ACTIVE LOUDSPEAKER SYSTEM

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BORANG PENGESAHAN STATUS TESIS♦

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"I hereby acknowledge that the scope and quality of this thesis is qualified for the award of the Bachelor of Electrical Engineering (Electronics)"

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A thesis submitted in fulfillment of the
requirements for the award of the
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To my beloved father, mother and sisters

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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.

ABSTRAK

Pembesar suara adalah salah satu daripada perkakasan elektronik yang berfungsi dengan cara menukar tenaga elektrik kepada tenaga akustik. Walaupun semua pembesar suara kelihatan sama, tetapi sebenarnya tidak. Setiap sistem mempunyai karakter, jenis-jenis dan fungsi tersendiri. Untuk sistem pembesar suara aktif saya ini, ia mempunyai litar sendiri iaitu penyaring frekuensi dan penguatkuasa bunyi terbina dalam. Isyarat bunyi dapat dihasilkan hanya dengan menyambung satu sumber kepada sistem tersebut. Litar penyaring frekuensi dalam sistem ini diletakkan di antara input dan penguatkuasa bunyi. Ia mempunyai dua penyaring; penyaring frekuensi rendah dan frekuensi tinggi. Penyaring ini akan memisahkan isyarat tersebut kepada 2 julat frekuensi; julat frekuensi rendah (0 hingga 3100 Hz) dan julat frekuensi tinggi (3100 Hz – 20000 Hz). Selepas itu, penguatkuasa bunyi akan menaikkan amplitud isyarat kepada kira-kira 20dB hingga 28dB. Kedua-dua isyarat bunyi kemudiannya akan disalurkan kepada dua *speaker*; *tweeter* dan *subwoofer*. *Tweeter* akan menghasilkan bunyi berfrekuensi tinggi manakala bunyi berfrekuensi rendah dihasilkan oleh *subwoofer*. Sistem ini juga mempunyai litar perlindungan *speaker* sendiri untuk mengelakkan *speaker* daripada rosak. Akhir sekali, kotak dibina sebagai platform kepada sistem ini dan sebagai tempat untuk meletakkan litar bersama-sama dengan *speaker*. Beberapa garis panduan seperti jenis kotak dan ukuran yang sepatutnya dipatuhi untuk memastikan produk akhir sama seperti jangkaan awal.

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LIST OF ABBREVIATIONS

AC	- Alternate current
BW	- Bandwidth
DC	- Direct current
F_C	- Cut-off frequency
IC	- Integrated circuit
OP-AMP	- Operational amplifier
PC	- Personal computer
POWER-AMP	- Power amplifier
PRE-AMP	- Preamplifier
RMS	- Root mean square
THD	- Total harmonic distortion

LIST OF SYMBOLS

cm	- Centimeter
dB	- Decibles
Hz	- Hertz
k	- kilo (10^3)
n	- nano (10^{-9})
W	- Watts

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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.

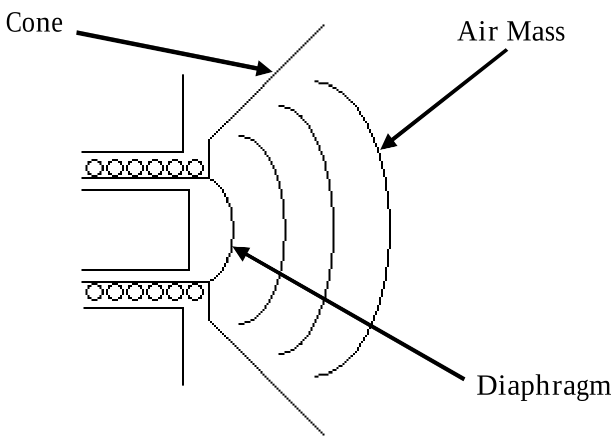

Figure 2.1 : Direct radiator type

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.

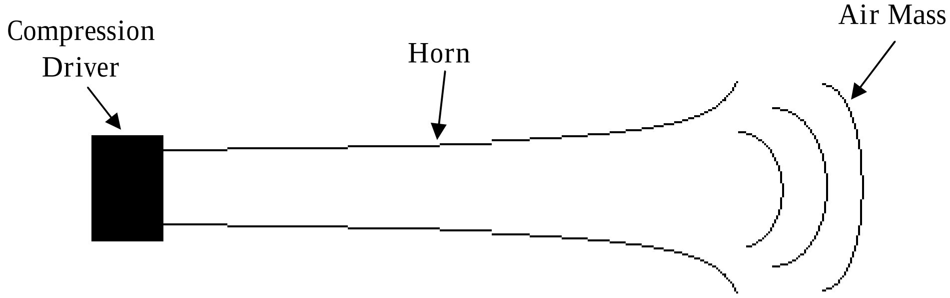

Figure 2.2 : Indirect radiator type

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.

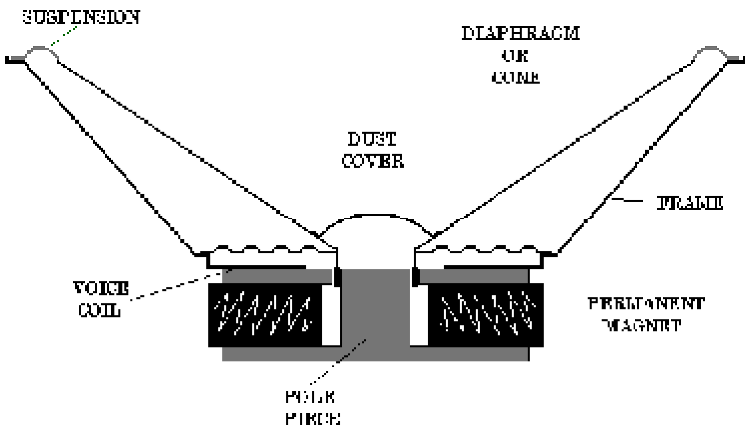

Figure 2.3 : Dynamic loudspeaker

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.

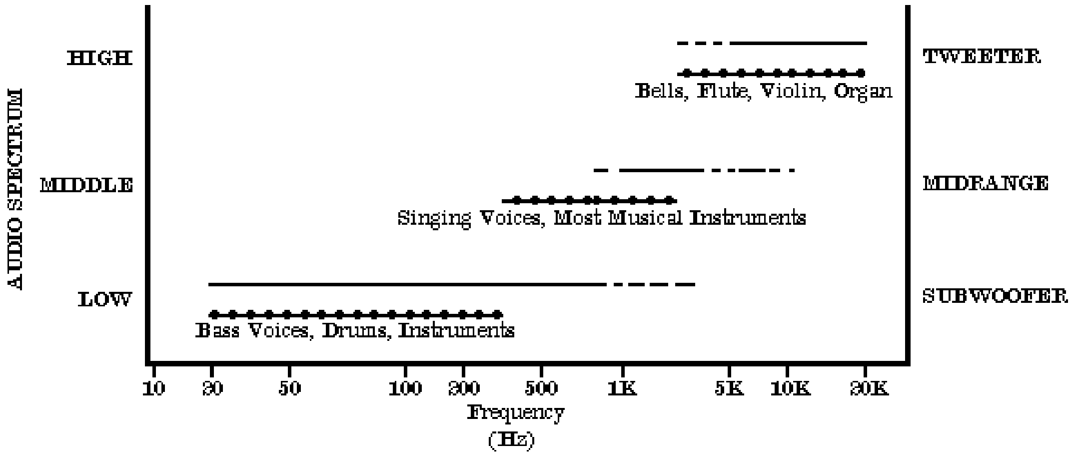

Figure 2.4 : Range of Sound

- 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.

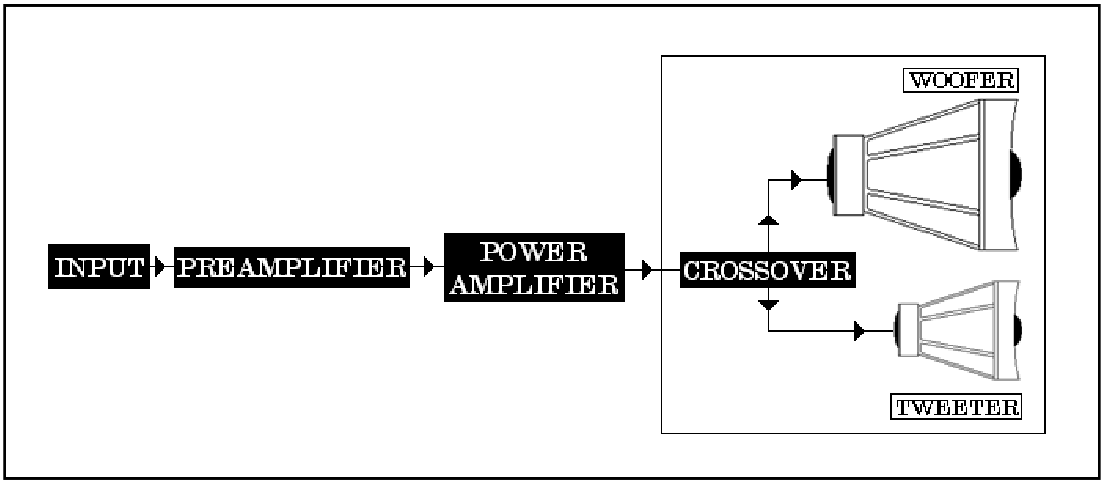

Figure 2.8 : Conventional audio system using passive crossover network

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.

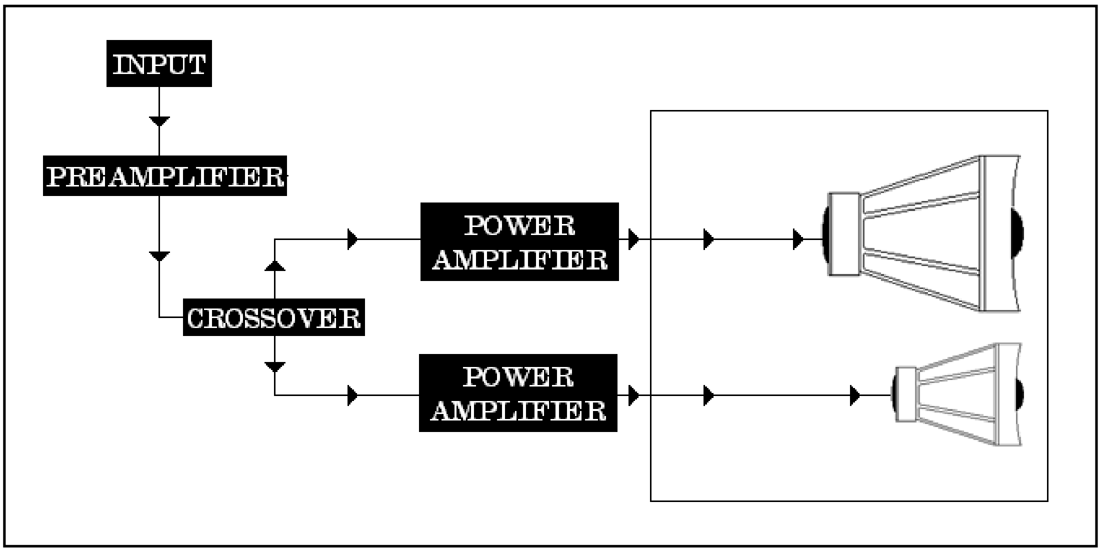

Figure 2.9 : Conventional audio system using active crossover network

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.

