ACTIVE LOUDSPEAKER SYSTEM

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ABSTRACT

Loudspeaker is one of electronic equipment that functioned by converting electrical energy into acoustical energy. Although the loudspeakers seem to have the same function, indeed, not all the system is alike. Each system has its own characteristic, types and functions. As for my active loudspeaker system, the system has its own circuit including crossover and built-in power amplifier. An audio signal can be produced just by connecting a source to the system. The crossover circuit applied in the system is placed between the input and power amplifier. It consists of two filters; low pass and high pass filter. They will split the signal into 2 frequency ranges; low frequency range (0 to 3100 Hz) and high frequency range (3100 Hz – 20000 Hz). After that, the power amplifier stage will increase the signal amplitude for about 20dB to 28dB. Both output signals will then be delivered to two speakers; tweeter and subwoofer. The tweeter will produce the high frequency sounds while low frequency sounds is produced by subwoofer. The system also has its own speaker protection circuit to prevent the speaker from broken down. Finally, an enclosure is constructed as the system platform to place the circuit along with the speakers. Certain guideline like the enclosure type and its appropriate dimension is followed to make sure the final product meets the earlier expectation.

CHAPTER 1

INTRODUCTION

This chapter will discuss on the problem statement which contributes to the creation and development of this project, objective and scope of this project.

1.1 Problem Statement

In the last few decades, people are getting more interested in the application of audio equipment at home or car audio system. Day by day, the engineers are introducing new products in order to get a better sound system. From just for an entertainment purpose, the function of sound system has become much bigger in our daily life. We need it in the supermarket, assembly hall, lecture room and other places. There will be one time when we need a sound system but the sound system provided didn’t meet our expectation either the sound system has a bad quality or it involve many separated components. In order to provide just a single system to produce sound with good quality, I have come to a conclusion to produce this 2-way active loudspeaker system.

In order to produce a good quality active loudspeaker system, we need to know what are exactly the meant of "active loudspeaker system" term, the circuit need to be included in the system and other factors, which will be taken into consideration. All these things are discussed briefly in the following chapters. The next important thing that needs to be considered is the type of enclosure that will be chosen.

1.2 Project Objective

The main purpose of this project is to study and build an active loudspeaker system. The research of the system will help us on knowing and mastering the functions of each stage. As earlier information, active loudspeaker system is a system that has a built-in amplifier. A source can be connected directly to the system without having to use external power amplifier.

1.3 Project Scope

Firstly, all the possible ideas were listed and the best was chosen to build the active loudspeaker system. Before constructing the system, we have to know the basic concept of the system and what are the main functions of the system.

1.3.1 To Know the Function of Basic Elements in the System

The system will have 2 basic elements in it which consists of crossover and power amplifier. Each of the part has its own function and may differ from each other. Knowing the basic concept of each part will help us on determining the overall function of the system. The parameters that need to be considered are the cut-off frequency used in the crossover stage and the total power that will be produced by the power amplifier. Nevertheless, the types of speakers that will be used at the end link of the system also need to be considered.

1.3.2 The Procedures before Constructing the Physical Part

There are few steps should be analyzed before building, such as the dimension and the volume of the enclosure. By having these values, we will have rough idea of the enclosure size.

1.4 Expected Result

The expected result of this project is a fully functional 2-way active loudspeaker system.

CHAPTER 2

LITERATURE REVIEW

2.1 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 system may contain either a single unit, or two or more of them, depending on the design, cost and requirements.

2.1.1 Loudspeaker Principle

The loudspeaker converts the electrical energy into acoustical energy. If an electrical signal is applied to the speaker terminals, the speaker cone moves forward and backward in response to the electrical signal. The air around the speaker is pressurized and depressurized, producing sound waves. High frequency signals cause the speaker cone to vibrate quickly while low frequency signals cause the speaker cone to vibrate slowly. Speakers should be efficient, able to handle high power, have a flat frequency and should have minimum distortion.[1]

Basically, there are two main types of loudspeakers that are:

Direct radiator type
Indirect radiator type

These types also presented the principle that been used by the loudspeaker. The loudspeaker will use either direct radiator principle or indirect radiator principle.

