range unit. It is, on the contrary, a loudspeaker that is very restricted in range; it is limited to the reproduction of the lower part of the acoustic spectrum. Because it is restricted, it may be designed to provide much more efficient and cleans bass reproduction than would be possible from a single wide range speaker in which compromises must be made between highs and lows to obtain reasonably good all around reproduction

The woofer is large in size, rugged in construction amd attuned to the very low end of the audible range.It is also capable of producing large amounts of sound. All these factors are integrally related to one another in the final design of the speaker.

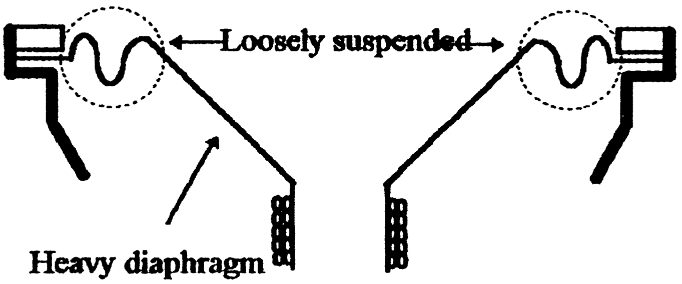


Figure 3.16 Woofer sructures with have large, heavy diaphragm and loosely suspended to obtain low resonant frequency.

Woofers have low resonance frequencies, therefore they are "tuned" to resonate and to be most efficient in the low frequencies. Low resonance frequency is obtain by heavy diaphragm loosely suspended as shown in figure 3.16 . The heavier the diaphragm, the lower the resonant frequency. The looser the edge compliance that holds it in place, the lower the resonance. Even though a diaphragm may have low resonance and be attuned to the low frequencies, it has to have a large vibrating surface for the low notes to be efficiently radiated and adequately projected into the listening area, such as the *bass drum* which is has a large vibrating membrane. The larger the diaphragm, the more efficient the reproduction of the low frequencies.

CHAPTER 4

NETWORK IN MULTI-SPEAKER SYSTEM

Since a good low-frequency speaker will not handle high frequencies well, and vice versa, most professional monitor systems consist of two or more speaker within an enclosure, with each speaker optimized for a segment of the complete audio bandwidth. However, do not expect that the high, middle and low frequencies to magically find the correct speaker. In fact, feeding high-power, low-frequency signals to an electrodynamic tweeter will quickly burn it out.

Therefore, it is the purpose of this chapter to introduce a basic principles of frequency-dividing or crossover network in multi-speaker system whereby it's objective is to ensure that the unit receives signal power at the bass end with a suitable roll-of into the middle part of the spectrum.

4.1 THE BASIC CROSSOVER NETWORKS

As mentioned, the function of the crossover network is to segregate specific bands of energy into specific channel. In other words, its divides the sound frequency spectrum into distinct ranges, and ensures that only the proper frequencies are routed to the appropriate speaker. In addition, this segregation of various bands of acoustic energy insures optimum utilization of audio power, resulting in better overall performance of the system.

The network may be either a passive or active design and we will discuss it latter. In either case, the network consists of two or more filters, with each output connected to a separate loudspeaker. Each filter is designed to pass frequencies within the range of its speaker and to attenuate all other frequencies. For example, a three-way monitor system such as the one shown in figure 4.1 might consist of the following speakers and filters:

Speaker's popular name	Frequency range	Types of Filter	Cutoff frequency
Woofer	Low < 1kHz	Lowpass	1 kHz
Squawker	Mid 1-4 kHz	Bandpass	1 kHz & 4kHz
Tweeter	High > 4 kHz	Highpass	4 kHz

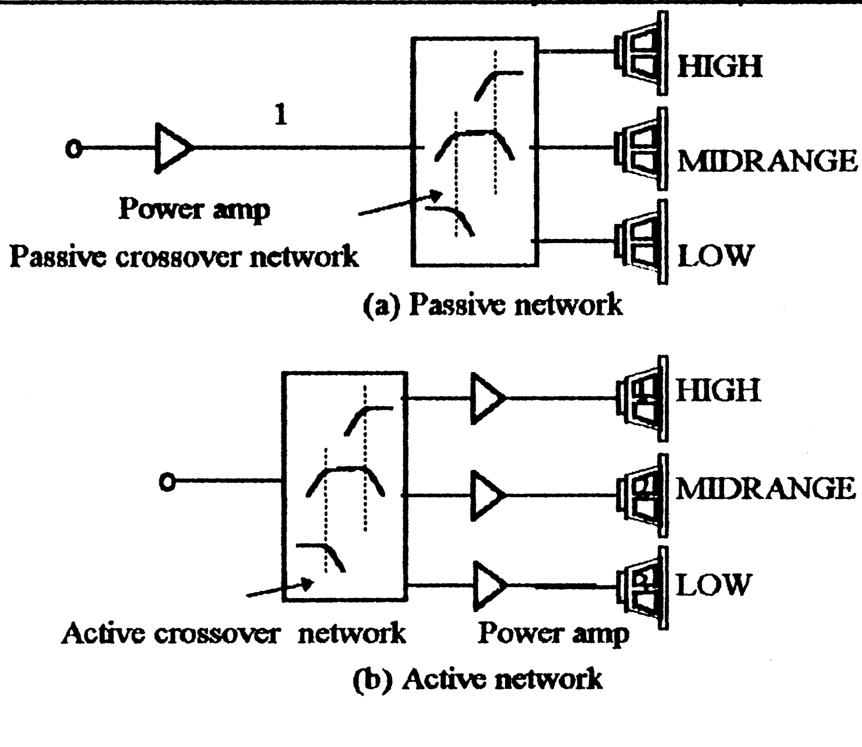


Figure 4.1 A three way monitor system, showing passive and active crossover network for low, midrange and high frequency speakers

4.1.1 Passive network

A passive crossover network consists of a series of response shaping filters inserted at some point ahead of the speakers. Each filters comprises one or more inductors and capacitor s whose value are determined by the desired crossover frequency and the impedance of the source (amplifier) and load (speaker), which are assumed to be equal. Such a network is often an integral part of the speaker system itself, which is driven by a single power amplifier as shown in Figure 4.1a

Furthermore, there is a few drawbacks which must be considered in using the single-amplifier system. For example :

- The network must be built to withstand the full power of the amplifier, much of which is wasted within the network itself
- The amplifier itself must be capable of delivering sufficient power to drive the complete speaker system.
- Network design usually assumes a purely resistive load, while a speaker's impedance contains an inductive component as well.
- If a speaker is replaced by another with a different impedance, the network element values will have to be changed.

4.1.2 Active networks

On the contrary, the crossover network inserted ahead of the amplification stage may be designed to operate at line level rather than at speaker level. Although such a network might also be a passive system, it is more often designed in conjunction with active components. In either position within the monitor system is as shown in figure 4.1b ·

Some of the advantage of an active crossover network :

- Network components need not be specified with large power-handling capabilities
- Amplifier power may be reduced to that required by a single speaker
- There are no passive components within the speaker line
- The sensitivity of each amplifier may be readily adjusted to suit the efficiency of the speaker it is driving.
- Speaker with different impedance may be interchanged without affecting network design
- The amplifier acts as a buffer between the crossover network and the frequency-dependant impedance of the speaker.

4.2 THE CROSSOVER COMPONENT

A crossover separates the sonic frequency spectrum into the ranges needed by each driver in a multi-way loudspeaker. More specifically, it is a combination of inductors and capacitors that form an electronic filter network. Therefore, the network are composed of elements which allow to pass, or prevent from passing certain bands of frequencies.

A capacitor will pass high frequencies and block low frequency. So, it is to be found in high-pass filter sections (figure 4.2a). A inductor or choke will block high frequencies and pass low frequencies. It is therefore found in low pass filter sections (figure 4.2b). If both a low-pass element and a high-pass element are in series in a single branch of a circuit, a band of frequencies is passed. The band has its low frequencies restricted by the high-pass element and its high frequencies restricted by its low pass element. Those frequencies in between these two restricted ends become the "pass band" (figure 4.2c).

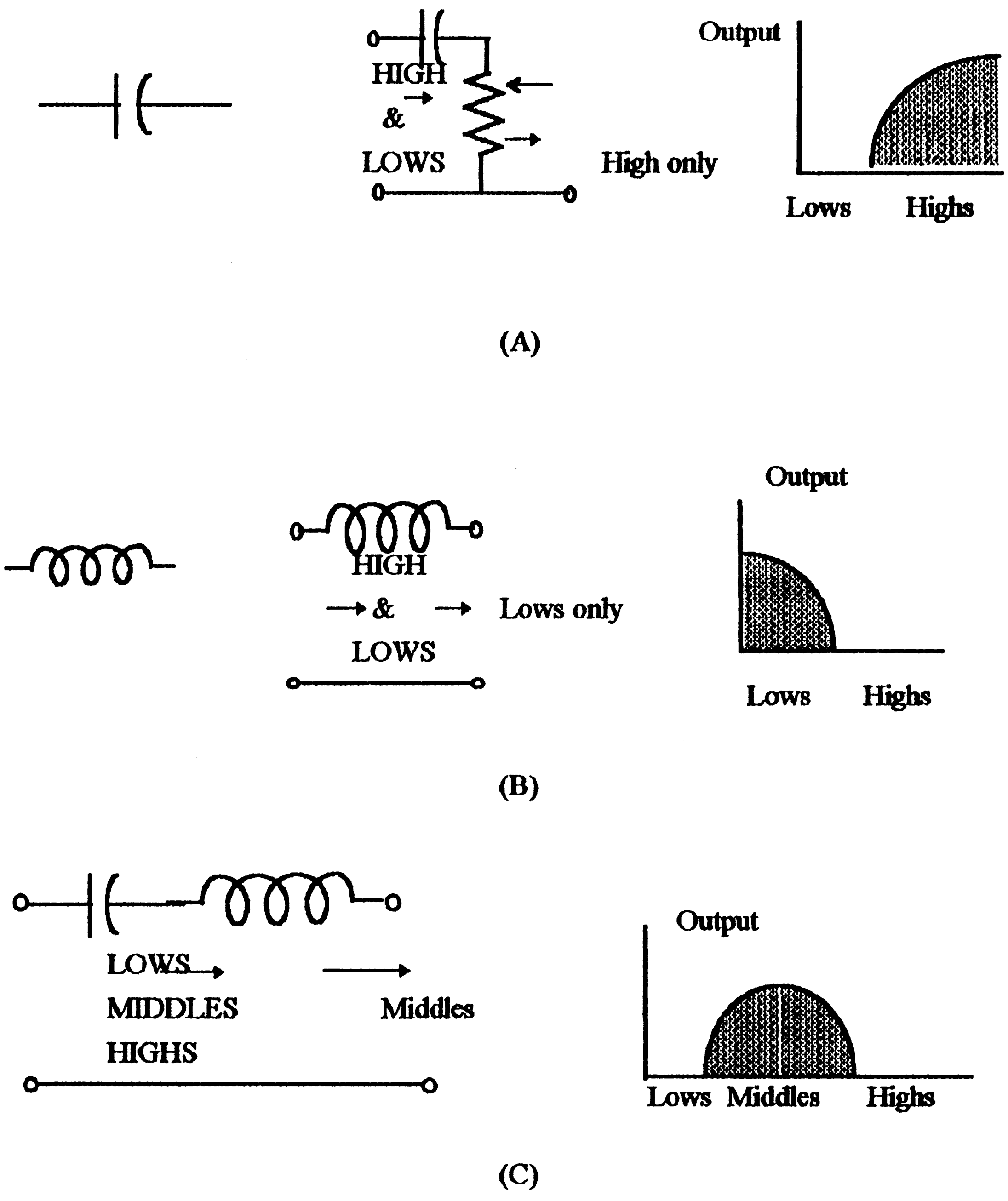


Figure 4.2. The Crossover network component with its basic concept function.