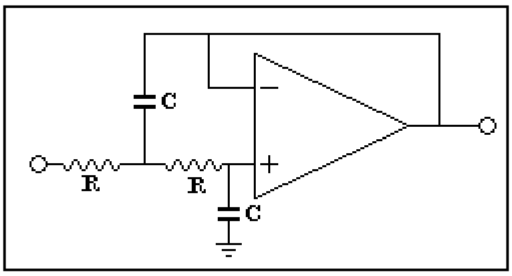

Figure 2.10 : Basic low-pass filter circuit

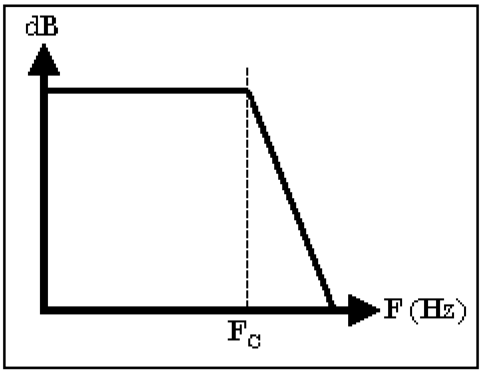

Figure 2.11 : F_C position for low-pass filter

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.

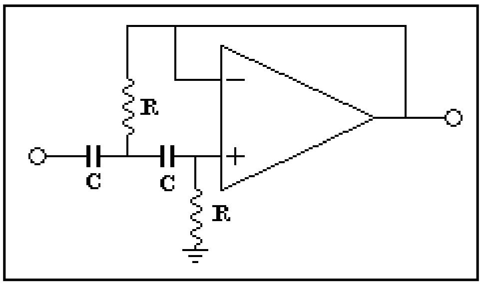

Figure 2.12 : Basic high-pass filter circuit

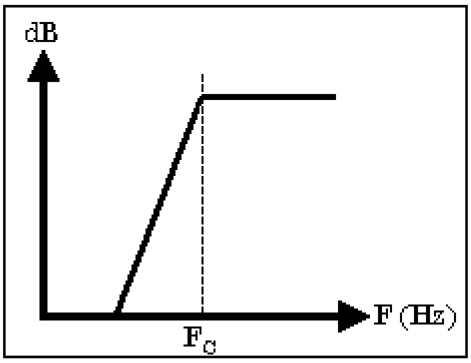

Figure 2.13 : F_C position for high-pass filter

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.

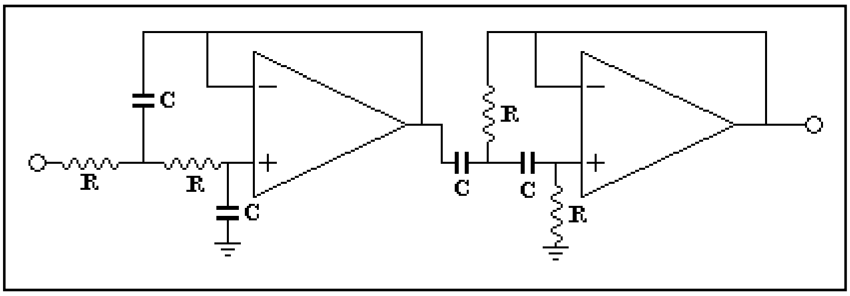

Figure 2.14 : Basic bandpass filter

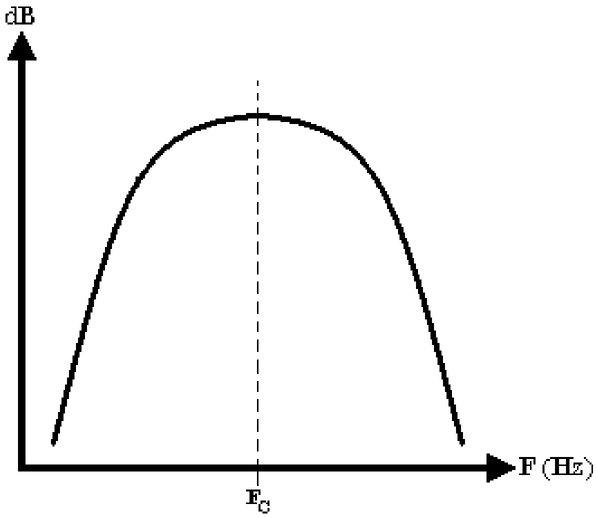

Figure 2.15 : F_C position for bandpass filter

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

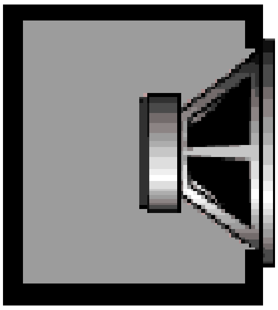

Figure 2.16 : Sealed speaker system

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

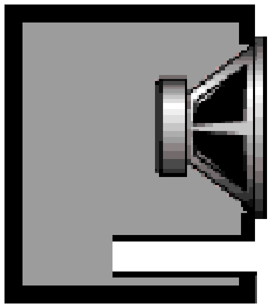

Figure 2.17 : Ported speaker system

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

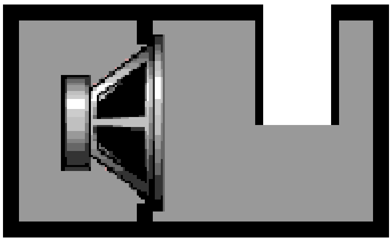

Figure 2.18 : Bandpass speaker system

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.

Bandpass system is more difficult to design and build than either sealed or ported system. They are very sensitive to speaker parameters, box volumes of the sealed and ported sub-enclosures, also port tuning. Bandpass system produces deep bass that is compared to that of equally sized ported system. It does need a big enclosure.[5]

2.4.4 Transmission Line

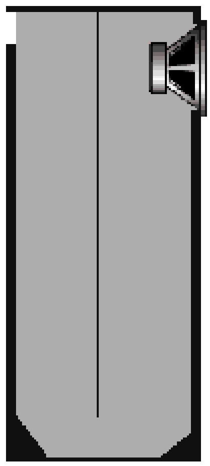

Figure 2.19 : Transmission line speaker system

Transmission line makes constructive use of the sound from the back of the speaker. This design will makes it possible to achieve deeper bass than with a sealed box. Transmission line enclosure delay the sound from the back of the speaker through a labyrinth structure. The delay is chosen to provide constructive reinforcement of bass at the proper frequency.

Transmission line enclosures utilize elaborate and expensive construction to create the internal labyrinth structure. They do not offer any advantages over properly designed ported enclosures and often have coloration problems in the mid-bass region.[5] An example of transmission line speaker system is shown in Figure 2.19 above.

CHAPTER 3

METHODOLOGY

3.1 Project Overview

This project only involves the building of hardware. The hardware can be divided into three different stages which are the crossover circuit, power amplifier circuit and lastly the speaker protection & delay circuit. When all the circuit is functioning and have the desired output, only then I will integrate all the stages together and then begin to construct the enclosure for both the circuit and the speaker.

Furthermore, there is a software used in this project called the “TrueRTA”. The software is used just to get the final output and is not involve on the hardware building process.

3.2 Circuit Explanation

3.2.1 Crossover Stage

3.2.1.1 Calculation

As mentioned in the chapter before, the crossover role is to split the frequency into several frequency ranges. The filter in the crossover circuit will allow some frequencies while eliminate the unwanted frequencies. F_C is the cross frequency that will shows the limitation of frequency that the filter will allow to pass by.

This project only involve of building low-pass and high-pass filter. F_C for both filters is determined as follow:

$$f_c = \frac{1}{2\pi\sqrt{2}RC}$$

$$= \frac{1}{2\pi\sqrt{2}(11k)(3.3n)}$$

$$= 3100\ Hz$$

This value of F_C is selected because it is near the cross frequency of many speaker available in the market.

3.2.1.2 Crossover Circuit

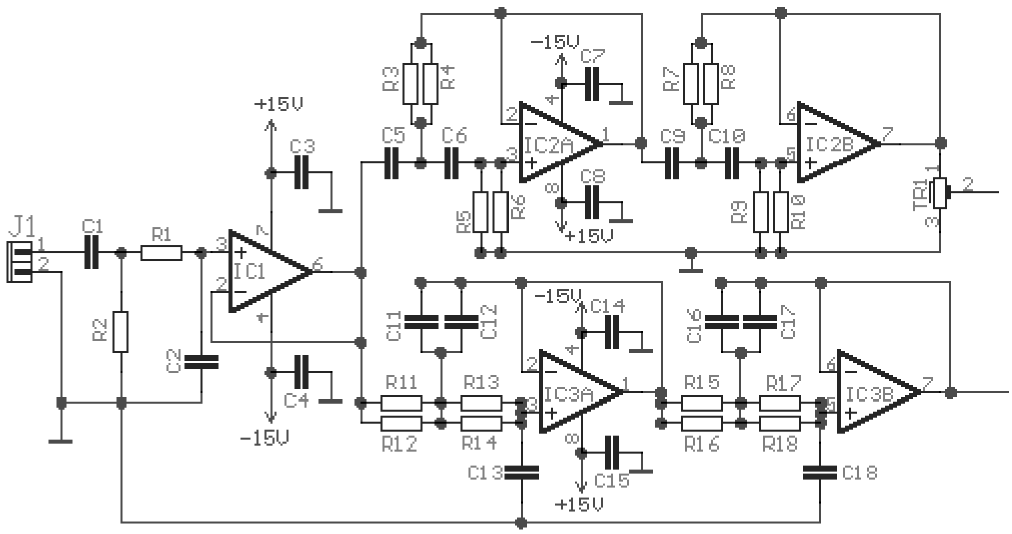

Figure 3.1 : Overall crossover circuit

As shown in Figure 3.1 above, this stage consists of 3 parts:

i.) Preamplifier
ii.) High-pass filter
iii.) Low-pass filter

In this circuit, I am using op-amp as the basic component to build the preamplifier, and the filters. Op-amps that involved are TL071 (IC1), TL072 (IC2) and NE5532 (IC3). Such op-amp is used because they are producing low harmonic distortion and low noise which suitable to be used for audio application. They also supplied with 15V positive and negative DC voltage. A 3.5” audio jack input is used as connecter between the sources and the circuit. The input is then soldered to J1.

3.2.1.2.1Peramplifier Circuit

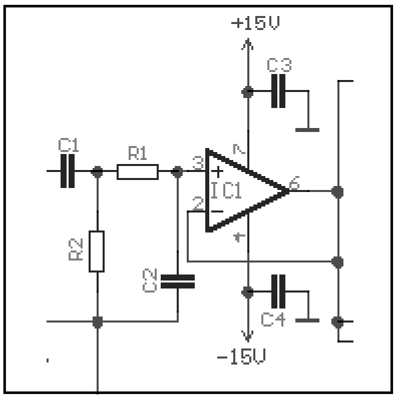

Figure 3.2 : Preamplifier circuit

When the signal is connected through the input jack, it will firstly face the pre-amp circuit as shown in Figure 3.2 above. The pre-amp (IC1) will increase the signal gain a little bit without changes its structure. The increasing of the signal is done to ensure the signal still can be delivered to next stage without facing any loss that may happened when the transferring process from the source to the circuit. After that, the signal will be split into two and each one will face 2 different filters that are high-pass filter and low-pass filter.