2.1.1.1 Direct Radiator Type

This type is commonly used in home radio receiving sets, phonographs, and in small public-address system. The vibrating surface (diaphragm) is in direct contact with the air mass of the surrounding environment which means that the driver directly radiates its energy into the listening area. The driver applying the principle is smaller 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. To increase the physical size of the moving system, the diaphragm of a direct radiator is usually surrounded by a speaker cone.[2] Figure 2.1 shows an example of direct radiator type loudspeaker.

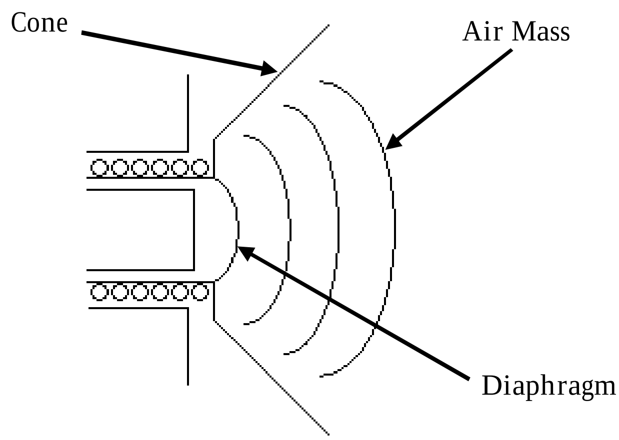

Figure 2.1 : Direct radiator type

2.1.1.2 Indirect Radiator Type

This type is used in high-fidelity reproducing systems, in large sound systems such as in theaters and auditoriums, and in music and outdoor-announcing system. The system consists 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.[2] Figure 2.2 shows an example of indirect radiator type loudspeaker.

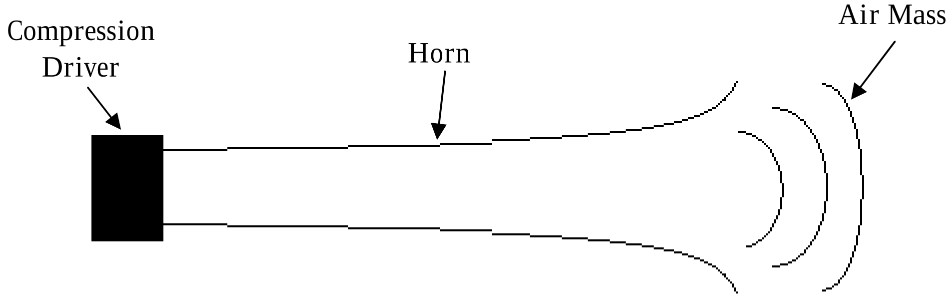

Figure 2.2 : Indirect radiator type

2.1.2 Types of Loudspeaker [1]

As mentioned before, the loudspeaker functioned based on 2 different principles. Although both seems different, but they have the same function that are to produce sound. From this 2 principle, the loudspeaker can be divided into several types.

2.1.2.1 Dynamic

This type of loudspeaker is the most common one. The construction of this type is shown as in Figure 2.3. It has a voice coil which is immersed in a fixed magnetic field. A powerful permanent magnet generates the fixed magnetic field, F1. The permanent magnet and the voice coil make up the driver of the speaker. The voice coil has many turns of fine wire wound on the bobbin. When an electrical audio signal is fed to the speaker, current flows through the voice coil, which generates a second varying magnetic field, F2. The interaction of the two magnetic fields produces motion and the diaphragm vibrates to produce sound waves. The voice coil bobbin is attached to the speaker cone. When the coil moves in response to an electrical signal, the bobbin moves the cone and causes it to vibrate. The dust cap forms the center of the cone and keeps dust and debris from entering into the small gap between the voice and the permanent magnet core.

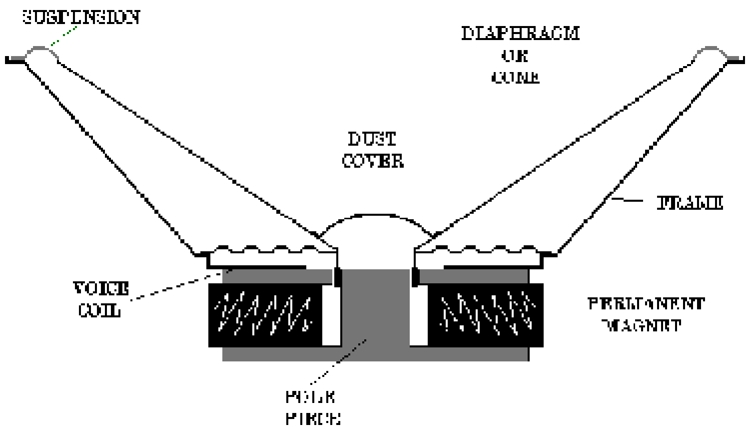

Figure 2.3 : Dynamic loudspeaker

The speaker suspension must be flexible because it must allow the speaker cone to vibrate. The suspension attaches the speaker cone to the speaker frame. The dynamic loudspeaker has a low AC resistance or impedance, in the range of 4 ohms to 16 ohms. The DC resistance of a dynamic loudspeaker is about 6 ohms.