CHAPTER 5

LOUDSPEAKER ENCLOSURE

Loudspeaker enclosure are the subject of more controversy than any other item connected with modern high-fidelity music reproduction. Because the behaviour of enclosures has not been clearly understood, and because no single authoritative reference has existed on the subject, opinions and pseudo theories as to the effects of enclosures on loudspeaker response have been many and conflicting. The problem is complicated further because the design of an enclosure should be undertaken only with full knowledge of the characteristics of the loudspeaker, the amplifier available and these data are not ordinarily supplied by the manufacturer.

This chapter simplifies the fundamental of loudspeaker enclosures. We will learn about the significance of enclosure. This is followed with the various type of enclosures and how each enclosure acts as a crossover element in the speaker system. In addition, we will discuss briefly the construction of enclosure according to its mecahnical aspect.

5.1 THE ENCLOSURE ESSENTIALS

Why we should use an enclosure?. Indeed, enclosures are used to house the speaker. Moreover, it protects the speakers against damage and most important, it improves the frequency response so that the sound is more natural.

As we had known , sound waves are created by vibrations of the speaker cone While in open air, sound waves are dispersed in all direction from the speaker. When the cone moves outward, it creates a positive pressure on the air in front of it and simultaneously creates a negative pressure (partial vacuum) on the air behind it. At low frequencies, where the cone diameter is much shorter than a wavelength, out of phase waves from the rear of the cone mix with and cancel the front wave. Therefore, this will reduce the speaker output.

When speaker is placed in a suitable enclosure, the second waves emitted from the rear of the speaker can't travel to the front of the cone. This will prevent low frequency cancellation. As a result, it will improve the speaker efficiency and sound output in the bass frequencies.

5.2 ENCLOSURE ACTS AS AN ACOUSTIC CIRCUIT

The enclosure is an essential part of any loudspeaker system. The prime purpose of any enclosure is to provide the proper acoustic circuit for the loudspeaker to work with, so that maximum efficiency and best performance may be obtained from the combination. In order to provide this acoustic circuit, the sound coming from the loudspeaker is routed into certain paths and prevented from going into other paths by blank walls put in its way. Thus, the term "baffle" as used in a technical sense connotes a means of routing the sound energy. The word enclosure and baffle are used interchangeably since most of today's baffle are built into more or less complex box-like structures. However, in certain instances, we shall specifically use the word *baffle* because in no sense will the word *enclosure* describe the function of the unit. So, let us find out what the baffle is all about.

5.2.1 Baffle

A baffle is a structure for shielding the front-side radiation of a loudspeaker diaphragm from the rear-side radiation. The purpose of any baffle is to provide the speaker with the proper load, into which it operates to produce acoustic power output.

'Baffling' can control the sound intensity and its distribution. Without increasing the input power, a mere whisper (especially a low frequency whisper) may be transformed into a roar if we modify the surrounding elements with which the loudspeaker has to react. The size of the baffle, its shape, and its construction will (along with other factors) determines how the air vibration modified. A good baffle is a battle half won in the struggle to perfect a high fidelity system.

The advantage of a baffle is that the many resonances and colorations produced by even a well-designed enclosure are eliminated. A compromise is to fit shallow sides to form an open-backed box so extending the front-to-back path with the addition of minimum coloration.

5.3 TYPES OF ENCLOSURES

There are several ways of designing enclosures to give an approach to the desired low-frequency response, but the big problem is the prevention of the 'boxy' sound colouration due to the many mechanical and acoustical resonances which can do occur in any cabinet design, unless special precautions are taken.

A properly designed enclosure should perform three functions :

1. To prevent interference due to out-of-phase sound radiation from the rear of the cone

2. The enclosure should at least partially damp the low-frequency resonance of the cone and its restoring surround and centring bellows. Thus it should incorporate some form of acoustic resistance.
3. The enclosure should raise the efficiency of the unit by increasing the radiation resistance of the effective air load applied to the cone at middle and low frequency.

Obviously, a number of speaker enclosure designs have been devised in the past of several decades. However, only several of the designs were popular and are still being used today.

In the following part, we will learn a few types of enclosures and its function

5.3.1 The infinite baffle or closed box enclosure

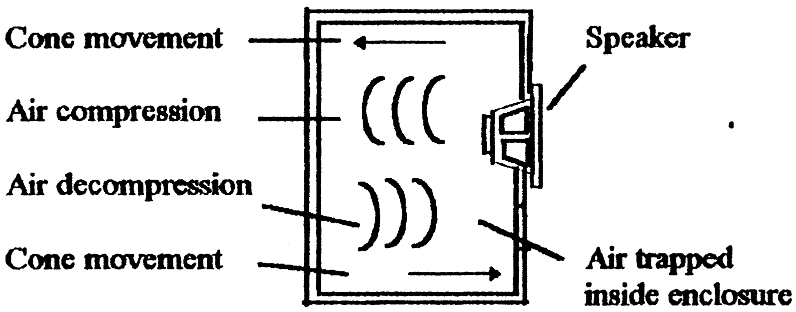


Figure 5.1 :A simple view of totally enclosed 'infinite baffle' cabinet as used for modern compact high-quality loudspeakers.

This is probably the most popular type of enclosure in commercial use today. The rear wave is 'smothered' by mounting the speaker at the front of a sealed box. The movement of the speaker acts as a piston to compress and decompress the air, as illustrated in figure 5.1. The enclosed air acts as a spring, reducing the suspension compliance and raising the resonance frequency of the speaker which is beyond its natural resonance point in free air. Mechanical suspension is made as compliant as possible to partly compensate this effect. Due to the characteristic just described, the closed box cabinet is referred to as an *acoustic suspension system*.

Compliance increases with box volume for a given driver of specified cone diameter and mass, to achieve a desired resonant frequency. The response falls at a rate of 12dB/octave below resonance.

Sensitivity

The sensitivity of the infinite baffle is low because the mass of the bass speaker cone is generally made high to bring down the resonant frequency and so extend the bass response. Efficiency could be increased by making the magnet stronger, but this reduces the Q and results in overdamping, which in turn prevents the use of resonance to boost the bass. So bass roll-off begins at higher point and the bass response is reduced. As with most of the other factors there is thus an optimum magnetic strength for a given set of parameters.

Air resonances

In addition to the main cone resonance, resonances exist that are functions of three enclosure dimensions which should be dissimilar to avoid resonances reinforcing each other. These can be damped by absorbent material placed at the points of maximum vibration, termed the antinodes. For the fundamental this is halfway along each dimension. For the second harmonic it is at one-third and two-thirds the length, and for the third harmonic it is one-sixth, one-half and five-sixths.

Panel resonances

The high internal pressures in a sealed box cause the panels to vibrate at their resonant frequency and radiate sound. These virations can be minimised by a solid well-braced construction. High density panels of metal, brick or concrete have been used, as well as cavity panels filled with sand. They are usually damped by lining with a heavy adsorbent. Some use light panels with heavy bituminous damping pads.

Reflected wave

The rear pressure wave is reflected from the rear panel back to the speaker and out through the cone. The delay produces reinforcements at one-quarter and three-quarter wavelengths and cancellation at one-half and whole values, resulting in response peaks and troughs.

Damping

The degree of damping depends on the magnitude of the opposing force, which in turn depends on the efficiency of the system as a generator. Generator effeciency is governed by the magnetic flux density and the length of the coil, and inversely by the coil resistance, the cone inertia which depends on its mass, and the frequency of resonance.

The total Q comprising electrical Q(Q_{es}) and mechanical Q(Q_{ms}), which is designated Q_{ts}, is that of the driver only. The total including the cabinat is described as Q_{tc}. This of course cannot be specified by a manufacturer because he does not know the size of the cabinet in which the driver will be used, but it does appear in formulae which can be used to determine the optimum cabinet size.

If Q_{tc} is equal to unity, there is no peak at the resonant frequency because the amplitude of the cone excursion is just 1 times that any other frequency. This would appear to be the ideal value. However, an undamped response consists of a peak that is sharp at its tip while being fairly broad at its base. If now we level the tip to unity value, there is still a slight rise on either side due to the 'foothills' of the base. The lower one disappears due to the bass roll-off below resonance, but the upper one remains. A Q_{tc} of unity therefore produces a small rise just above the resonant frequency.

Enclosure volume

The formula for designing a sealed enclosure for a particular driver is complex one, but it can be greatly simplified if its use is restricted to a Q_{tc} of 0.7; for other values it is less accurate. The governing factors are :

- The compliance of the drive unit's suspension
- The volume of air having the same compliance
- The mechanical Q, Q_{ms} and the electrical Q, Q_{es}

These four factors are combined into two in the Thiele-Small parameters quoted in maker's specifications. The volume of a body of air having the same compliance as the suspension of the drive unit is given in litres and denoted by the term V_{as}. The mechanical and electrical Q are usually combined ; designated as Q_{ts} .

So, the formula is:

$$V_b = \frac{V_{as} * Q_{ts}^2}{Q_{tc}^2 - Q_{ts}^2}$$

and as Q_{tc} is 0.7, then :

$$V_b = \frac{V_{as} * Q_{ts}^2}{0.49 - Q_{ts}^2}$$

Not all bass drivers have the high compliance required for use in sealed enclosure, many are designed for open-backed systems. The formula is not intended for these and will give an incorrect answer if used for them.