3.2.1.2.2 High-pass Filter Circuit

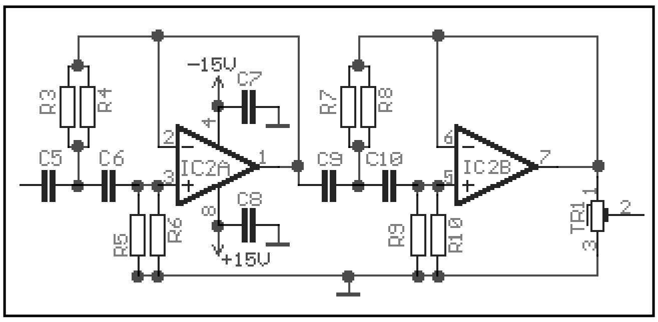

Figure 3.3 : High-pass filter circuit

In high-pass filter, the signal with the frequency from 3100 Hz and above is allowed to pass while other frequencies will be eliminated. The value of 3100 Hz that we used as the limitation is set earlier by placing the suitable capacitor and resistor in the circuit as shown in Figure 3.3 above.

3.2.1.2.3 Low-pass Filter Circuit

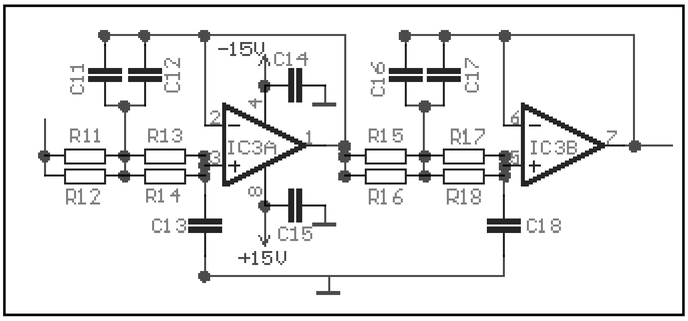

Figure 3.4 : Low-pass filter circuit

While the high-pass filter performing its job, the low-pass filter (placed below the high-pass filter position) will play its role to stop any frequencies from 3100 Hz and above while allowing the frequencies from the range of 0 – 3100 Hz. Both filters will do their job at the same time to ensure the signal still unchanged. The circuit for low-pass filter is shown in Figure 3.4 above.

Need to be mentioned that, this is a 2-way system. It means that only 2 types of filter will be used and there will be just 2 ranges of frequency that will be produce by the crossover stage. Each frequency will be connected to its own separated power amplifier at the next stage.

3.2.2 Power Amplifier Stage

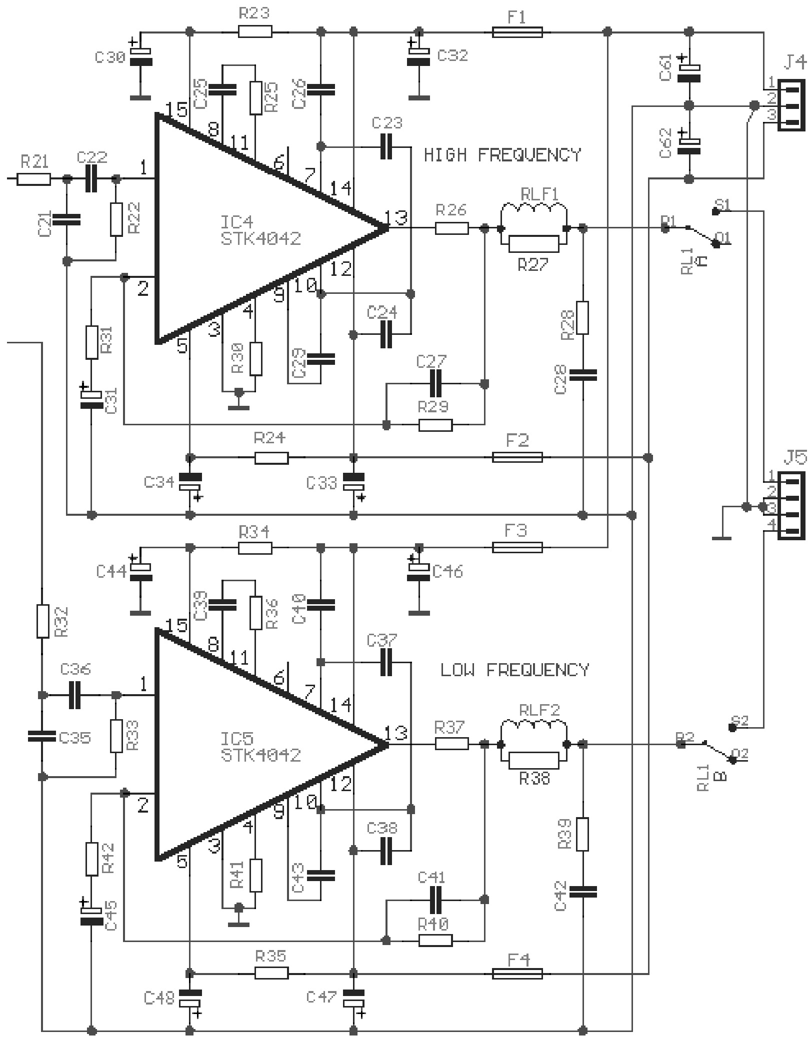

Figure 3.5 : Overall power amplifier circuit

After the signal has been split into 2 different ranges from the previous stage, they will face the power-amp circuit. High-pass filter that producing high-frequency range is connected to IC4 while IC5 will receive low frequency range that come from low-pass filter.

As we can see in the circuit as shown in Figure 3.5, there are 2 power-amps used for each range. Both circuits are identical and they also have the same function. The power-amp will takes the signal and makes it larger without adding or taking away the original signal. Its function is like the preamplifier but the main difference is that the power-amp will increase the signal larger than what the pre-amp does.

The power amplifier IC used in this stage is STK4042II. This is an audio amplifier that will provide a minimum output power of 80 Watts. Although the output power is quite high but it is suitable for almost tweeter and woofer sold in the market. It has THD of 0.4% and has a load resistance of 8 ohm.

Both power-amps are supplied with 30V positive and negative DC voltage. Noted that before the power supply is been connected to the STK4042II, there are 4 fuses placed between the power-amp and the power supply. The fuses are used to ensure that the STK4042II is safe from over-current that may occur during the process. The used of fuses is important because the process of amplify involves the circuit to play with high current and over-current can happen anytime that will cause the IC used to be broken

3.2.2.1 Power-amp for High Frequency Part

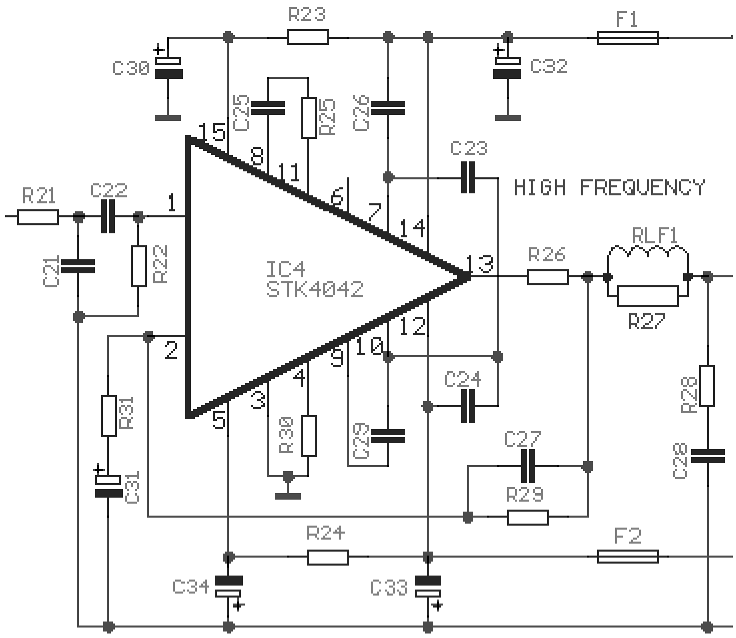

Figure 3.6: Power-amp for high frequency part circuit

Figure 3.6 above shows power amplifier circuit for high frequency part. IC4 that connected to the output of high-pass filter will increase the high frequency signal gain. The increasing is about 20-28 dB. After the signal has been amplified, the signal will then be delivered to the tweeter. The tweeter is then will convert the signal into sound wave. The tweeter is used because it is capable of producing high frequency sound.

3.2.2.2 Power-amp for Low Frequency Part

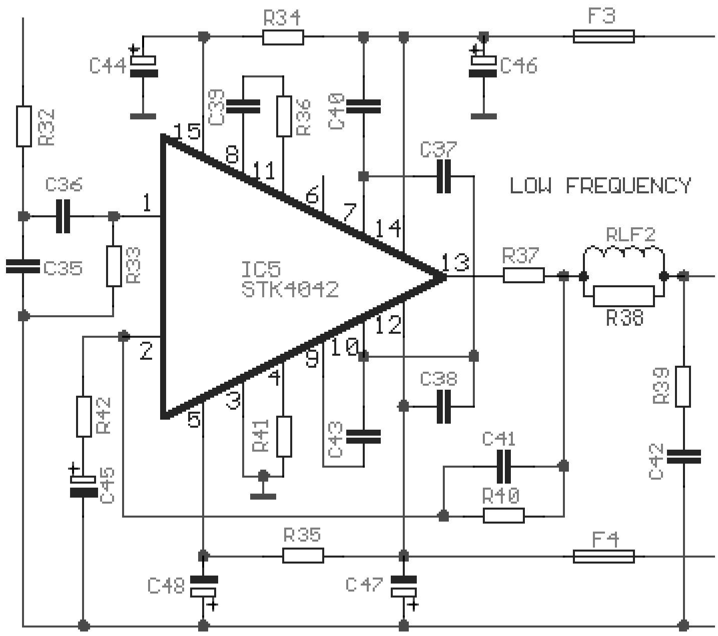

Figure 3.7 : Power-amp for low frequency part circuit

As shown in Figure 3.7 above, we can see that IC5 also has the same circuit as IC4. The difference is that it is connected to the output of low-pass filter that is producing low frequency range. After IC5 has increase the signal gain in about 20-28 dB (same as IC4), the signal is then delivered to J5 that use a subwoofer as a converter to convert the electric signal into audio signal.