2.1.2.2 Electrodynamic

It operates similarly to the dynamic loudspeaker. Like the dynamic loudspeaker, the electrodynamic loudspeaker is sensitive to current. Unlike the dynamic loudspeaker, the magnetic field of an electrodynamic loudspeaker is energized by an external power sources.

2.1.2.3 Condenser (or electrostatic)

This loudspeaker is sensitive to voltage that makes it has high impedance. It converts electrical audio signal into mechanical movements of the diaphragm. The diaphragm vibrations are caused by electrostatic forces of attraction and repulsion which are generated at the electrodes. The electrodes are energized by voltage to produce variation in capacitance. The electrodes have to be closely spaced; therefore, the electrostatic loudspeaker is not suitable for reproducing low frequency audio signals.

2.1.2.4 Crystal (or piezoelectric)

The loudspeaker operates on the theory that crystal expands or vibrates when an alternating electric current is applied to the surface of the crystal. The crystal loudspeaker has a very limited low frequency response and a low power output. The piezoelectric loudspeaker makes an excellent tweeter or high frequency speaker.

2.1.3 Loudspeaker Specification [1]

Each speaker has their very own characteristics or specification. The specification will determine the speaker's quality, the suitable circuit for it and the suitable enclosure.

2.1.3.1 Impedance

This is the parameter that shows the measurement of electrical resistance, in ohms, to AC signals. The speaker impedance should be the same as the output impedance of the driver amplifier. Speaker impedance is in the range of four to eight ohms.

2.1.3.2 Resistance

Resistance is showing the value of electrical resistance to DC signal measurement. It also counted in ohms. The value is five or six ohms.

2.1.3.3 Frequency Response

This parameter shows the range of frequencies of sound that the loudspeaker can produce. The subwoofer is a speaker designed to respond to frequencies less than about 2000 Hz. While the tweeter is designed to respond to the frequencies higher than about 4000 Hz, the midrange is responding to middle frequencies (between 1000 Hz and 5000 Hz).

2.1.3.4 Free-air Resonance

It is the frequency at which the speaker cone resonates. This specification is given for subwoofers only and is useful in designing bass-reflex speaker enclosures.

2.1.3.5 Moving Mass

The parameter shows the effective mass of all the moving parts of the loudspeaker. It is given for subwoofers and some midrange speakers. The moving mass and the speaker compliances (or stiffness) determine the free-air resonant frequency.

2.1.3.6 Compliance

It is the volume of air which has the same compliance as the speaker's suspension.

2.1.3.7 Sensitivity

Also known as Sound Pressure Level. It is the volume of sound produced by the speaker when it is fed one watt of electrical power within its frequency.

2.1.3.8 Power Rating

The peak power rating tells how much power the speaker can handle for only a very short time while the average power rating shows the amount of power that the speaker can handle continuously. The speaker system must have an average power rating which is equal to or greater the per-channel RMS power output of the amplifier that is used to drive the speaker.

2.1.3.9 Magnet Weight

It is the weight of the speaker's permanent magnet. The magnet weight affects the damping and the efficiency of the speaker. Subwoofers need large magnets because the speaker cone must move large distance to produce low frequency, high volume sounds. Meanwhile the tweeter needs a smaller magnet because it must move in a short distance to produce high frequency sound.

2.1.4 **Range of Sound** [3]

The audio spectrum of 20 Hz to 20 kHz can be subdivided into three major categories: low, medium and high as indicated in Figure 2.4 below.