Dimensional resonances

The infinite baffle enclosures are greatly affected by the internal air resonances. These are air resonances that are functions of the three enclosure dimensions. They occur at frequencies having half-wavelengths corresponding to the three cabinet dimensions.

If any two dimensions, or worse, all three are the same or nearly so, a very strong resonance is obtained. Hence height, width and depth must be different and one dimensional is not a multiple or part multiple of any other. The ideal enclosure ratio are 1:1.6: 2.5 or multiple of them.

One important disadvantage of the infinite baffle enclosure is the fact that radiation from the rear of the speaker is completely lost in the closed box. It cannot be made to do useful work because it has no way of getting out of the box. This condition is rectified in the "bass-reflex" enclosure which will be discussed in the following pages.

5.3.2 The bass-reflex enclosure

Until the advent of the infinite baffle system, the most popular hi-fi speaker was based on the vented or reflex enclosure. Again this is an artifice to provide the best possible bass response. Instead of the enclosure being completely sealed as in the infinite baffle system, it features an opening which is known variously as vent, port or tunnel.

Like the infinite baffle, the completely sealed enclosure is a relatively inefficient device, in that it does not permit energy from the rear of the speaker to radiate into the listening area. Efficiency is further reduced by the increased stiffness of the mechanical system brought about by the action of the air column trapped within the enclosure.

To make use of the energy lost within the sealed enclosure, a small hole which is usually two to three inches in diameter is cut in the face of the enclosure. Therefore the rear sound can escape freely. Usually, the hole will be placed below the speaker cutout as shown in figure 5.2a

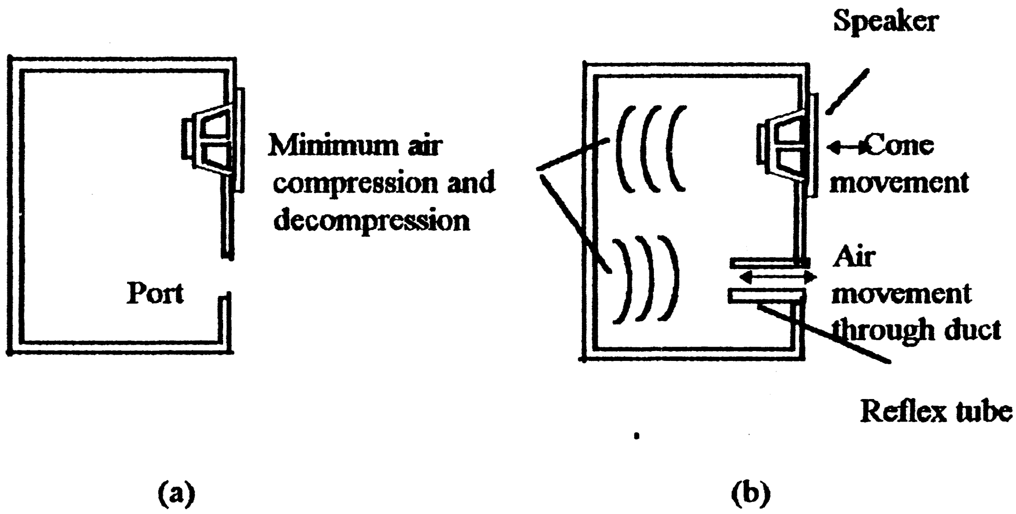


Figure 5.2: A simple view of Bass Reflex enclosure that used for higher power handling, monitoring loudspeakers etc.

In most designs, a tube is then inserted in the hole as shown in figure 5.2b. The tube lets a certain amount of air travel in and out of the enclosure as the speaker cone is in motion. On the other hand, the tube acts as an air duct, providing a partial vent for the compressing and decompressing air. The air in the duct or pipe can be considered as a wad having mass, inertia, and its own resonant frequency. The frequency is adjusted to be the same as that of the cone enclosure air volume resonance; so there are two resonances tuned to the same frequency. The effect is similar to that of the Helmholtz resonator or ported reflex type, which resonates at a frequency determined by the volume of the cavity and the length of the neck.

The action of the resonator is very much like the effect one gets when blowing over the mouth of a large bottle or a jug. Speaker motion excites the resonator and it emits sound from the port. When the port/box combination is properly tuned, low frequency air vibrations within the port are in phase with cone motion and output is increased. The deep bass efficiency of a ported reflex design can be 50 to 100% greater than that of an acoustic suspension system.

A longer tube containing more air would be needed to balance the larger mass of air in the larger enclosure, but the greater air mass is more easily compressed and so has a higher compliance. The tube length can best be determined experimentally by trying different lengths of 3 inch drain pipe while sweeping the bass frequencies with an audio oscillator and measuring the impedance with a series ammeter to give twin dips.

In the reflex type, the loudspeaker cone and suspension forms a series mechanical-resonant circuit, and the enclosed air volume forms a parallel resonance with the mass of the air associated with the port (actually the mass of the air in the port pipes plus the radiation mass of the air added to the orifice of the port). If the two resonances are tuned to the same frequency , i.e. the unit bass resonance coincides with the frequency to which the resonator is tuned, the result is a coupled double-resonant circuit exhibiting two resonant peaks. One is below and one is above the bass resonant frequency of the unit. If these two peaks can be damped to the correct degree, the bass response of the loudspeaker is extended considerably below the bass resonant frequency of the unit. In practice it is quite difficult to obtained the correct damping of both the resonant peaks in the overall bass response, unless a fairly large cabinet is used with correctly positioned acoustic resistance material mounted in exactly the right places. A common defect with compact reflex cabinets is that the upper bass response peak is not well damped, producing an unfortunate bass 'colouration' at a relatively high frequency (often 120-150Hz).

The formula (in imperial measurement) for calculating the volume of a reflex enclosure is :

$$V = \pi r^2 \left[\frac{4.66 * 10^6}{f^2 (L + 1.7r)} \right] + L \quad in^3$$

in which V is the enclosure volume ; L is the length of the pipe (in inches); r is the effective cone radius (in square inches); f is the cone resonant frequency . In this case the area of the port must be the same as that of the cone.

The formula for metric measurements, with all dimensions in cm, is:

$$V = \pi r^2 \left[\frac{304 * 10^6}{f^2 (L + 1.7r)}\right] + L \ cm^3$$

A simpler alternative formula for calculating the volume of the enclosure in litres (a litre is 1,000,000 cubic milimetres or 61 cubic inches), using Thiele-Small parameters is :

$$V_b = 20V_{as} * Q_{ts}^{3.3}$$

in which V_{as} is the volume of air having the same compliance as the drive unit suspensions and Q_{ts} is the total driver Q.

This does not give the tube area and length, and as the two resonant systems must be matched, the dimensions are critical. The resonant frequency depends in the mass of air in the tube, and also its compliance. If mass was the only factor, the calculation would simply involve the volume of air in the tube, which is proportional to its area and length.

5.3.3 The Labyrinth/Transmission line enclosure

One of the most satisfactory solutions to the rear wave disposal problem. The enclosure contains a long passage formed by internal baffles similar to the folded horn, but the passage is longer and the area does not increase. The operation is similar to a long, 'lossy' electrical transmission line in which most of the energy is lost and very little appears at the end to be reflected back. It was claimed that, when properly designed, the system behaved like the coaxial cable that connects a properly-matched radio transmitter with its antenna. It carried the sound waves down the line without allowing any to reflect back at the driver itself, thus the name 'transmission line'.

A key to the success of the transmission-line enclosure is the use of damping materials at the rate of 1/2 -pound-per-cubic-foot. A cutaway view of a transmission-line speaker is shown below (figure 5.3)

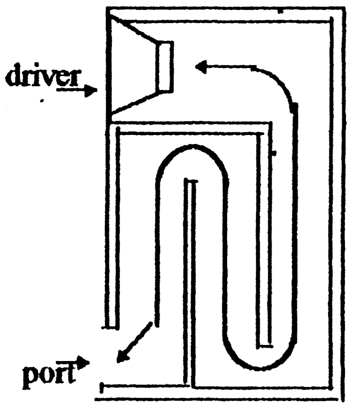


Figure 5.3: A cutaway view of Transmission line enclosure

Length of the line is made a quarter wavelength of the resonant frequency of the bass driver, or a little longer. The sound appearing at the end then is in phase with the front wave and reinforces it as a reflex enclosure does. Ideally, the area of the line should be the same throughout its length, being equal to the area of the cone. In as much as the line is folded, resonances are likely over the individual sections due to back reflections, as well as the complete length, especially if the bends are sharp. These can be spread by tapering the area along its length, but can better be dealt with by arranging for the sound to be completely reflected around the bends.

The bends are often left without means of reflection, but the performance is improved by reflectors placed at a suitable angle in the bend. These usually consist of a flat wooden board, but being in the direct path of the pressure wave, they can absorb part of the energy and set up vibrations at their own resonant frequencies, thereby adding colouration. Ceramic tiles have proved very succesful for reflection especially if bedded on a shallow concrete wedge in the corner of the bend. For a U bend, two tiles in a V formation at a 90^0 angle to each other give excellent result .

Resonant pipe

The transmission line also behaves as a tuned resonant pipe closed at one end. Such a pipe has a strong fundamental resonant frequency which is utilised to produce an in-phase output at the vent. However, any resonant chamber stores energy which is released after the source has ceased. The fundamental should be well damped by filling the pipe with absorbent material in order to reduce coloration from this cause. The node is at the closed (loudspeaker) end and the antinode at the open end. By packing the material more densely at the vent, damping is effected.

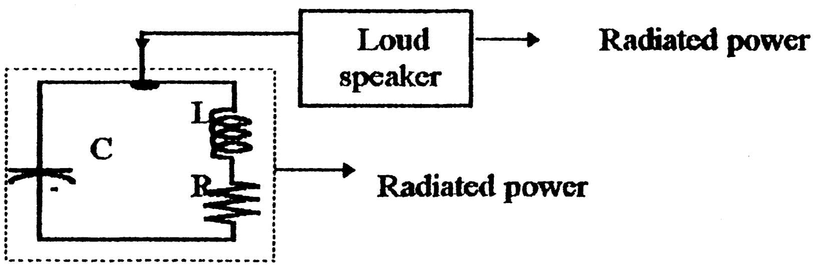
A closed pipe generates no even harmonics, so there is no second. The third harmonics has antinodes at the one third and open end positions. The packing arranged at the vent for the fundamental also serves to dampen the latter, and an increase in density at the one third position takes care of the first antinode. There is no fourth harmonic and the fifth is sufficiently weak to be well damped by the general filling .