3.2.3 Speaker Protection and Delay Stage

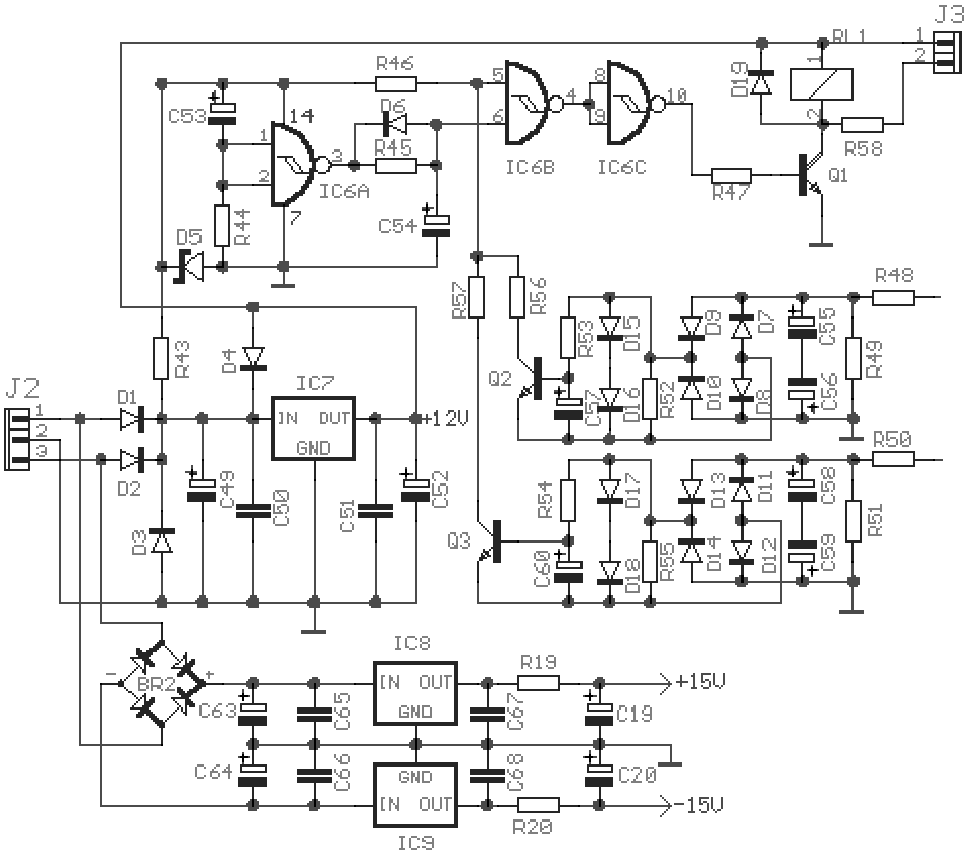

Figure 3.8 : Speaker protection and delay circuit

Actually, this stage was not the part of this project scope. It just provided by the circuit builder along with the other 2 circuits. But, that not meant that the circuit as in Figure 3.8 will not be used. The circuit is providing a protection to the speaker that been used as the last link in the system.

The relay in this stage is connected to the power-amp output and on the other side of relay the speaker is connected. Its operation is very simple as if the relay is "on", the output of the power-amp will be delivered to the speakers. If the relay is "off", no sound will be produced by the speaker as there will be no energy signal to be converted into audio signal.

The stage is functioned by making 5 second delay after power is supplied to the circuit. The delay is needed because this will prevent the speaker from producing the sound of capacitor been charged. Need to be mentioned that the capacitor been used is a large capacitor; two 15000µ 71V. A period of time is needed to charge and discharge the capacitors.

After the power is turned off, the relay will cut the output line connected to the speakers. It is important to make sure the noise of discharging capacitor is never produced by the speaker as it will blow down the speaker. Also, it will protect the speaker from the DC voltage that may produced by the power amplifier. Overall, the stage main function is to provide the protection to the speakers used to ensure they will not blow down easily.

In addition, the stage also includes the production of positive and negative 15V that will be used to supply the op-amp IC in the crossover stage.

3.3 Enclosure Construction

This is the last part of hardware building. The main purpose of the enclosure 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. The enclosure should be rigidity, or resistance to vibration. The material used to construct the enclosure must be rigid so that they will not be vibrated by the sound pressure within the enclosure. If they are vibrated, acoustic work is done on them and acoustic power is wasted. Therefore, the right material should be chosen to fulfill the requirement.

Actually, the enclosure can be built using various forms. The basic requirement to be fulfilled as can as possible is their strength, workability and appearance. The density, elasticity, and internal losses are important in that they given the thickness of panels of any given size in order to obtain the desired rigidity.[2] Table 3.1 below shows the density of woods and other materials as well as their workability.

Table 3.1 : Mechanical properties of various loudspeaker cabinet

Material	Density (lb/ft^3)	Workability
Concrete Boxwood	70	Moderate
Oak	50	Moderate
Plywood (low density)	40	Satisfactory
Plywood (high density)	80	Satisfactory
Blockboard (without voids)	40 - 50	Satisfactory
Chipboard (high density)	50 - 60	Precautions needed

For this project, I am using plywood to construct the enclosure because it has satisfactory workability with the lowest price. To avoid the air leaks and give the strength to the enclosure, all joints are screwed firmly. Also, to ensure the enclosure is maintaining good low frequency response, the internal cabinet height, width, and depth is following to the ratio of 8:5:3.

The exact measurement used in the construction is shown as in Figure 3.4 below. The ratio is not exactly same as the needed ratio but still not too far away (8:4.8:3.2). Need to be mention that the depth for the speaker cabinet part is just 20 cm and the remaining 15 cm is spared to place the circuit.

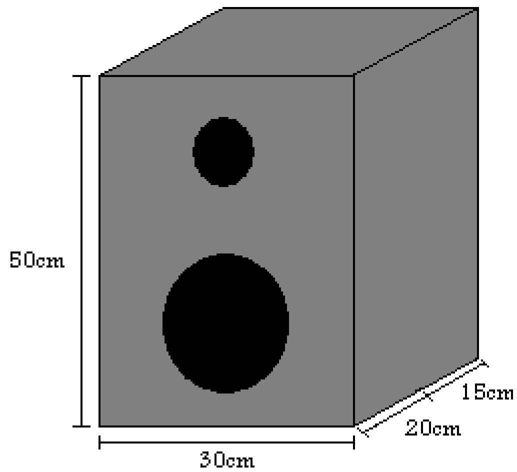

Figure 3.9 : Exact measurement of the enclosure

Overalls, the steps involve on constructing the enclosure are:

i.) Designing the shape and measurement
ii.) Sawing the wood
iii.) Joining and screwing the separated part
iv.) Screwing the speakers to their appropriate holes
v.) Placing and screwing the circuit on its place
vi.) Painting the enclosure
vii.) Testing the system

3.3.1 Designing the Shape and Measurement

The designing process is done by following several guidelines. The measurement is calculated so that the volume of the enclosure that wants to be built is still in its appropriate range. Also, the internal cabinet height, width, and depth are following to the ratio of 8:5:3. Actually, the enclosure has 2 parts in it. The front part is used as the speakers box while the back part is used to place the circuit.

3.3.2 Sawing the Wood

This process involve on using the electric wood chainsaw. It is easy to be used as it didn't require much human power. But caused by the plywood's thickness, the measurement from earlier process need to be adjusted a little bit to ensure the product still same as the designed one.

3.3.3 Joining and Screwing the Separated Parts

In this process, the separated woods are joined together using a tough wood as a based. The screws are used because they provide bond between the plywood compared to the nails.

3.3.4 Screwing the Speakers to Their Appropriate Holes

The speakers are placed into its holes that already been drilled. For the tweeter, a 3.5 inch holes while for the subwoofer an 8 inch holes is drilled. The speaker is then screwed tightly to ensure there are no air leaks.

3.3.5 Placing and Screwing the Circuit on Its Place

The circuit is screwed and placed on the back of the enclosure. Such process is done to follow the meaning of active loudspeakers itself; an audio sound that has its own circuit and built-in power amplifier.

3.3.6 Painting the Enclosure

The process is done to make sure the enclosure more attractive. The color that been used is black.

3.3.7 Testing the System

This final process is done to ensure our system can work properly. This process is made by connecting some audio source to the system. If the system works properly, then the system has meets its purpose.

3.4 Software Explanation

As mentioned earlier, there is only one software used in this project that is TrueRTA Version 3.3 program. The software was not involve directly on the process of building the circuit but used as a tool to get the final result. Figure 3.10 below shows the TrueRTA program.

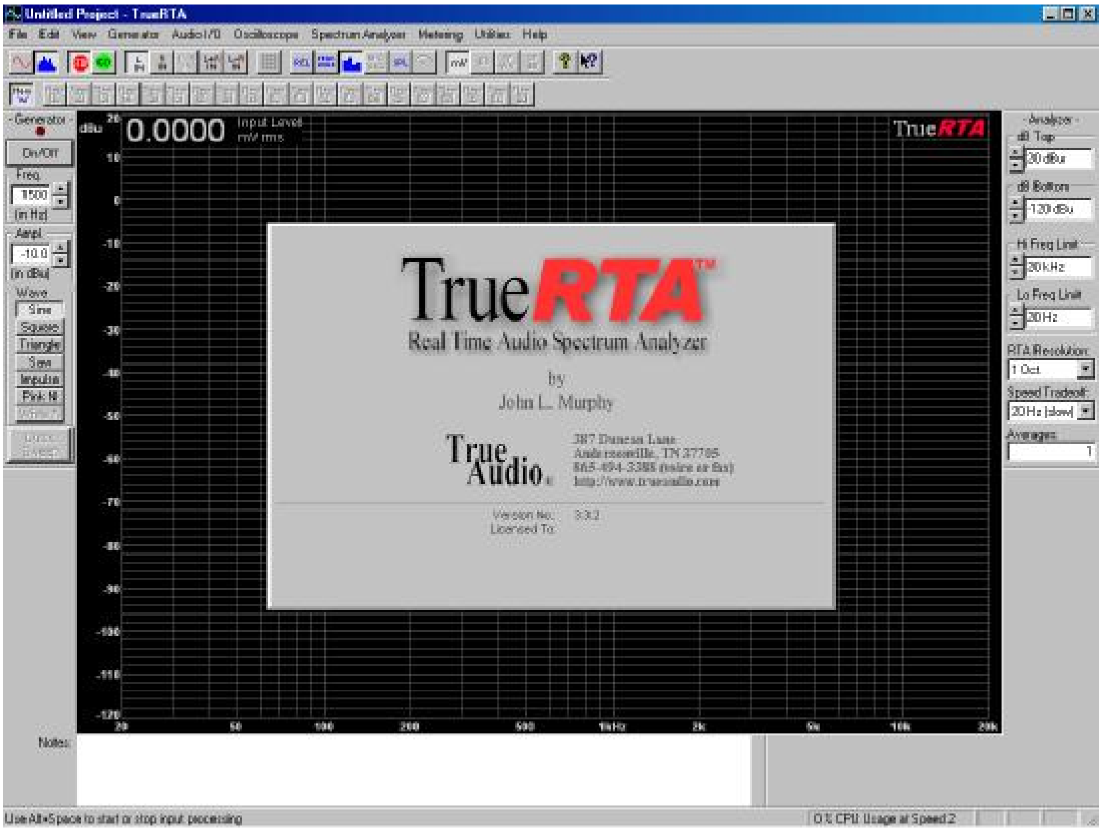

Figure 3.10 : The TrueRTA program [7]

3.4.1 Introduction of TrueRTA

TrueRTA is a collection of real-time software-based instruments for testing and evaluating audio systems using a PC with basic sound input and output capability. The instruments found in TrueRTA include a low distortion signal generator, a digital level meter, a crest factor meter, a dual trace oscilloscope and a high-resolution real time analyzer.[7]

For this project purpose, the program just been used as a signal generator and as a spectrum analyzer. A computer output will be connected to the audio system and a microphone used to catch the audio frequency also connected to the computer. The testing process block diagram is shown in Figure 3.11 below.