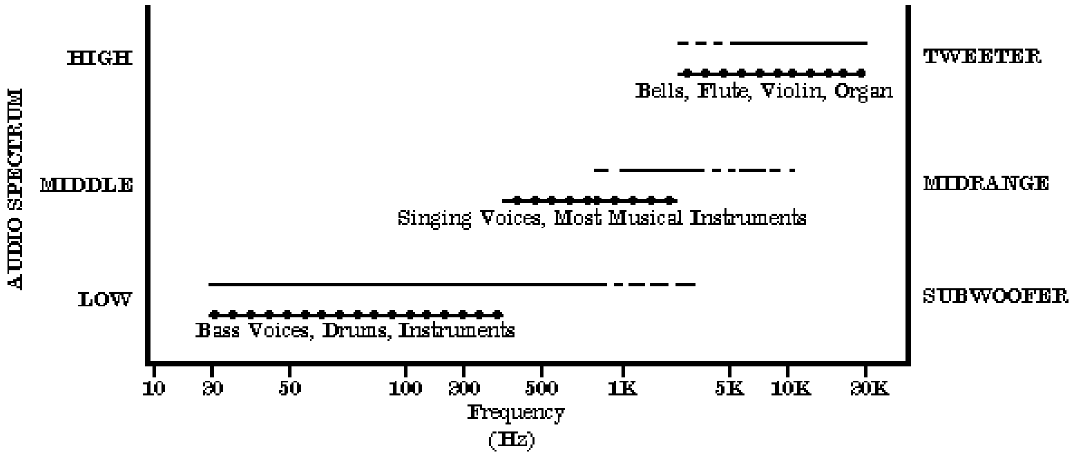

Figure 2.4 : Range of Sound

- Low sound (bass) – Made from bass instruments such as kettledrums, tubas, bassoon and string bass.
- Middle sound (midrange) – Made by most singing voices, guitars, and most other musical instruments.
- High sound (treble) – Made by bells, cymbals, flutes and violin.

There are specific speaker that has been built to cover each ranges of audio spectrum.

2.1.4.1 Subwoofer

This type of speaker designed to produce low frequency sound. Example of this type is shown on Figure 2.5 below. Generally, subwoofers cover the frequency range from 20 Hz to 2000 Hz. The size of subwoofer usually 4 to 15 inches in diameter. Felted paper or polypropylene is used to make subwoofer speaker cones.

Figure 2.5 : Subwoofer speaker [4]

2.1.4.2 Midrange

Midrange is a loudspeaker that designed to covers the middle frequency. Middle frequency includes from 1000 Hz through 5000 Hz. Midrange also producing some low frequency and high frequency sound. Other name for midrange; although less commonly used, is squawkers. Midranges are usually electrodynamic cone types or electrodynamic dome types, or compression horn drivers. They usually are smaller than subwoofer but bigger than tweeter. Midrange speaker is shown in Figure 2.6 below.

Figure 2.6 : Midrange speaker [4]

2.1.4.3 Tweeter

This type of speaker is designed to produce high frequency sound, typically from around 4000 Hz to 20000 Hz. Nearly all tweeters are electrodynamics type. They are usually 3 inches or less in diameter. Tweeter speaker are made of paper or cloth. Tweeter also, can is made from a metal dome as shown in Figure 2.7 below.[1]

Figure 2.7 : Tweeter speaker [4]

2.2 Crossover Network

The crossover network is required in an audio system that consist two or more type of speaker within an enclosure. The crossover network splits the incoming audio signal into appropriate frequency ranges. Each output is then been connected to its very own speaker. Each filter is designed to pass frequencies within its range and to ease all other frequencies. Different filter applies different type of principle to do its job.

There are 2 types of crossover network either it is passive or active network. The process of choosing what type of network need to be done properly because it will affects the performance of the audio system that we want to build.

2.2.1 Passive Crossover Network

This kind of network consists of a series of response shaping filters inserted at some point ahead of the speakers. Each filter comprises one or more inductor and capacitors whose values are determined by the desired crossover frequency and the impedance of the amplifier and the speaker, which are assumed to be equal.[1] The overall key point of the network is the crossover was placed after the power amplifier and before the load (speaker). Although it seems to be much easier to build, it has some drawback. For example :

- The amplifier used must have bigger output power because a sufficient power is needed to drive the complete speaker system.
- Because the crossover is placed after the power amplifier, there are some power is wasted within the network.
- If the speaker is placed by another speaker with different impedance, the network element values will have to be changed.

Figure 2.8 below shows block diagram of conventional audio system using passive crossover network.