5.4 THE ENCLOSURE AS A CROSSOVER ELEMENT

Crossover networks are vital components of multi-speaker system. They perform the important function of segregating specific bands of energy into specific channels.

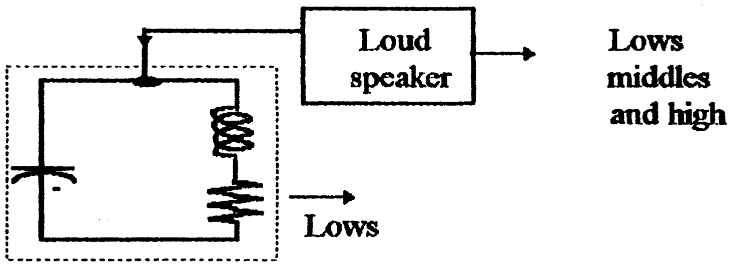
This section would elaborates the function of enclosure in providing the acoustical crossover characteristic. It will concern with the action of woofer enclosures as part of the acoustic network.

The bass-reflex enclosure is a resonant acoustic device which is being made up of an acoustic capacitance and an acoustic inductance, usually referred to as an inertance. This combination can be illustrated using electrical symbols as shown in figure 5.4

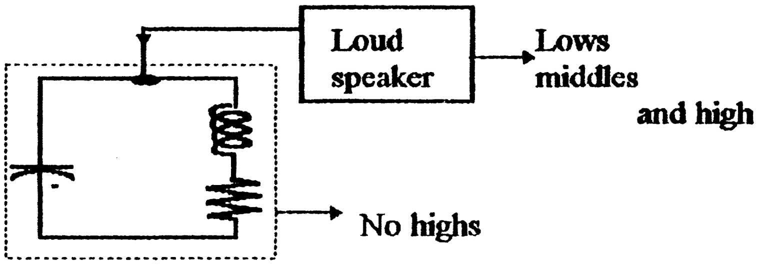


Bass reflex

C = Capacitance of enclosure volume
L = Inertance of port
R = Radiation resistance of port



For low frequencies, capacitance of enclosure offers a high opposition and the port is a low opposition passing the low frequencies into the port to be radiated



For high frequency, port of enclosure offers a high opposition blocking the passage of highs from the port radiation resistance.Highs get shunted into capacitance of enclosure volume.

Figure 5.4. The capacitance of the bass-reflex enclosure and the inertance of its port acts as L-C element of a crossover network, allowing lows from the port and restricting middles and highs to the speaker.

The section shown within the dotted square represents the bass-reflex enclosure. The electrical symbol of the capacitor represents the volume (capacity) of the enclosure, while the coil represents of the inertance of the air in the port of the enclosure. Lastly, the electrical symbol of resistance represents the radiation resistance of the air load that the port sees.

The speaker can be assumed as a generator of electrical signals of all frequencies. The signals is then transmitted through the enclosure network that is connected to it. The choke coil will pass low frequencies and block high frequencies, that is, its reactance goes up with frequency. Thus, when the generator (loudspeaker) is producing low frequency signals, these signals will pass through the choke (the port) into the resistive load (radiation resistance of port), and power will be developed in the resistance.In acoustic terms, we can say that when the loudspeaker is reproducing low notes, these notes will come out of the port and into the acoustic radiation resistance of the port, and low frequency power will flow out through the port.

If we raise the frequency of the signal being generated, the choke will begin to offer more and more opposition to the passage of the signals of increasing frequency, and fewer will get to the resistance. At the same time, the capacitor will begin to offer less opposition to those same signals. The higher frequency currents will thus begin to flow through the capacitor as they cease to flow in the other branch of the network. Again, in acoustic term, as the frequency of reproduction of the loudspeaker goes up, the port "closes up" acoustically, and those higher frequencies find their way into the acoustic capacitance volume of the chamber, instead of into a useful radiation resistance. As these higher frequencies enter the capacitance, no acoustic work is produced. Instead, these higher frequencies bounce back and forth from wall to wall within the cavity unless proper soundproofing is applied to the interior walls. Hence, the bass-reflex enclosure acts as a true crossover element, working entirely on acoustic principles.

At low frequencies, it transmits sound from the port and there is both forward and rear transmission of power from the loudspeaker outside. At higher frequencies, the box closes up acoustically, and only direct forward radiation takes place from the diaphragm.

5.5 ENCLOSURE CONSTRUCTION

Generally, enclosures and baffles are means of transferring acoustic energy from the loudspeaker to the surrounding air. Whether the enclosure is a simple closed box or the most complex of horn structures, its main purpose is to cause the sound from the loudspeaker to be most efficiently radiated into free space so that it may be properly heard and enjoyed. Obviously, they do not produce any acoustic power by thenselves; neither should they absorb any power, for if they do, the total acoustic efficiency of the system will suffer.

The key to loudspeaker enclosure construction is rigidity, or resistance to vibration. The wooden panels must be rigid so that they will not be vibrated by the sound pressures within the enclosure. If they are vibrated, acoustic work is done on them and acoustic power is wasted. Therefore, the right material should be choosen to fulfill the requirements as we will discuss further.

5.5.1 Mechanical aspects

Most loudspeaker cabinets are constructed from various forms of timber as it generally fulfils most of the basic requirements as regards strength, workability and appearance. A number of processed or made-up timbers such as high-density chipboard and blockboard free from voids are now available as well as plywoods.

A fair amount of work has been done to asses the relevant properties of the various forms of wood as well as other materials such as metals, plastics, concrete, slate, marble etc, which may also be used to construct cabinet. The density, elasticity and internal losses are important in that they govern the thickness of panels of any given size in order to obtain the desired rigidity. Table 2 belows gives the density of woods and other materials as well as the mechanical Q determined from vibrational decay tests.

MATERIAL	DENSITY lb/ft3	EFFECTIVE MECHANICAL Q	WORKABILITY
Marble	160		Difficult
Concrete			
Boxwood	70	40	Moderate
Teak	60		Satisfactory
Oak	50	20	Moderate
Mahogany	40		Satisfactory
Pine	25	40	Satisfactory
Plywood (low density)	40		Satisfactory
Plywood (high density)	80	20-25	Satisfactory
Blockboard (without voids)	40-50		Satisfactory
Chipboard (high density)	50-60	20-30	Precautions needed at edges etc

Table 1: Mechanical properties of timbers etc, used in loudspeaker cabinet.

It is much more difficult to obtain sufficient rigidity in larger panels. As a general guide, it may be taken that wood of between 7/8 and 1 in thickness should be used for larger cabinet or baffles, whilst 1/2 to 3/4 in thickness may be found satisfactory for the smaller bookshelf type of enclosed cabinet. The smaller panels used in the smaller cabinets will probably also be sufficiently rigid not to need any internal surface damping treatment or bracing.

Precise construction and good overlaps on all joints are needed in order to give strength and to avoid air leaks. Removable rear panels and terminal plates may require particular attention in this respect.

Loudspeaker units must be firmly bolted or screwed to a flat front mounting plate. The fixings must be strong so as to avoid the unit working loose in transit. Non-magnetic screws of brass or bronze are preferred in order to avoid the risk of steel slivers or swarf being produced during assembly. Similarly, all other materials and tools must be swarf-free in order to avoid any risk of subsequent gap contamination. The loudspeaker chassis sealing ring will make a good joint and the chassis will be free from mechanical distortion if the mounting surface is suitably smooth and flat.

The physical proportions of loudspeaker cabinets are dictated by the size and the mounting positions of the motor unit, the acoustic constants involved and the desired shape and means of mounting proposed for the cabinet. Examples of good constructions practise is shown in appendix 1.

5.5.2 Acoustic properties of cabinets

The acoustic design of a cabinet involves the realization of the correct acoustic termination to the rear side of the loudspeaker diaphragm. At low frequencies, the enclosed air behaves as a simple stiffness, but at frequencies over those given by $f = c/2l$, standing waves will develop. Some of these will be effectively coupled to the diaphragm and may represent high-Q resonances causing middle and higher frequency colouration.

In order to get the greatest acoustic damping, the material (absorbing material such as wool, glass fibre, expanded polystyrene, corrugated paper etc) must be applied at a point in the wave where the particle velocity is high. Thus material applied to the walls is not so effective as rolls of material or curtains hung in the centre of the cabinet. Perforated partitions with fabric of a suitable flow resistance over the holes may be mounted inside the cabinet. If the packing in the cabinet is sufficiently extensive and has the correct thermal properties, it can have a useful effect in reducing the effective stiffness of the air volume by up to 40% at low frequencies. This is due to absorption of heat from the air during the cycle by changing the effective mode of operation from adiabatic to isothermal operation.

Example :

Table 2 :The optimum volume of infinite-baffle enclosed loudspeaker cabinet

Nominal diameter of loudspeaker unit	6 in	8 in	10 in	12 in	15 in	18 in
Maximum power input 4 in watts	6	10	15	25	50	
Low-frequency cut-off 130Hz	90Hz	70Hz	55Hz	45Hz	35Hz	
Required volume of cabinet in cubic feet	0.35	0.9	2.2	4	9	16.5

The cabinet should be slightly but firmly packed with a soft absorbing material in order to obtain an approach to isothermal working.

CHAPTER 6

LOUDSPEAKER DESIGN AND SELECTION

Loudspeaker design may seem more than a bit mysterious, using various arcane terms such as *Thiele/Small parameters* and *resonance*. Indeed, for many building a speaker, simply means buying some "raw drivers" and putting them in any old cabinet they might have around, and hoping for the best. Typically, the result is a big disappointed.

Therefore, this chapter simplified the design of the loudspeaker with a few simple program to develop a loudspeaker. It followed by a briefly discussion about severel aspect in loudspeaker selection. However, we should start with understanding the number of parameters needed in order to produce a good loudspeaker design. As mention previously in chapter 2, speaker specifications is a part of parameters that being used in designing loudspeaker.

6.1 THIELE-SMALL PARAMETERS

Thiele-Small parameters is a numerical specification that today is almost universally supplied by most reputable driver manufacturers and retailers.