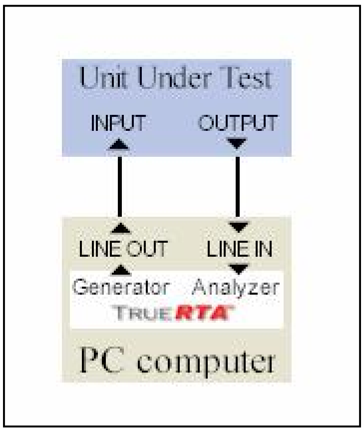

Figure 3.11 : Block diagram on how the testing process is done [7]

3.4.2 As the Generator [7]

The TrueRTA signal generator produces an ultra low-distortion sine wave variable from 5.0 Hz to 48 kHz. The output level is specified in dB. In addition to the sine wave, the generator can also generate square, triangle, saw tooth and impulse waveforms as well as pink noise and white noise. The duty cycle of the square wave is adjustable.

During the testing, we are producing sin wave with various frequencies and permanent output level. The wave is then transmitted to the audio system that already connected to the computer output.

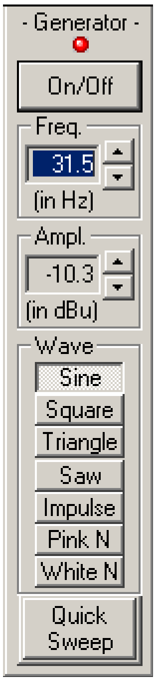

Figure 3.12 : Signal generator dialog bar

At the top of the dialog bar in Figure 3.12, there is On/Off button that starts and stops the generator output. Below the On/Off switch is the Frequency field where the frequency for the sine wave generator can be entered. The up/down buttons allow you to step the sine wave frequency up or down in steps of various sizes. The keyboard up/down cursor also can be used to control arrows to step the up/down buttons.

Below the Frequency field is the Amplitude field where the signal level (in dB) can be set. The up/down buttons allow the amplitude to be stepped up or down in steps of various sizes. The push buttons below the Amplitude field are used to select the type of noises hat will be used either it is sine, square, triangle, saw tooth, impulse, pink noise or white noise.

3.4.3 As the Spectrum Analyzer [7]

TrueRTA also can be used as a spectrum analyzer. The spectrum analyzer front panel is shown in Figure 3.13 below. As the spectrum analyzer it will displays the magnitude of the input signal versus the frequency of the signal. The horizontal axis is represented by the frequency response while the magnitude (in dB) is displayed on the vertical axis. The input to the Spectrum Analyzer is selected at the Toolbar. The input channel can be either left input channel, right input channel, the sum L+R or the difference L-R.

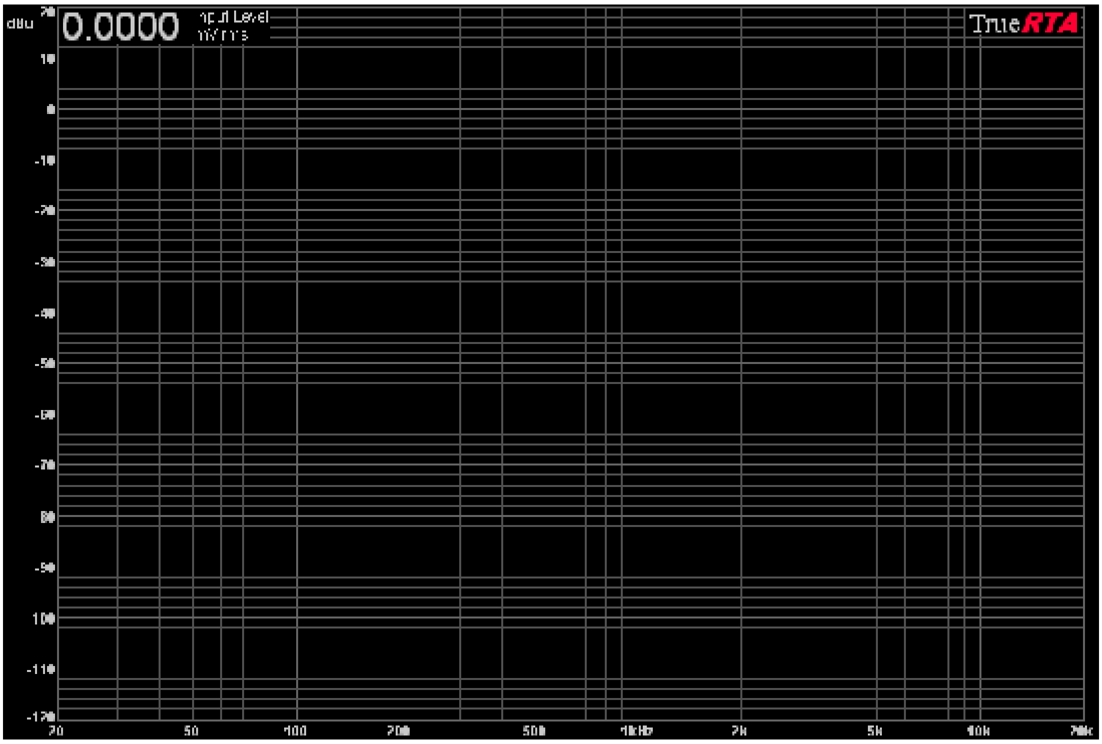

Figure 3.13 : Spectrum analyzer front panel

Figure 3.14 : Spectrum analyzer dialog bar

At the top of the spectrum analyzer dialog bar as shown in Figure 3.14 above, there are two edit fields for selecting the upper and lower dB limits of the display. The up/down buttons are used to set the dB in 10 dB increment. The upper limit can be set between +20 dB and –150 dB while the lower limit ranges from +10 dB down to –160 dB. Below the dB limit buttons are two edit fields which can be used to select the lower limit (from 10 Hz up to 20 kHz) and the upper limit (from 50 kHz down to 50 Hz). The keyboard cursor up/down arrows can also be used to control the edit field up/down buttons.

At the bottom of the dialog bar are three fields for setting the resolution, the speed tradeoff and the number of averages. The resolution field has a popup list with fractional octave resolution selections of 1 octave, 1/3, 1/6, 1/12 and 1/24th octave. The Speed Tradeoff list allows the selection of 20 Hz (slow but precise), 40 Hz (medium speed) or 80 Hz (fastest) as tradeoffs in buffer and FFT sizing.

The Averages field allows you to enter any number for the number of averages to use for the display data. Number 1 represents the fastest updates. Higher number used to reduce the display activity and see a time average of the spectrum.

CHAPTER 4

RESULT AND ANALYSIS

4.1 Introduction and Method

This chapter will show the result and analysis of the project. The result is obtained after the overall process of the system construction is done. The result is obtained using two different methods. The methods are:

Using TrueRTA program
Using function generator and oscilloscope

When the result is obtained using TrueRTA program, there are not changing is made in the circuit. It's just the place of obtaining the result that differs from each other. As when the result is obtained using function generator and oscilloscope, the connection to the speakers has been cut off and been replaced by the input of the oscilloscope.

4.2 Obtaining the Result Using TrueRTA

4.2.1 No Input Connected

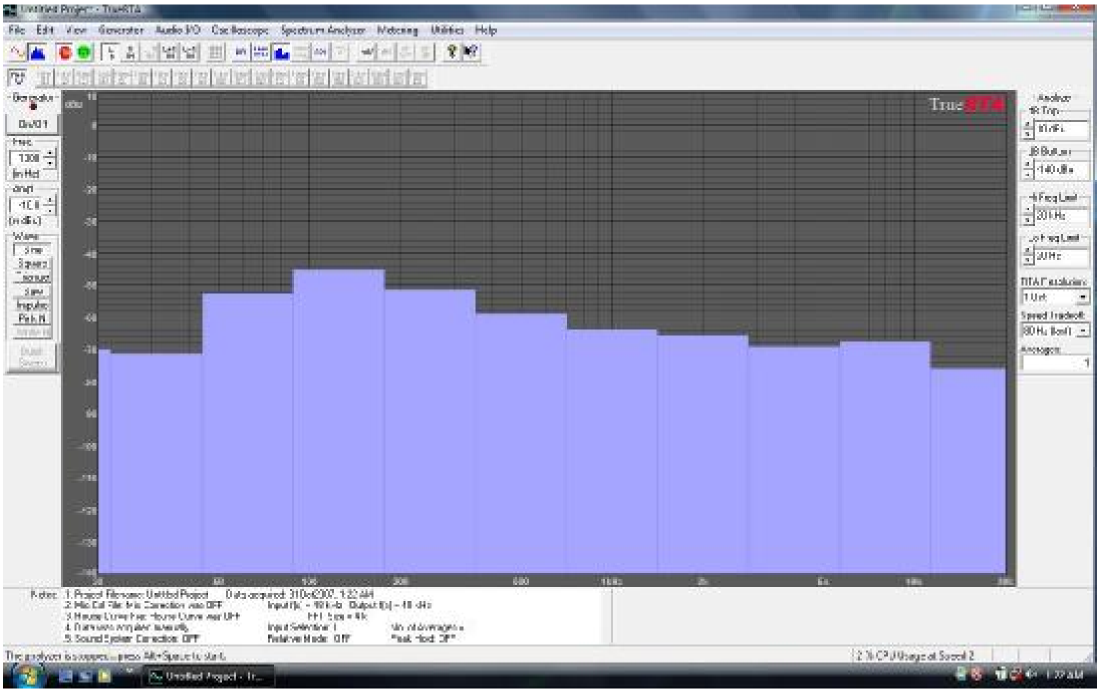

Figure 4.1 : TrueRTA shows the output when no input connected

Figure 4.1 above shows the output obtained from the TrueRTA program when no input is connected. Supposedly, there will be no output but the program still catches something. Such output is obtained maybe because the effect of noises. The noises are come from:

i.) The circuit itself

ii.) The computer that being used as the testing platform

4.2.2 High Frequency Input

4.2.2.1 Output after Crossover Stage

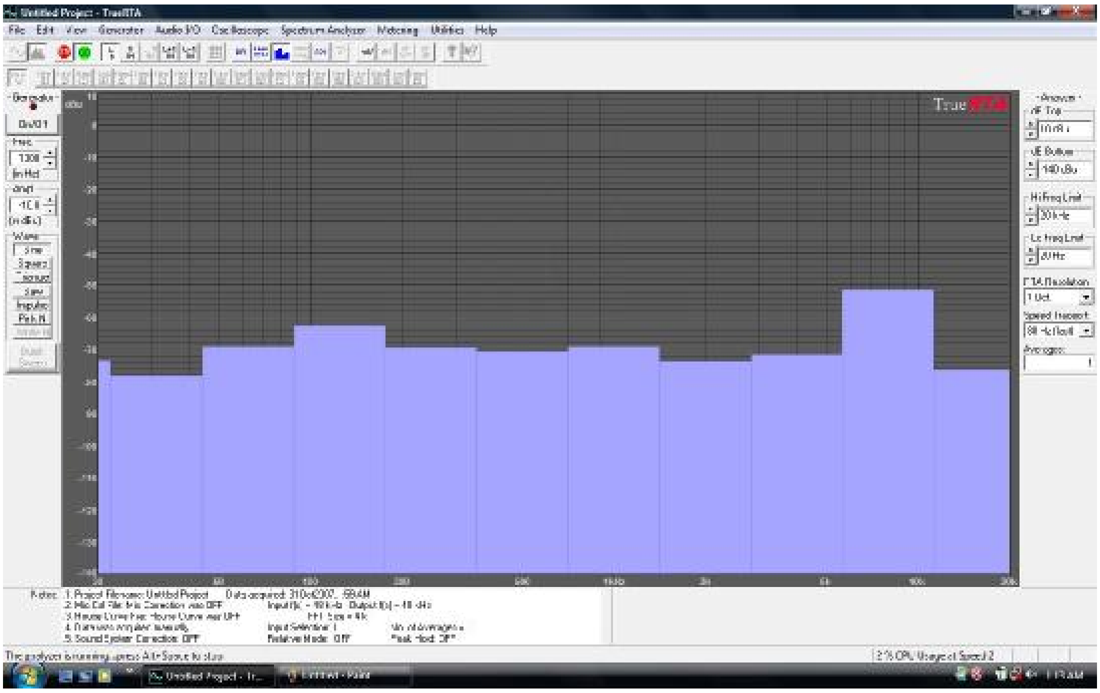

Figure 4.2 : TrueRTA shows the output after crossover stage if a high frequency is set as an input

Figure 4.2 above shows the output after the crossover stage if a high frequency is set as an input. The testing is prepared by giving the system an input frequency of 10kHz and cut the connection to the subwoofer. It is done to know what will be produced by the tweeter only.

As we can see, the TrueRTA catch the frequency that has been set earlier (10kHz). It means that, the high-pass filter has cut off the low frequency and just allows the high frequency.

4.2.2.2 Output after Power Amplifier Stage

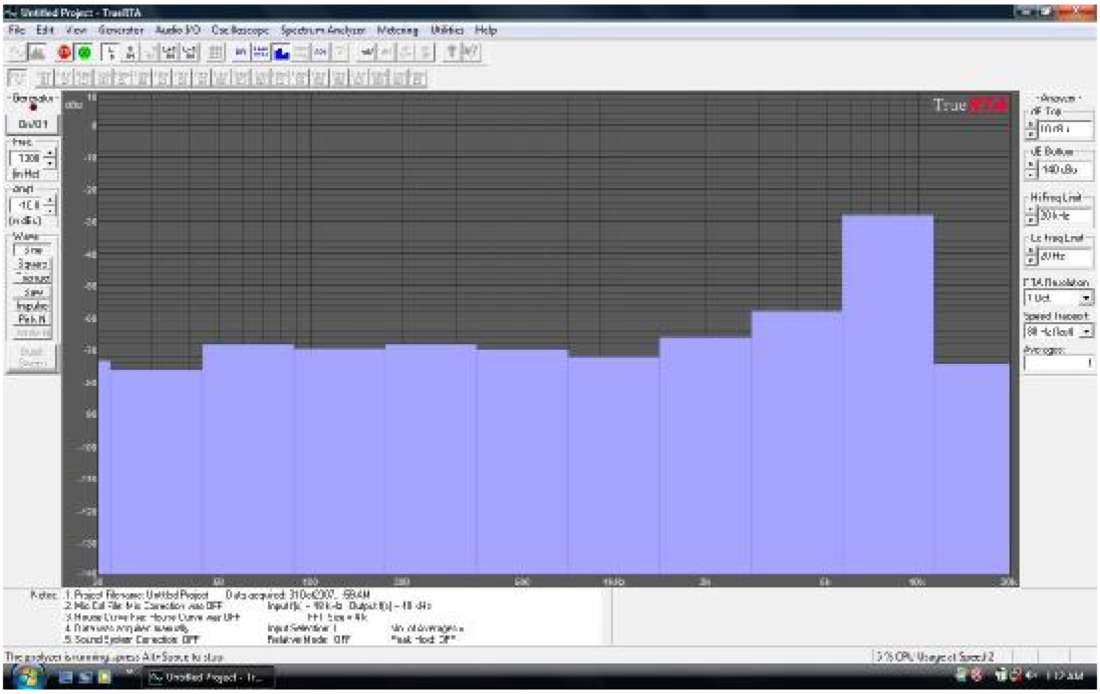

Figure 4.3 : TrueRTA shows the output after power amplifier stage if a high frequency is set as an input

Figure 4.3 shows the output after the power amplifier stage. Need to be mentioned that the input signal still same as before (10kHz), only this time the output is connected to power amplifier firstly before its been feed to the tweeter.

As we can see, the amplitude has been increased into -28dB. Comparing the result obtained after the power-amp stage and after the crossover stage means that the total amplitude increasing that the power amplifier has done is 24dB.

4.2.3 Low Frequency Input

4.2.3.1 Output after Crossover Stage

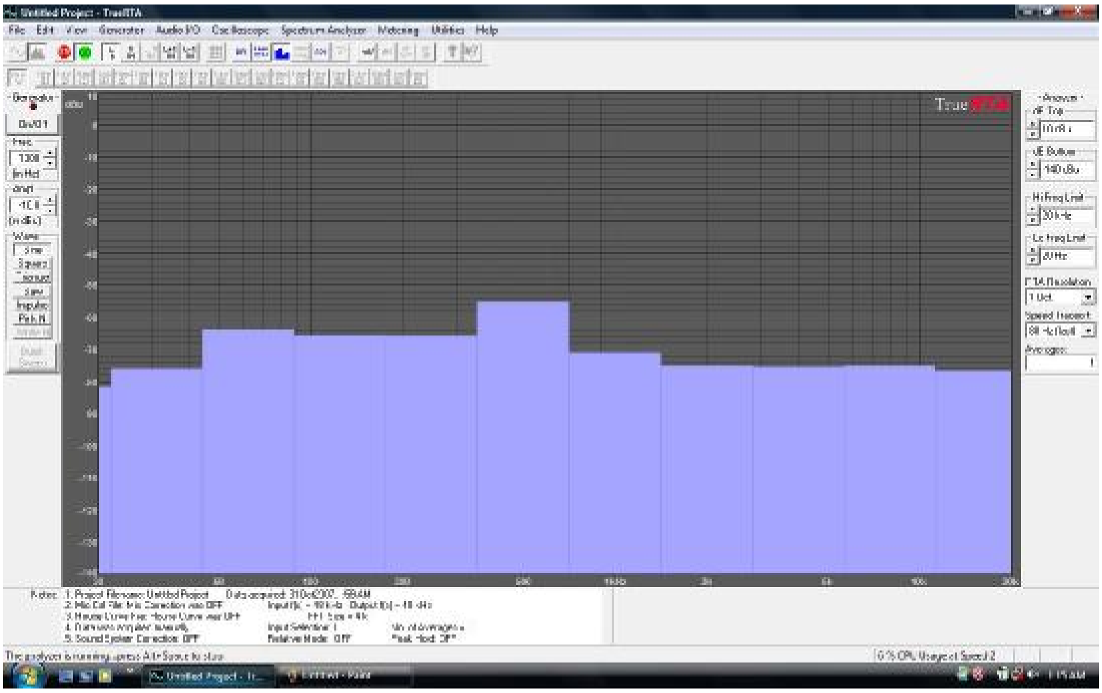

Figure 4.4 : TrueRTA shows the output after crossover stage if a low frequency is set as an input

Figure 4.4 above shows the output after the crossover stage if a low frequency is set as an input. At this time, the testing is performed by giving the system an input frequency of 500Hz and cut the connection to the tweeter. It is done to ensure the result obtained is from subwoofer output only.

As we can see, TrueRTA catch the frequency that has been set earlier (500Hz). The low-pass filter has eliminates the high frequency and just allows the high frequency to pass through.

4.2.3.2 Output after Power Amplifier Stage

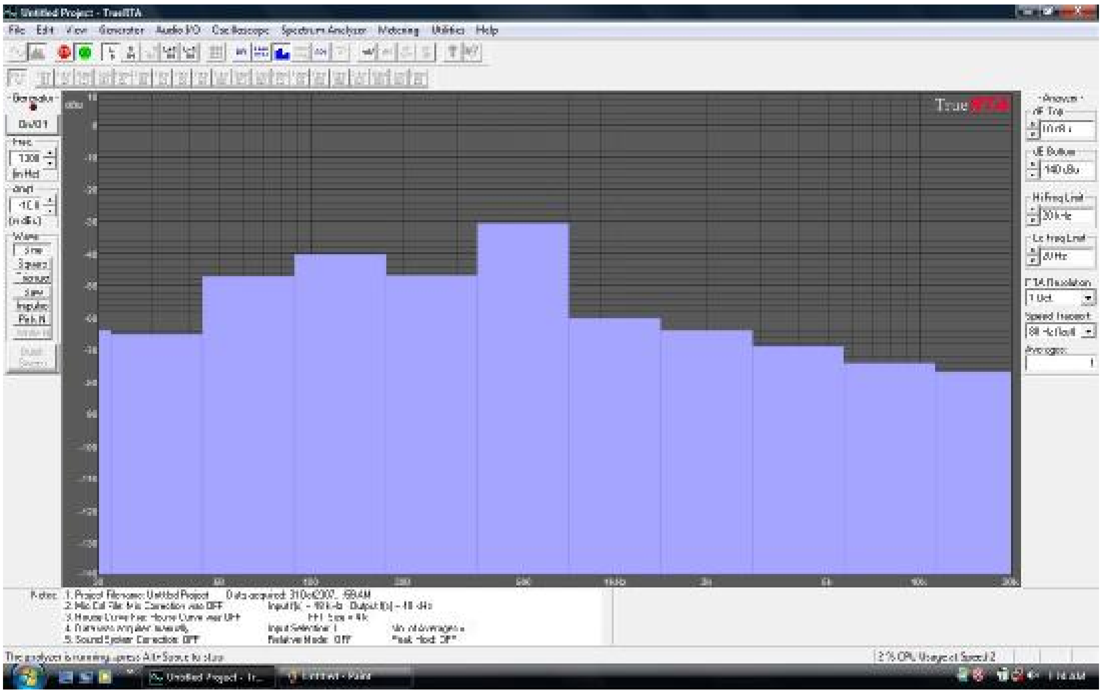

Figure 4.5 : TrueRTA shows the output after power amplifier stage if a low frequency is set as an input

Figure 4.5 shows the output after the power amplifier stage. The input signal still same as before; 500Hz, but only this time the output is obtained after the subwoofer has been connected to the power amplifier output.