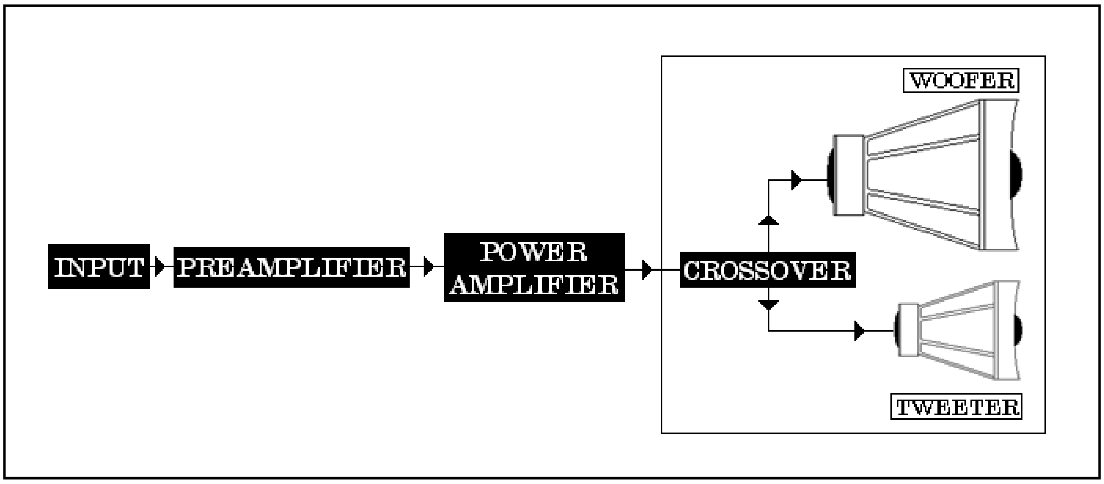

Figure 2.8 : Conventional audio system using passive crossover network

2.2.2 Active Crossover Network

Active Crossover Network is built by placing the crossover before the power amplifier. As the power amplifier is connected directly to the load (speaker), we can surely say that this network is as twice as efficient as passive. The assumption is made because there will be no more components between the speaker and power amplifier. As the speaker will receive all of the output that may come from the power amplifier, it is said that the speaker is working at its most optimal scenario. There are several advantages of an active crossover network compared to the passive one. For example :

- Amplifier power may be reduced to that required only by a single speaker
- There will be no passive components within the speaker line
- The speakers will work in its most optimal scenario because everything produced by the power amplifier goes directly to the speaker

Figure 2.9 below shows block diagram of conventional audio system using active crossover network.

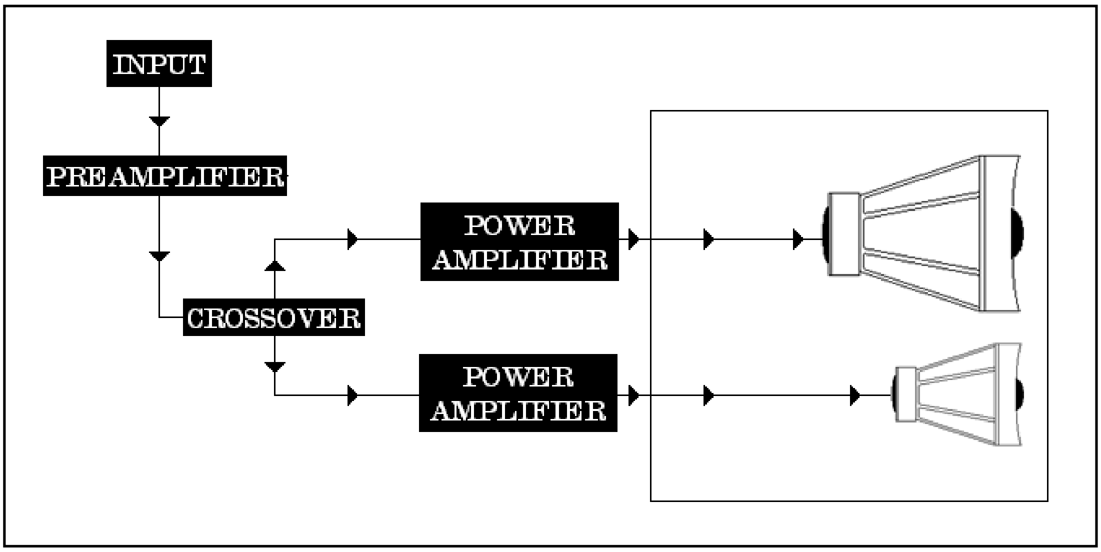

Figure 2.9 : Conventional audio system using active crossover network

2.2.3 Filters

To ensure both types of network can split an input to several frequency ranges, 2 or more filters will be used. A multi-way system may have 2 or 3 filter in it. It will use the combination of inductor, capacitor and resistor to form an electronic filter network. Therefore, the networks are composed of elements which allow passing, or preventing from passing certain bands of frequencies.[1]