In the early 1970s, Neville Thiele and Richard Small calculated and quantified the various parameters affecting the performance of infinite baffle and ported reflex loudspeakers, giving these a designation which has become a standard. In many cases the design of enclosures can be simplified by the use of these parameters. The main parameters are as listed in table 3 :

Parameter	Description	Unit
Bl	Product of flux density and length of coil in magnetic gap (not total coil length)	Tm
C_{ms}	Acoustic compliance of suspension	$mN^{-1} * 10^{-6}$
d	Effective piston diameter	mm
f_s	Free air resonance	Hz
M_{ms}	Total moving mass of driver	gm
Q_{es}	Electrical Q factor	
Q_{ms}	Mechanical Q	
Q_{ts}	Total Q of driver	
Q_{tc}	Total Q of system including cabinet	
R_e	D.C resistance of coil	Ω
S	Piston range sensitivity	dB
V_{as}	Volume of air having same acoustic compliance as the suspension	Litres
$\frac{(Bl)^2}{R_e}$	Motor factor	

Table 3: Thiele-Small parameters

The total Qab system including the cabinet is described as Qtc. This of course cannot be specified by the manufacturer because he does not know the size of the cabinet in which the driver will be used, but it does appear in the formulae which can be used to calculate the optimum cabinet size.

V_{as}, the volume of air having the same acoustic compliance as the mechanical suspension of the driver is a particularly useful parameter when quoted by manufacturer and is one of those that simplifies the calculation of enclosure volumes.

6.2 DESIGNING LOUDSPEAKER

Designing loudspeaker might seem like an accult art to those who are not initiated in the techniques used. There are obviously a lot more involved that just hooking up a few drivers together and applying a signal. Now, with the help of the simple computer programs which is QBASIC programming presented in appendix 2 and appendix 3, we can help ourselves to create terrific sounding speaker system.

Admittedly, the speakers we can design using this programs are fairly basic which is written by William R. Hoffman. However, they are surprisingly good performers that can give us years of enjoyable liistening.

6.3 CHOOSING LOUDSPEAKER

Choosing loudspeaker system seems perfectly logical to acquire all the technical data, specifications and leaflets on the models which interest us most. However, the value depend on our technical background. We need to be aware of what is very important, what is not important and what is virtually of no importance.

6.3.1 The personal aspect

To know whether we really like a speaker and could live with it for a long time, it is essential to try it out under the exact conditions of ultimate listening. That is in the home with the audio system it is to partner. However, it is not feasible to try a speaker system at home before making a decision to purchase.

A large part of the difficulty of selecting a loudspeaker and its enclosure arises from the fact that the psychoacoustic factor involved in the reproduction of speech and music are not understood. Listeners will rank-order differently four apparently identical loudspeakers placed in four identical enclosures. It has been remarked that if one selects his own components, build his own enclosure, and is convinced he has made a wise choice of design, then his own loudspeaker sound better to him than does anyone else's loudspeaker. In this case, the frequency response of the loudspeaker seems to play only a minor part in forming a person's opinion.

6.3.2 Objective parameters

The objective parameters are one that can be measured by instrument. For instant, the pressure response, directivity, distortation, sesitivity, transicnt response etc.

Briefly, the ***pressure response*** curve by itself does not tell the intending purchaser much about how the speaker system will sound in his own house. Nevertheless, it is an important tool in the designing and ultimate tailoring of a system for the best overall results and smoothness of response.

The ***sensitivity*** of loudspeaker relates the acoustic output power to the input electric power. For example, if 10W electric input is required at a given frequency to produce 0.1 W of acoustic power, then the speaker can be said to have a sensitivity or efficiency of 1% at that frequency. Small speaker generally are less sensitive and the large horn are most sensitive.

The ***directivity*** factor as mention in chapter 2 can be important under certain listening condition. The severe changes in directivity over the frequency range are not desirable as they can impair the stereo image, giving the impression of 'stereo instability'.

Since music waveforms differ from simple sinwave in that they have fast-rising and fast-falling leading and trailing sides and are often of very short duration, the speaker must respond to such sudden and speedy changes if colouration and lack of definition are to be avoided. This means that the moment of inertia of the moving system must be small and the damping is good. Otherwise the signal might come and go before the cone or diaphragm can possibly move. Inadequate damping can colour the reproduction with a disturbing 'overhang' effect while poor low frequency damping yields a 'boomy' bass effect. Thus, we should listen for this when selecting a speaker from listening test.

6.3.3 Subjective factors

Subjective factors are those that rely on listening. Sometimes the nature of the sound produced by a speaker system is described by such term as 'smooth', 'gritty', 'rough' and so forth. For example, a speaker said to 'smooth' would almost certainly be devoid of sharp changes in output with frequency and the directivity would be fairly consistent over the frequency spectrum. Conversely, the term 'rough' might be applied to a speaker having violent undulations in its response characteristic.

The aim of designers is to create a speaker system which itself fails to 'colour' the reproduction. Sadly, all speakers have tendency towards colouration, some in more subtle ways than others.

Colouration is a general term used to express a specific character of sound. It is not present on the actual signal, which is introduced by the acoustics of the speaker itself. Thus, we should remember when listening to a speaker for colouration, the acoustics of the listening room are sometimes partly to blame, especially when the main lowest frequency resonance occurs towards the bas resonance of the speaker. Therefore, we should try the speaker in various position and listen to the sound it is producing in different parts of the room before we make the desicion to buy the speaker.

CHAPTER 7

RECOMMENDATION AND CONCLUSION

7.1 RECOMMENDATION

As a recommendation, I would like to suggest that we should study more further and details in order to achieve high knowledgement about speaker system. Therefore, the saler will have no chance to cheat us while buying the speaker. Besides, we can made the best decision when trying to buy a new speaker depending on its quality of sound.

For more interesting, we can use all the information about speaker system to build up our own loudspeaker with all the bass response we want without wasting many money to buy an expensive one with-performance speaker system.

Furthermore, we should expand the using of computer program to make the designing more easier, sophisticated and gaining the best result.

7.2 CONCLUSION

Although the loudspeaker is the prime source of sound in any reproducing system, the sound we actually hear when we listen to a radio, or phonograph is not entirely the result of the loudspeaker performance. We hear the result of many interacting factors, which constitute the subject matter of applied and practical acoustics. Not untill the electrical signal acts upon the loudspeaker mechanism is the signal transformed into sound waves.

Generally, as we can see that the basic of loudspeaker system are very simple and everyone can learn about it with a great understanding. However loudspeaker system is actually a very complex system to study. There are a lot of things involved and to be considered because the loudspeaker consist of many aspect with different sizes, types, function and material uses.
Thus, we should take into account many factor that involved in reproducing sound while trying to choose or designing the loudspeaker.

Typically, a well designed loudspeaker system is prized for having a very smooth, clean, pure and undistorted bass response.

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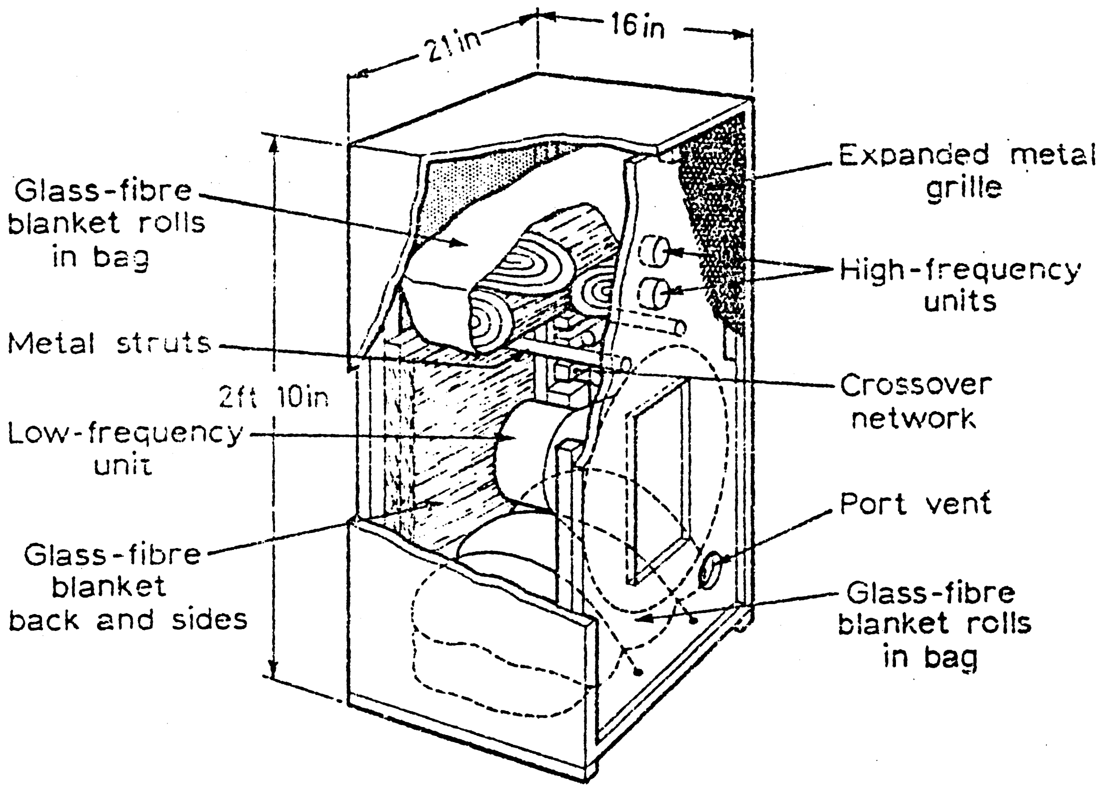
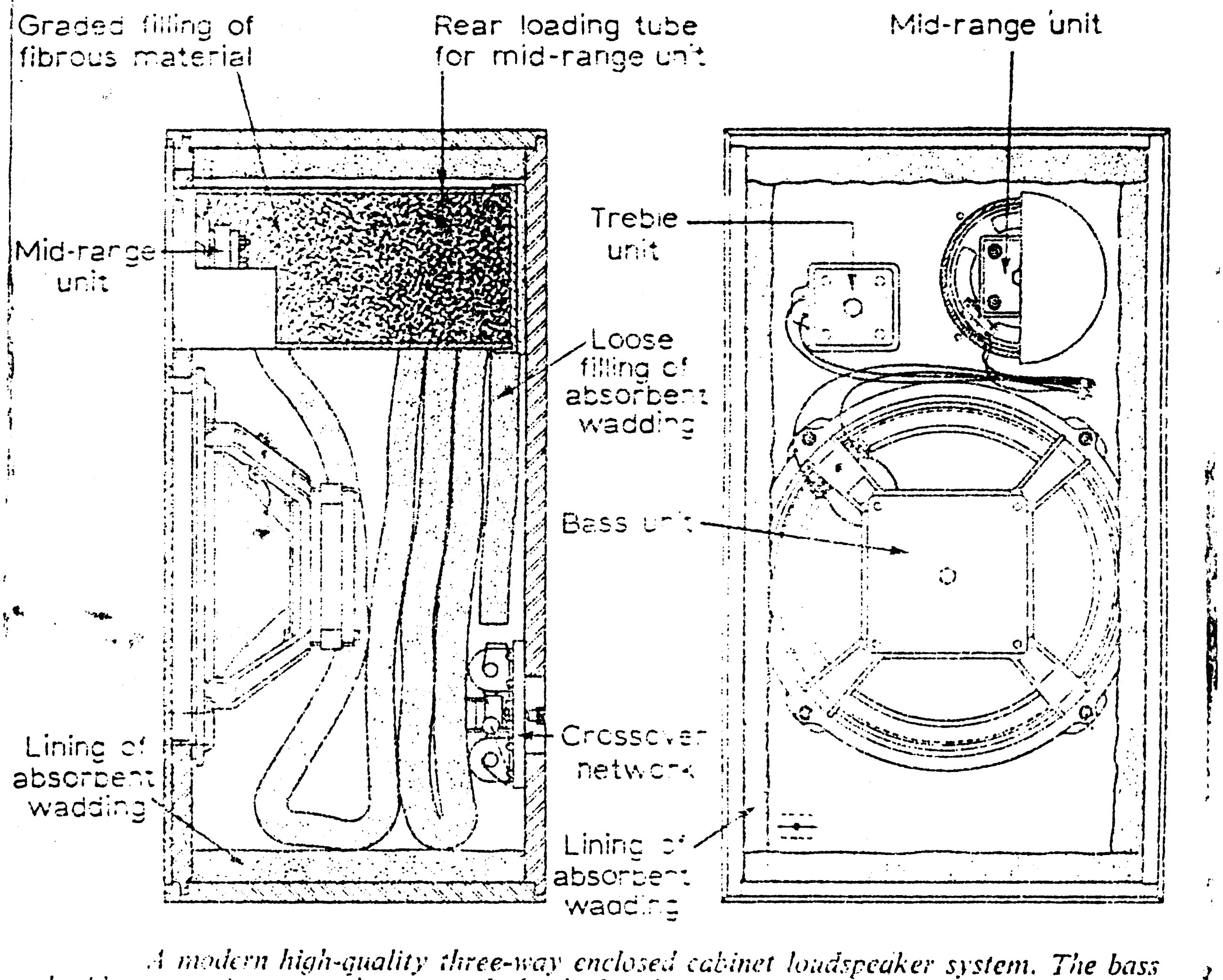
APPENDIX 1