As we can see, the amplitude of the signal has been increased into -31dB; increase about 24dB from the crossover stage (-55dB). This is totally same as the output that we obtained earlier for low frequency input that the power amplifier also provides total amplitude increasing about 24dB.

4.3 Obtaining the Result using Oscilloscope

4.3.1 High Frequency Input

4.3.1.1 Output at High Frequency Line

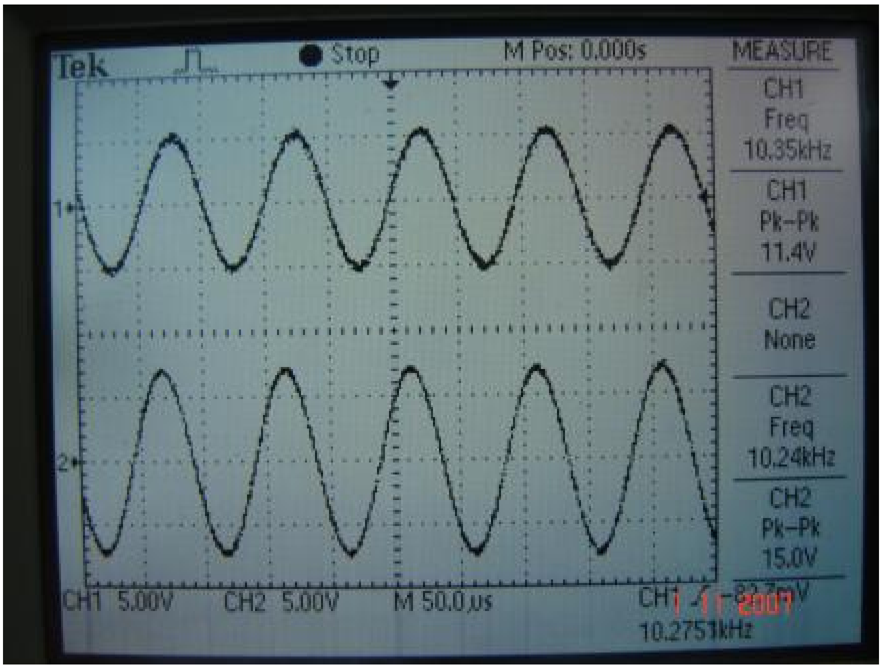

Figure 4.6 : Oscilloscope shows high frequency input and high frequency line output

Figure 4.6 above shows the input signal and the output at high frequency line. The input signal is 10.35kHz with 11.4V peak-to-peak voltage. After it has been filtered and amplified, the frequency slightly different from the original signal (10.24kHz) with peak-to-peak has been increased into 15.0V. Actually the frequency was not perfectly at 10.24kHz but changing in the range of 10.35kHz through 10.22kHz. It means that the circuit does not adding or taking away anything from the original signal. It's just the voltage that has been increased (amplified) as peak-to-peak voltage of the original signal is just 11.4V while the output signal is producing 15.0V peak-to-peak voltage.

4.3.1.2 Output at Low Frequency Line

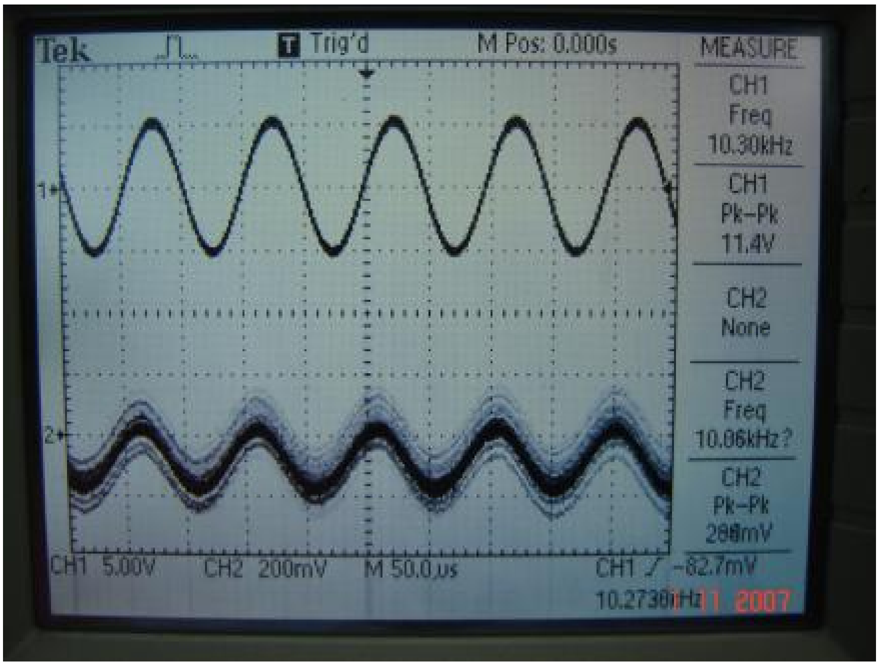

Figure 4.7 : Oscilloscope shows high frequency input and low frequency line output

Figure 4.7 shows the input signal and output at low frequency line. The input is 10.30kHz with peak-to-peak voltage of 11.4V. Meanwhile the output is determined as 10.06kHz and peak-to-peak voltage as 289mV.

As we can see, the output is not smooth enough because the low-pass filter has filtered the high frequency input. The symbol “?” at the output frequency means that the oscilloscope is unable to determined the accurate frequency of the output signal. The peak-to-peak voltage value also not at a fixed value and this is caused by the inconsistent signal produced. The output also maybe just a noise that has been amplifies by the power amplifier.

4.3.2 Low Frequency Input

4.3.2.1 Output at High Frequency Line

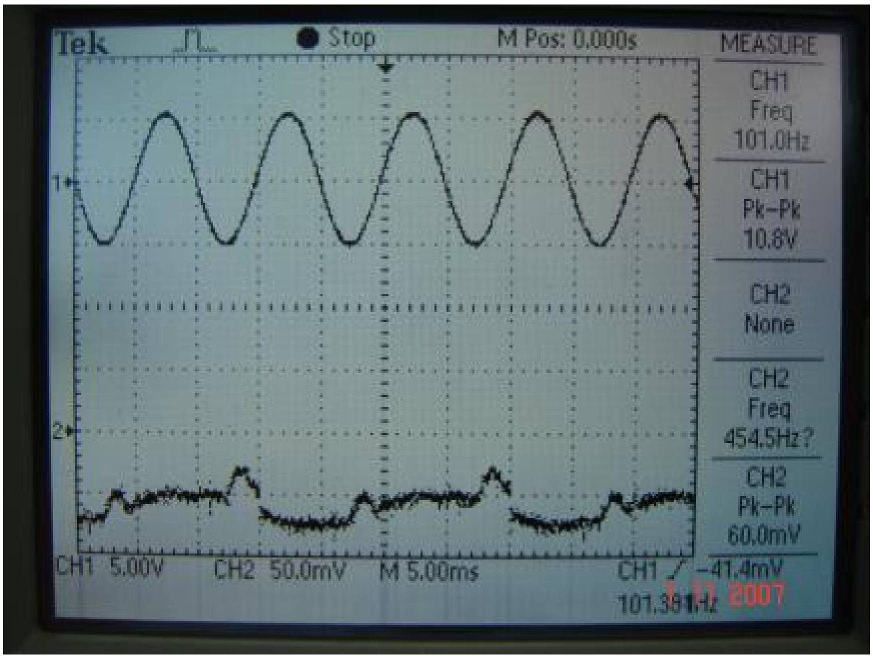

Figure 4.8 : Oscilloscope shows low frequency input and high frequency line output

Figure 4.8 shows the input signal and output at low frequency line. The input is 101.0Hz with peak-to-peak voltage of 10.8V. The output frequency is determined as 454.5Hz and peak-to-peak is 60.0mV.

As we can see, the output is totally not same as the original input signal. This is because the high-pass filter has filtered the low frequency input. As from before, the symbol "?" present at the output frequency means that the oscilloscope is unable to determined the accurate frequency of the output signal. The peak-to-peak voltage value of the output signal is too low and maybe means that the signal producing is just an amplified noise.

4.3.2.2 Output at Low Frequency Line

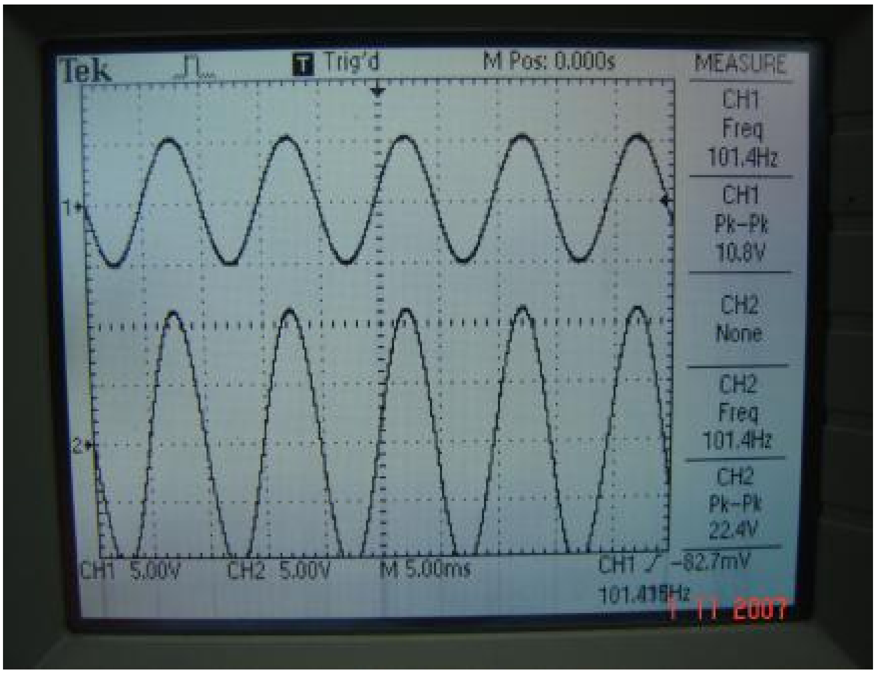

Figure 4.9 : Oscilloscope shows low frequency input and low frequency line output

Figure 4.9 above shows the input signal and the output at low frequency line. The input signal is set at 101.4Hz with 10.8V peak-to-peak voltage. After it has been filtered and amplified, the frequency is still same as the original signal with peak-to-peak has been increased into 22.4V.

Both frequencies for input and output signal are same because the process of filtering and amplifying is not involving on changing the original frequency. The output clearly shows the increasing of the peak-to-peak voltage of the signal. This process is done by the power amplifier without changing the signal frequency.

CHAPTER 5

CONCLUSION AND RECOMMENDATION

5.1 Conclusion

As a conclusion, the project conducted has met its objective, which is to study and build an active loudspeaker system. Hopefully, the device will help us on connecting any sources to the loudspeaker to produce sound without have to buy separated power amplifier.

The construction of the system is a science as much as it is an art. There are a specific procedure had to be followed to ensure that the system will meet its earlier expectation. A time is needed to gain knowledge and mastering audio system.