The range of frequency that can pass by can be set by setting the value of F_C. F_C is the cross frequency that will shows the limitation of frequency that the filter will allow to pass by. Generally, the equation to obtain F_C is:

$$\boxed{F_C = \frac{1}{2\pi\sqrt{2}RC}}$$

Where R = Resistor value and C = Capacitor value

This equation can be used to determine the F_C value for both low-pass and high-pass filter.

2.2.3.1 Low-pass Filter

Low-pass filter will allow low frequencies to pass and eliminate high frequencies. Basic low-pass filter circuit is shown in Figure 2.10 below while Figure 2.11 shows the position of F_C for low-pass filter.

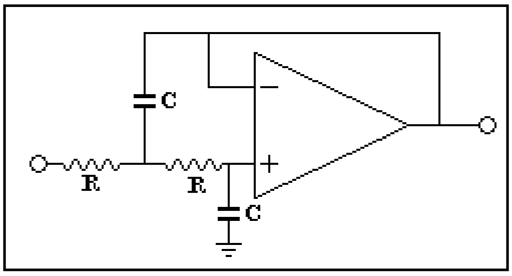

Figure 2.10 : Basic low-pass filter circuit

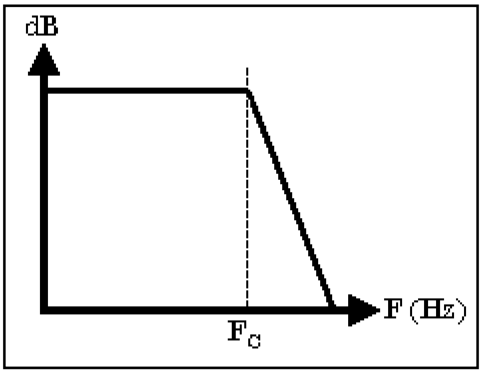

Figure 2.11 : F_C position for low-pass filter

2.2.3.2 High-Pass Filter

A high-pass filter will allow high frequencies to pass and at the same time eliminate the low frequencies. Basic high-pass filter circuit is shown in Figure 2.12 and F_C position for high-pass filter is shown in Figure 2.13.

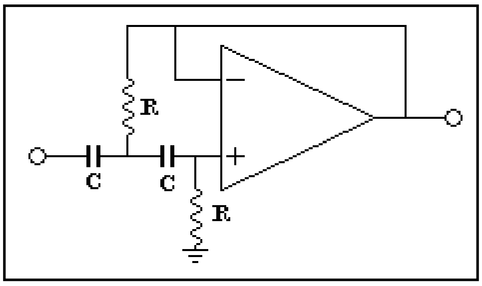

Figure 2.12 : Basic high-pass filter circuit

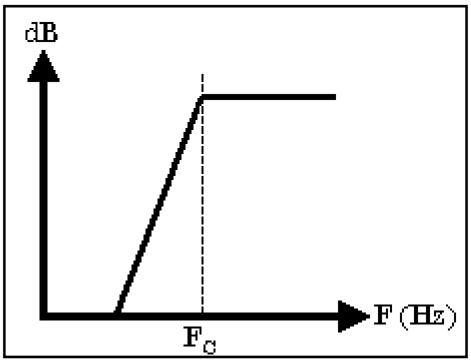

Figure 2.13 : F_C position for high-pass filter

2.2.3.3 Bandpass Filter

Basically, it is the combination of low-pass and high-pass filter. Only frequencies in frequency band are allowed to pass. Figure 2.14 below shows basic bandpass filter circuit while Figure 2.15 shows F_C position for bandpass filter.