MECHANICAL CONSTRUCTION OF

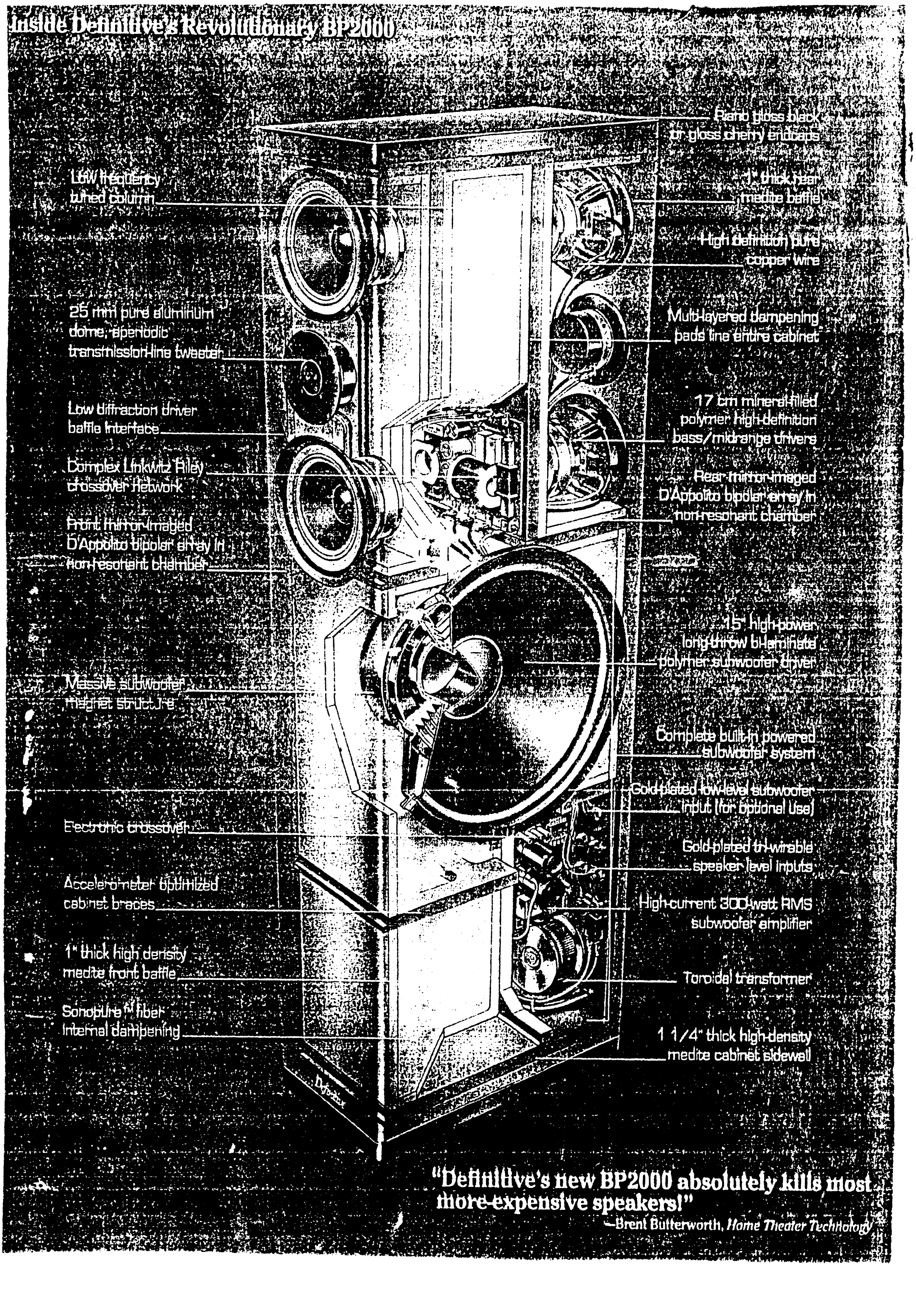
LOUDSPEAKER



A high power monitoring loudspeaker in a ported enclosure with carefully disposed internal damping and cabinet reinforcing struts. The low-frequency unit radiates through a relatively wide slit to match the polar response to that of the tweeter at crossover. (D. E. L. Shorter and BBC)



A modern high-quality three-way enclosed cabinet loudspeaker system. The bass and mid-range units are each separately backed with an optimum damped enclosure volume. The treble unit is totally enclosed within its rear case. (*Wharfedale*)


Inside Definitive's Revolutionary BP2000
Piano gloss black or gloss cherry endcaps
Low frequency tuned column
1" thick rear medite baffle
High definition pure copper wire
25 mm pure aluminum dome, aperiodic transmission-line tweeter
Multi-layered dampening pads line entire cabinet
Low diffraction driver baffle interface
17 cm mineral-filled polymer high-definition bass/midrange drivers
Complex Linkwitz Riley crossover network
Rear mirror-imaged D'Appolito bipolar array in non-resonant chamber
Front mirror-imaged D'Appolito bipolar array in non-resonant chamber
15" high-power long-throw bi-laminate polymer subwoofer driver
Massive subwoofer magnet structure
Complete built-in powered subwoofer system
Gold-plated low-level subwoofer input (for optional use)
Electronic crossover
Gold-plated tri-wirable speaker level inputs
Accelerometer optimized cabinet braces
High-current 300-watt RMS subwoofer amplifier
1" thick high density medite front baffle
Toroidal transformer
Sonopure™ fiber internal dampening
1 1/4" thick high-density medite cabinet sidewall
"Definitive's new BP2000 absolutely kills most more-expensive speakers!"
—Brent Butterworth, Home Theater Technology

APPENDIX 2

THE BASIC PROGRAMMING

There are a couple of preliminary concept of speaker design that we must first introduce. As mention above, we should consider the parameters that Thiele and Small have given us in order to design the loudspeaker. Then, we put these parameters into the formulae to obtain the value needed in designing loudspeaker. As a result, we have to calculate many times to achieve the proper values. It is wasting time of course.

Therefore, to make the calculation easy, we can use a simple program to make it faster without being worried about the calculation error. This section will provide some program listing to calculate the values to design loudspeaker such as the volume of the enclosure for acoustic suspension loudspeaker and etc which will discuss afterwards.

As mention previously, the program listed are written in "QBASIC" by William R. Hoffman in his articles of Popular Electronic and also mention in references.

1.1 Designing low frequency acoustic suspension loudspeaker

To begin with, we need to know the three parameters of Thiele and Small which is commonly abbreviated 'fo', 'Qts' and 'Vas'. As mentioned earlier, those parameters are commonly supplied by a driver's manufacturer. However, in this programming, we use any value for trial and error calculation. The purpose of this program are to calculate the volume of the enclosure and the bass resonance frequency of the system.

Now, consider the fo, or the frequency of oscillation. That refers to the natural bass-resonance frequency of a loudspeaker driver without a cabinet. It will be used to determine how low in frequency of completed system will efficiently go, or $f_{cab.}$ It is important to recognize that a given size driver has a fairly well defined lower frequency limit (fo). That is because the movement of the cone of a driver must quadruple for every halving of frequency to maintain the same acoustic output.

Table 4 indicate some practical values that can be used in our design work. Attempting to push a driver much below those recommended frequency limits will usually result in weak or distorted output, or even to the driver damage.

Driver sizes (inches)	Low limit (Hz)
4	90 - 95
5	85 - 90
6	65 - 70
8	40 - 45
10	30 - 35
12	25 - 30
15	20 - 25

Table 4 : Lowest frequencies

Secondly is Qts, which is the degree of peaking that occurs in the driver's response, again without a cabinet, at the bass-resonance frequency. That is related to the Q within a cabinet, which known as Q_{cab}. The parameter Q_{cab} can be decided more or less by the designer. Typically, the higher the Q, the stronger the bass output will be and the faster the speaker output will fall off below fcab. Also, just as in an electrical circuit, too high a Q can cause ringing and oscillation, which means highly distorted bass.

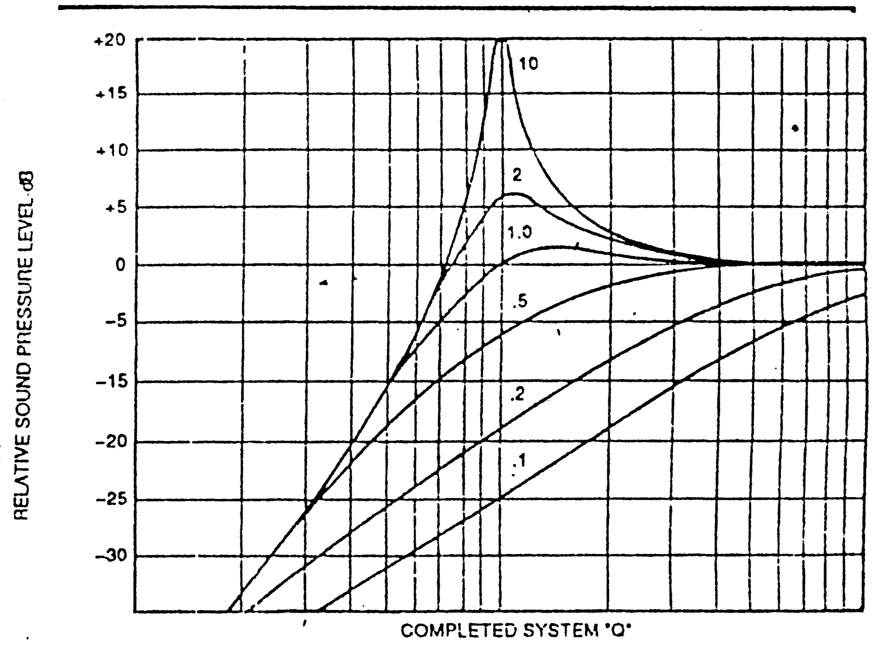


Figure 6.1: Q curves

Figure 6.1 shows some Q curves possible with a completed speaker system. Notice that for values much over 1, a large peak in the output occurs, growing with increasing Q. This peak should be avoided in order to get a good sound. Furthermore, as a guide, table 5 indicates suggested values of Q for various bass-driver sizes.

Driver sizes (inches)	Q value range
4 - 5	0.9 - 1.2
6	0.8 - 1.0
8	0.7 - 0.9
10	0.6 - 0.9
12 - 15	0.5 - 0.8

Table 5 : The possible Q's

Lastly is Vas which is a volume of air contained within a closed space that would have the same compliance or "springiness" as the suspension of the driver's moving part (the cone).

The program are as listed in ***appendix 3***. It also shows the value that being used and its result.

1.2 Designing a transmission line loudspeaker system

To begin with, we should consider two things. The first is the bass driver's diameter (in inches) and the second thing is the bass resonance frequency of the driver in free air, (fo).

Once we have loaded the program as listed in ***appendix 3***, and have started it running with the appropriate command-line prompt, the opening screen will ask for the two pieces of information about the bass driver we are using which had been mention above. Once those values are entered, the program will calculate and display on the screen five parameters of the design.

The first is "line length" which is the physical length of the folded line, in feet. So, it will be acoustically 1/4 wavelength long at the bass-resonance frequency of the driver. The next parameter displayed is "line Cross-Sectional Area" which is the minimum area allowed for the line, and equal to the driver's cone area. if we making it smaller than it should be, it might cause excessive turbulence in the air flow, or cause other non-linear effects.

The third parameter is the "vent Area" or the area of the port at the line's open end. Again, that is the same as the surface area of the driver, and can remain that size regardless of what cross-sectional area we choose for the line itself. The fourth is the "Approximate Box Volume" in cubic feet, which is simply the line's length multiplied by the cross-sectional area. Therefore, we might get some idea of the size of the speaker cabinet, but not the exact because it does not take into account the thickness of any of the wood.

The final parameter is "Weight Of The Damping Material", which is simply the calculated volume of the line multiplied by the weight of the stuffing material figured at 1/2-pound-per-cubic-foot.

With the information just calculated, we can draw out the design of the tansmission line system with a folded line path anyway we might like it to be. The design may use figure 5.3 in chapter 5 as a guideline in designing transmission line system

1.3 Designing speaker enclosure dimension

This is a program designed to calculate optimum internal dimensions which is height, width and depth as indicated in figure 6.2 for a loudspeaker enclosure when only the desired volume is known. The program is shown in ***appendix 3*** with its result.

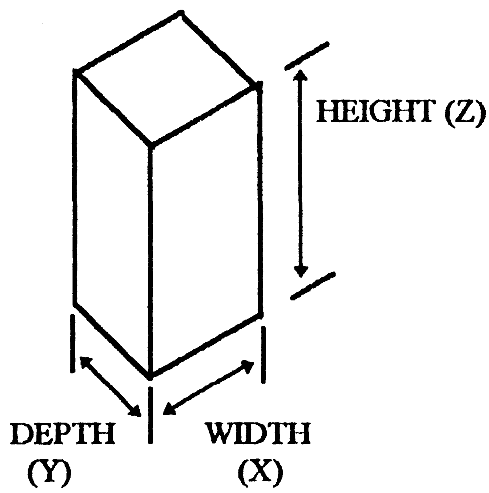


figure 6.2: Optimum internal dimension of enclosure

Once we enter the program, we will be asked for the volume of enclosure volume. Typically, the volume of the enclosure is determined by the size and characteristics of the woofer. An easier way to estimate enclosure volume is to use figure 6.3. The illustration shows acceptable enclosure volumes for woofers of varying sizes from 4 inches to 15 inches. Note that there is a great deal of flexibility in enclosure volume. A typical 12-inch woofer, for example, may be housed in an enclosure that has a volume of between 1.75 and 3.5 cubic feet.

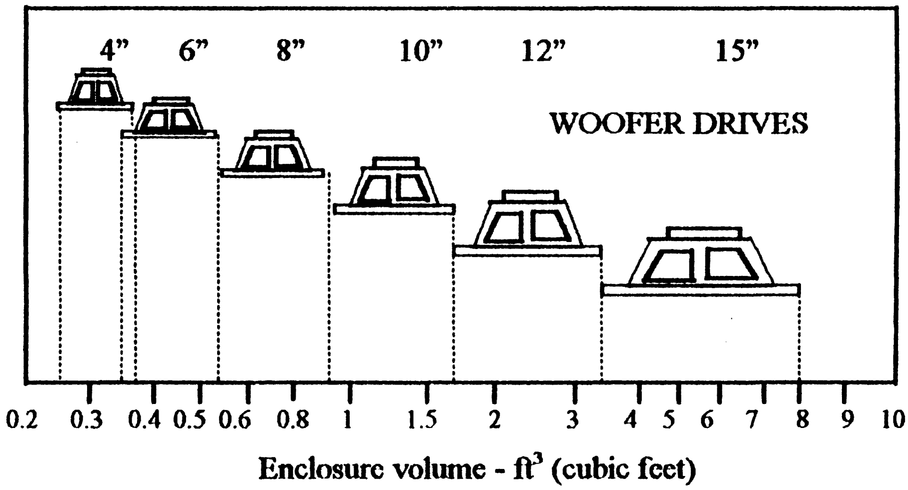


Figure 6.3 The chart which based on typical woofers, to estimate enclosure volume.

After the value of enclosure volume is entered, the program will calculate and display a set of values for the three inside dimensions, in inches. With these, then we can start build up the speaker.

1.4 Designing a loudspeaker crossover network

As mention previously, a crossover network used to prevent large drivers or "woofers" from distorting while trying to reproduce highs, and small drivers called "tweeters" from blowing out from lows. A crossover separates the sonic frequency spectrum into the ranges needed by each driver in a multi-way loudspeaker. More specifically, it is a combination of inductors and capacitors that form an electronic filter network.

Therefore, this section will give a simple program to design a two-way speaker which we will use to calculate the values of the inductance and capacitance for crossover network. As with the previous program, the program also written in QBASIC by William R. Hoffman and the listing are shown in ***appendix 3***.

APPENDIX 3

1. LOW FREQUENCY ACOUSTIC SUSPENSION LOUDSPEAKER DESIGN PROGRAM.

Running the program

If we wish to proceed, after answering "1" (yes), then we will be asked for the values of fo, Vas and Qts. Then we will be prompted for a value for Qcab, the completed system Q at bass resonance. That value must always be higher than Qts.

Once the values is entered, the program will automatically calculate the interior volume of a cabinet that yield the Qcab we specified.

For example, we use a Radio Shack 12-inch dual-voice-coil subwoofer:

Where,	Free air resonance	=	21 Hz
	V (as)	=	13.3 cubic feet.
	Q (ts)	=	0.38
	Q b	=	0.95

Program Listing

```
5  PRINT : PRINT : PRINT : PRINT
6  PRINT " LOW FREQUENCY ACOUSTIC SUSPENSION LOUDSPEAKER"
10 PRINT : PRINT : PRINT
20 INPUT " PROCEED?(1=YES,2=NO) AND ENTER"; P
22 PRINT : PRINT : PRINT : PRINT
25 ON P GOTO 30, 990
30 INPUT " FREE AIR RESONANCE OF DRIVER(Hz)="; B
40 PRINT " AIR VOLUME COMPLIANCE EQUIVALENT OF"
45 INPUT " DRIVER(Vas) IN CU.ft,="; C
50 PRINT " FREE AIR TOTAL Q OF DRIVER AT"
55 INPUT " BASS RESONANCE (Qts)="; D
65 PRINT " WHAT IS DESIRED COMPLETED SYSTEM Q"
70 INPUT " AT BASS RESONANCE (Qcab)="; E
75 PRINT : PRINT
100 LET Z = (E / D) ^ 2 * (1.05) - 1
110 LET Y = B * E / D
120 LET X = C / Z
130 PRINT " FOR AN IN CABINET BASS DRIVER Q OF E"
140 PRINT " CABINET VOLUME ="; X; "cu.ft"
150 PRINT " AND THE DRIVER BASS RESONANCE FREQUENCY"
155 PRINT " WILL BE ="; Y; "Hz"
160 PRINT
165 PRINT " IF YOU WISH TO PRINT THIS INFORMATION,"
170 PRINT " USE <PRN SCR> KEY AT THIS TIME."
172 PRINT
175 PRINT " WOULD YOU LIKE TO TRY ANOTHER DRIVER?"
180 INPUT " (1=YES, 2=NO) AND ENTER"; F
182 PRINT
185 ON F GOTO 30, 990
187 PRINT
990 PRINT " GOOD LISTENING"
999 END
```