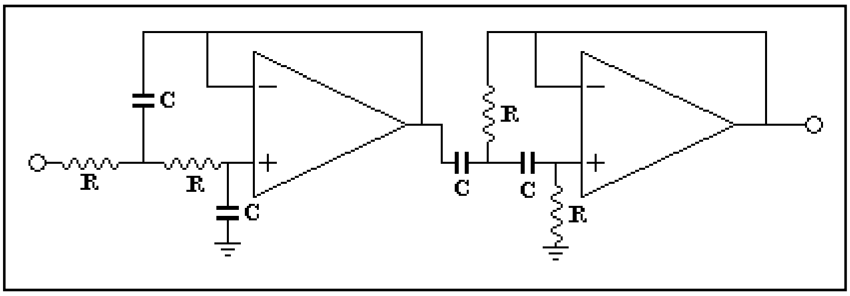

Figure 2.14 : Basic bandpass filter

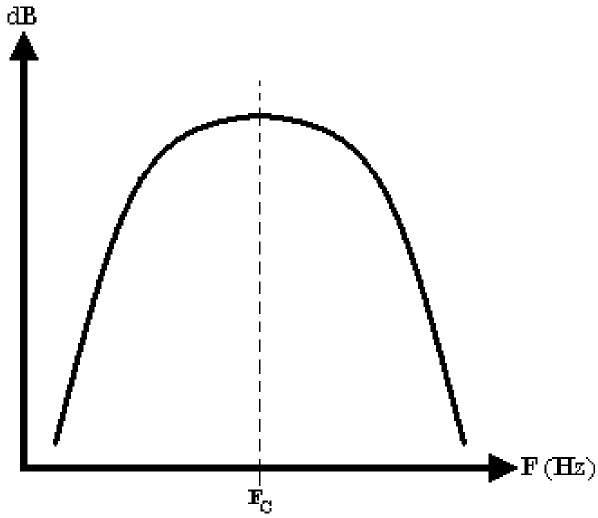

Figure 2.15 : F_C position for bandpass filter

2.3 Power Amplifier

Generally, an amplifier is any device that uses a small amount of energy and converts it to a larger amount of energy. Specifically, electronic amplifier is a device that used to increase the power of a signal. It does this by taking power from a power supply and controlling the output to match the input signal shape but with a larger amplitude. In this sense, an amplifier may be considered as modulating the output of the power supply.[4]

2.3.1 Amplifier Classes [4]

Amplifier can be classed into 4 main classes. These classes indicate an amplifier's efficiency and sound quality. Typically, the more efficient the amplifier, the poorer the sound quality will be. All 4 classes are:

2.3.1.1 Class A

A Class A amplifier has a clean output, but poor efficiency. Class A amplifiers are typically more linear and less complex than other types. Most Class A amplifier operates at about 20% to 30% efficiency. This type of amplifier is most commonly used in small-signal stages or for low-power applications.

2.3.1.2 Class B

This class of amplifier has twice the efficiency of Class A amplifier. Even though Class B amplifier might seem ideal, they cause audio distortion and are rarely used in car audio.

2.3.1.3 Class AB

Class AB is probably the most common 12V amplifier today. This amplifier provides good sound quality while maintaining the efficiency. Most car audio amplifier use a Class AB designs.

2.3.1.4 Class D

The Class D amplifier is very efficiency, but requires complex output filter, which will increase the cost. The efficiency is due to the power supply following the input signal and adjusting itself accordingly. The Class D amplifier switches on and off hundreds of thousand times per second. Class D is not a designation for Digital.

2.3.2 Amplifier Characteristics

Most amplifiers can be characterized by a number of parameters.

2.3.2.1 Output Power

This parameter shows the amount of power that the amplifier can deliver. There are 2 factors that limited the output power. First the limitation from the switching power supply and second the limitation of the audio amplifier itself. The output power of an amplifier can be calculated using equation

$$\mathbf{P = V^2/8R}$$

where P = power (in watt), V = peak to peak voltage, R = impedance of the speaker.

2.3.2.2 Gain

The gain is the ratio of output power to input power. It is usually measured in decibels (dB).

2.3.2.3 Noise

This is a measure of how much noise is introduced in the amplification process. Noise is an undesirable but inevitable product of the electronic devices and components. Noise is measured either in decibels or the peak output voltage produced by the amplifier when no signal is applied.[4]

2.3.2.4 Total Harmonic Distortion (THD)

The THD is the ratio of the sum of the powers of all harmonic frequencies above the fundamental frequency to the power of the fundamental frequency. It is usually measured in decibel.