The result

```
PROCEED?(1=YES,2=NO) AND ENTER? 1


FREE AIR RESONANCE OF DRIVER(Hz)=? 21
AIR VOLUME COMPLIANCE EQUIVALENT OF
DRIVER(Vas) IN CU.ft,=? 13.3
FREE AIR TOTAL Q OF DRIVER AT
BASS RESONANCE (Qts)=? 0.38
WHAT IS DESIRED COMPLETED SYSTEM Q
AT BASS RESONANCE (Qcab)=? 0.95


FOR AN IN CABINET BASS DRIVER Q OF E
CABINET VOLUME = 2.391011 cu.ft
AND THE DRIVER BASS RESONANCE FREQUENCY
WILL BE = 52.5 Hz

IF YOU WISH TO PRINT THIS INFORMATION,
USE <PRN SCR> KEY AT THIS TIME.

     DO YOU WISH TO TRY ANOTHER SET OF SPECIFICATIONS?
     (1=YES,2=NO) AND ENTER=? 2

     GOOD LISTENING!


Press any key to continue
```

2. TRANSMISSION LINE DESIGN PROGRAM

Running the program

Once we enter the program, we will be asked for the bass driver's diameter in inches and the bass resonance frequency of the driver in free air, (fo). Then, the program will calculate and display the five parameters of the design.

For example, we also use a Radio Shack 12-inch dual-voice-coil subwoofer:

Where, Free air resonance = 21 Hz

Bass driver's diameter = 12 inches

Program Listing

```
145 PRINT : PRINT : PRINT
150 PRINT "        THE FREE AIR BASS RESONANCE FREQUENCY OF THE DRIVER"
155 INPUT "        (FO),IN HZ. IS = "; FO
165 INPUT "        THE DIAMETER,IN INCHES,OF THE BASS DRIVER IS="; D
200 LET A = (1130 / FO) / 4
210 LET F = 1 + (.5 / .074)
215 LET F = SQR(F)
220 LET F = 1130 / F
230 LET B = (F / FO) / 4
240 LET XS = (D / 2) ^ 2 * 3.1459
250 LET XF = XS / 144
260 LET BV = XF * B
270 LET WS = BV * .5
290 CLS : PRINT : PRINT : PRINT : PRINT : PRINT
300 PRINT "        FOR A BASS DRIVER"; D; "INCHES IN DIAMETER,WITH A FREE"
310 PRINT "        AIR RESONANCE OF"; FO; "HZ,AND AN ENCLOSURE PACKED WITH"
320 PRINT "        DAMPING MATERIAL AT 1/2 IB.PER CU. FT, THE BASIC"
322 PRINT "        SPECIFICATIONS FOR A TRANSMISSION LINE ENCLOSURE ARE:"
325 PRINT
350 PRINT "        LINE LENGTH="; B; "FEET."
360 PRINT "        LINE CROSSECTIONAL AREA="; XS; "SQ.IN."
370 PRINT "        VENT AREA="; XS; "SQ.IN."
380 PRINT "        APPROXIMATE BOX VOLUME="; BV; "CU.FT."
390 PRINT "        WEIGHT OF DAMPING MATERIAL="; WS; "IBS."
395 PRINT
400 PRINT "        DO YOU WISH TO TRY ANOTHER SET OF SPECIFICATIONS?"
410 INPUT "        (1=YES,2=NO) AND ENTER="; X
420 IF X = 1 THEN 145 ELSE 990
990 PRINT
995 PRINT "        GOOD LISTENING!"
999 END
```

The result

```
THE FREE AIR BASS RESONANCE FREQUENCY OF THE DRIVER
(FO),IN HZ. IS = ? 42
THE DIAMETER,IN INCHES,OF THE BASS DRIVER IS=? 8



FOR A BASS DRIVER 8 INCHES IN DIAMETER,WITH A FREE
AIR RESONANCE OF 42 HZ,AND AN ENCLOSURE PACKED WITH
DAMPING MATERIAL AT 1/2 IB.PER CU. FT, THE BASIC
SPECIFICATIONS FOR A TRANSMISSION LINE ENCLOSURE ARE:

LINE LENGTH= 2.415066 FEET.
LINE CROSSECTIONAL AREA= 50.3344 SQ.IN.
VENT AREA= 50.3344 SQ.IN.
APPROXIMATE BOX VOLUME= .8441731 CU.FT.
WEIGHT OF DAMPING MATERIAL= .4220865 IBS.

DO YOU WISH TO TRY ANOTHER SET OF SPECIFICATIONS?
(1=YES,2=NO) AND ENTER=? 1
```

3. SPEAKER ENCLOSURE DIMENSION DESIGN PROGRAM

Running the program

If we wish to proceed, after answering "1" (yes), then we will be asked for the values of enclosure volume. Then, the program will calculate and display a set of values for the three inside dimensions, in inches.

For example, we use a Radio Shack 12-inch dual-voice-coil subwoofer:

Where, Enclosure volume = 2.5 cubic feet

Program Listing

```
20 PRINT : PRINT : PRINT : PRINT
27 INPUT " PROCEED?(1=YES 2=NO) AND ENTER"; P
28 PRINT : PRINT : PRINT : PRINT
30 ON P GOTO 40, 990
35 PRINT : PRINT : PRINT : PRINT
40 PRINT " THE DESIRED ENCLOSURE INTERNAL"
41 INPUT "  VOLUME IS (cu.ft)="; A
43 PRINT : PRINT : PRINT
45 ON A GOTO 50
50 LET X = A ^ (1 / 3) * (12)
55 LET Y = X * .62
60 LET Z = X * 1.62
70 PRINT " THE FOLLOWING  ARE THE INSIDE DIMENSIONS OF A BOX"
71 PRINT " OF"; A; " CUBIC FEET THAT ARE OPTIMIZED TO PROVIDE THE"
72 PRINT " LEAST NUMBER OF STANDING WAVES (OR RESONANCE)"
75 PRINT : PRINT :
80 PRINT " HEIGHT="; Z
85 PRINT " WIDTH="; X
90 PRINT " DEPTH="; Y
95 PRINT : PRINT
100 PRINT " DO YOU WISH TO TRY ANOTHER VOLUME?"
110 INPUT " (1=YES 2=NO) AND ENTER"; B
120 ON B GOTO 35, 990
990 PRINT : PRINT
995 PRINT " GOOD LISTENING"
999 END
```

The result

```
PROCEED?(1=YES 2=NO) AND ENTER? 1



THE DESIRED ENCLOSURE INTERNAL
 VOLUME IS (cu.ft)=? 2.5


THE FOLLOWING  ARE THE INSIDE DIMENSIONS OF A BOX
OF 2.5  CUBIC FEET THAT ARE OPTIMIZED TO PROVIDE THE
LEAST NUMBER OF STANDING WAVES (OR RESONANCE)


HEIGHT= 26.38414
WIDTH= 16.28651
DEPTH= 10.09763


DO YOU WISH TO TRY ANOTHER VOLUME?
(1=YES 2=NO) AND ENTER? 2


GOOD LISTENING
```

4. LOUDSPEAKER CROSSOVER NETWORK DESIGN PROGRAM

Running the program

Once the opening screen appears, we choose "1" to proceed the program. Then we will be prompted for the frequency in Hz that we need the crossover function to take place at. It is known as crossover frequency and we will take 3kHz as an example for this program. Then we will be asked for the impedance of the bass driver at the crossover frequency we have just selected (3kHz). The value of the impedance may found from an impedance curve supplied by the manufacturer. If not, we can use an impedance bridge to measure the driver's impedance. However, we also can get the value from the nominal impedance of the driver which is usually 8 ohms as used in this program

Finally, we will be prompted for the tweeter's impedance at the selected crossover frequency which is also 3kHz. To find that value, we can use the same strategy mentioned above for the woofer which is using the nominal impedance of the driver if necessary. The value is of course 8 ohms.

Once all the above information has been entered, the program will automatically calculate the inductor and capacitor values. Therefore, we can use all these values to design the crossover network of the speaker.

Program Listing

```
10 PRINT : PRINT : PRINT
20 INPUT " PROCEED?(1=YES 2=NO) AND ENTER"; P
22 PRINT : PRINT : PRINT
25 ON P GOTO 30, 990
27 PRINT : PRINT : PRINT
30 INPUT " CROSSOVER FREQUENCY IS (Hz)="; A
40 PRINT " IMPEDANCE OF LOW FREQUENCY DRIVER AT"
45 INPUT " THE CROSSOVER FREQUENCY IS (OHMS)="; B
50 PRINT " IMPEDANCE OF HIGH FREQUENCY DRIVER AT"
55 INPUT "     THE CROSSOVER FREQUENCY IS (OHMS)="; C
60 LET X = ((B) / (6.28 * A)) * 1000
65 LET Y = ((1) / (C * 6.28 * A)) * 10 ^ 6
66 LET Z = ((1) / (B * 6.28 * A)) * 10 ^ 6
70 PRINT : PRINT : PRINT
75 PRINT " L="; X; "mHy"
80 PRINT : PRINT
85 PRINT " C1="; Y; "uF"
86 PRINT : PRINT
87 PRINT " C2="; Z; "uF"
90 PRINT : PRINT : PRINT
95 PRINT " WOULD YOU LIKE TO TRY ANOTHER SET OF"
96 INPUT " VALUES?(1=YES 2=NO) AND ENTER"; Q
97 ON Q GOTO 27, 990
990 PRINT : PRINT
995 PRINT " GOOD LISTENING"
999 END
```

The result

```
PROCEED?(1=YES 2=NO) AND ENTER? 1


CROSSOVER FREQUENCY IS (Hz)=? 3000
IMPEDANCE OF LOW FREQUENCY DRIVER AT
THE CROSSOVER FREQUENCY IS (OHMS)=? 8
IMPEDANCE OF HIGH FREQUENCY DRIVER AT
    THE CROSSOVER FREQUENCY IS (OHMS)=? 8


L= .4246284 mHy

C1= 6.63482 uF

C2= 6.63482 uF

     DO YOU WISH TO TRY ANOTHER SET OF SPECIFICATIONS?
     (1=YES,2=NO) AND ENTER=? 2

     GOOD LISTENING!


Press any key to continue
```