2.3.2.5 Efficiency

Efficiency is a measurement of how much the input power is usefully applied to the amplifier's output. The efficiency of the amplifier limits the amount of total power output that is usefully available. Note that more efficient amplifiers run much cooler, and often do not need any fans even in multi-kilowatt designs. The efficiency of the amplifiers is depends on what class the amplifiers are.

2.3.2.6 Bandwidth

The bandwidth of an amplifier is usually defined as the difference between the lower and upper half power points. This is therefore also known as the −3 dB BW. Bandwidths for other response tolerances are sometimes quoted (−1 dB, −6 dB etc.).[4]

2.4 Enclosure

All speakers are designed to be mounted inside an enclosure or cabinet. Failure of providing an enclosure for the speaker will cause the system to operate inefficient. The shape, size and construction of a speaker enclosure affect the overall performance of the speaker(s). The enclosure directs the sound waves, determines the frequency response of the system and controls the sound intensity.

A proper box can help the speaker to deliver 100 times greater sound intensity at low frequency. The air from the high-pressure side of the cone mixes with the air from the low-pressure side, which will cause sound cancellations. At high frequency, it does not make much difference because only a little mixing happens but for frequency wavelength that is longer than the diameter of the speaker, the wave curves back to the cone and out-of-phase waves mix. The main reason for building the speaker enclosure is to avoid unwanted mixing of the waves.

There are only a few rules to be followed when designing the speaker enclosure. The diameter of the subwoofer determines the internal volume of the speaker cabinet. Table 2.1 shows the list of the volumes required for different subwoofer diameters.

Table 2.1 : Volume required for different subwoofer diameters

Subwoofer Diameter (inches)	**Internal Cabinet Volume (cubic inches)**
Four	450 – 675
Six	600 – 1000
Eight	1500 – 2500
Ten	2500 – 5000
Twelve	5000 – 10000
Fifteen	8000 – 15000

There are few types of enclosure that will be discussed in the following part.

2.4.1 Sealed

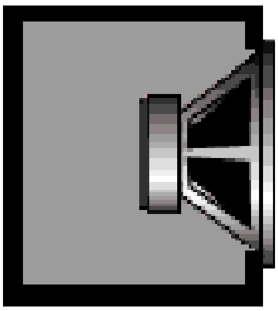

Figure 2.16 : Sealed speaker system

This is probably the most popular type of enclosure in commercial use today. Sealed box; as shown in Figure 2.16, have the tightest, cleanest sound and they are the easiest to make. This type is preferred due to the simplicity of its design which promotes a smooth frequency response, excellent cone control which translates into accurate sound reproduction.[3] A sealed box requires a lot of audio power. Other downside of this design is that it does this all at the expense of low efficiency. The volume of air within a sealed enclosure is less than the V_{as} of the driver; the air trapped in the enclosure helps control the movement of the cone somewhat like a shock absorber controlling the springs on a car.

2.4.2 Ported

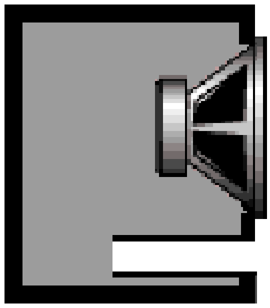

Figure 2.17 : Ported speaker system

The ported speaker system as shown in Figure 2.17 above uses a port to reinforce the low frequencies. The port is a partial vent for the compressed and decompressed air. The port increases the effective enclosure volume; therefore, the ported system is efficient and can be driven properly by a “modest” amount of audio power. The ported box can produce lower frequencies than sealed system. A high compliance subwoofer (folded paper suspension) must be used in a ported speaker system.[5]

2.4.3 Bandpass

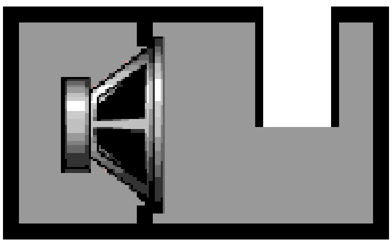

Figure 2.18 : Bandpass speaker system

Bandpass enclosure is based on ported enclosure, except that the direct sound from the speaker is blocked off using a secondary sealed chamber. This means that the sound that will be heard is from the tuned output from the port. Unlike the other enclosure types, bandpass boxes only pass sound over a narrow frequency band. An example of this system is shown in Figure 2.18 above.