Generally, the basic of audio system is very simple and everyone can learn about it with a great understanding. However, the loudspeaker system itself is a very complex system. The systems require us to have a clear understanding on it and therefore mastering it. There are also many aspects that needed to be considered such as the circuit types, function, speaker chosen and material used to build the enclosure.

5.2 Recommendation

As the recommendation, I would like to suggest that we should study further and more details on this system. Further system can consider on producing a stereo system instead of mono system that I have built. The system that will be built will be more effective in term of sound producing.

Future project should try to build a 3-way system. Although the system is more complex than the system that I have built, it will provide more knowledge to the project maker and at the same time the community will also gain useful output from the research.

Furthermore to avoid the system from feeling the effect of noises, a proper and effective action is needed. The wire, connector and the component used to build the system need to be chosen properly as the noise may come from the circuit itself. An equalizer or filter also can be added in order to eliminate the noises and at the same time contributing on producing a much better system.

5.2.1 Costing and Commercialization

The total cost for this project is about RM455.00. This cost includes the cost of circuit along with the enclosure. Although the product is a hand-made it still can be commercialize because it is a useful equipment which suitable to be used by everyone. It’s just that the product need to compete with same system produced by established companies such as Altec Lansing, Creative, Behringer and others.

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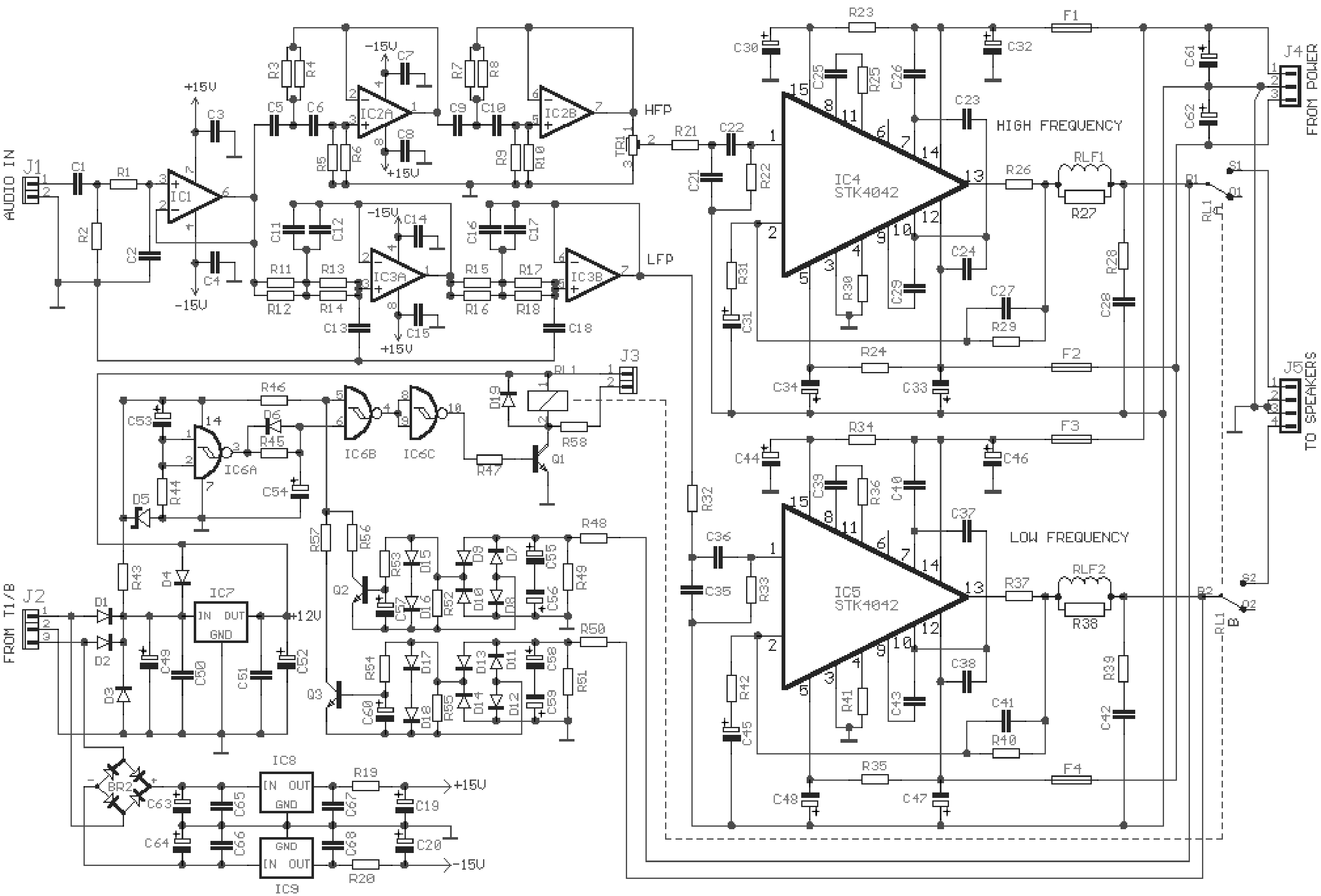
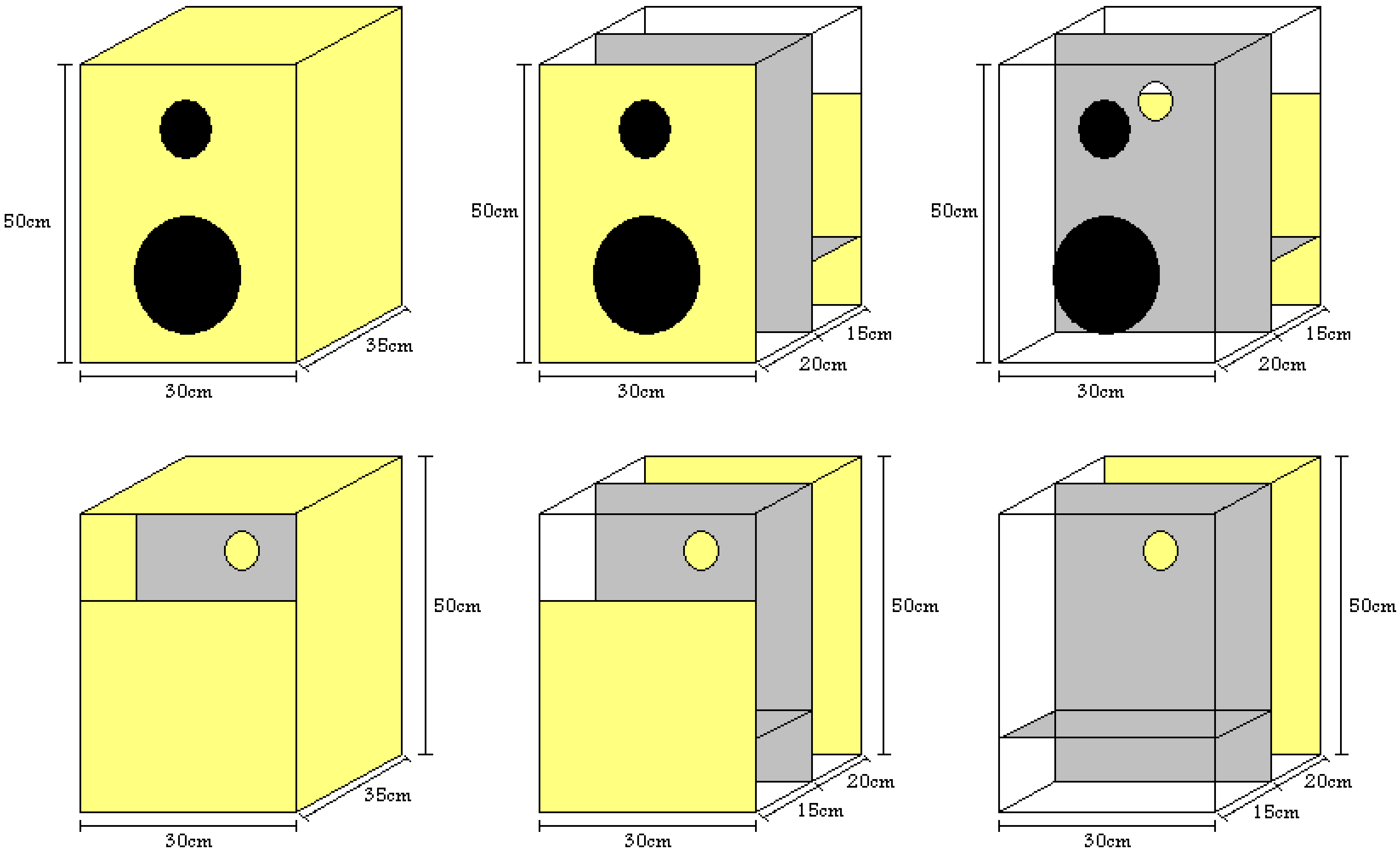
URL : http://www.crest-performance.com/loudspeakers/whyactive.cfm

APPENDIX A
COMPLETE CIRCUIT DIAGRAM


AUDIO IN
J1
HFP
LFP
HIGH FREQUENCY
IC4
STK4042
LOW FREQUENCY
IC5
STK4042
J4
FROM POWER SUPPLY
J5
TO SPEAKERS
J3
J2
FROM T1/B
IC6A
IC6B
IC6C
IC7
IN OUT
GND
+12V
IC8
IC9
+15V
-15V
RLF1
RLF2
RL1

APPENDIX B
ENCLOSURE DESIGN DIAGRAM


50cm
35cm
30cm
50cm
15cm
20cm
30cm
50cm
15cm
20cm
30cm
50cm
35cm
30cm
50cm
20cm
15cm
30cm
50cm
20cm
15cm
30cm

APPENDIX C
OPERATION MANUAL

1. Plug-in the power socket
2. Turn on the power switch
3. Connect the device input jack to audio source
4. Turn on the device
5. Play the audio

APPENDIX D
LIST OF COMPONENTS

R1,21,32,25,36,58=1 Kohms	C19,20,52=10uF 25V	IC1=TL071
R2=47 Kohms	C21,35=470pF	IC2,3=TL072,NE5532
R3,4,5,7,8,9,46=22 Kohms	C22,36=470nF 63V MKT	IC4,5=SKT4042[II]
R6,10=N.C	C24,25,26,38,39,40=100pF	IC6=4093
R11 - 18=22 Kohms	C27,41=10pF	IC7=7812T
R19,20=47 ohms	C28,42=100nF 100V MKT	IC8=7815T
R22,33=33 Kohms	C29,43=1nF 63V MKT	IC9=7915T
R23,24,34,35=100 ohms	C30,34,44,48=100uF 63V	RL1=Relay 12V [G2R2 Omron]
R26,37=0.22 ohms 5W	C31,45=220uF 25V	RLF1,2=
R27,38=10 ohms 3W	C32,33,46,47=10uF 63V	F1,2,3,4=1.6A FAST 5X20mm
R28,39=6.8 ohms	C49=47uF 25V	F5=1A SLOW 5X20mm[Fig.2]
R29,40=12 Kohms	C50,51=100nF 63V MKT	T1=220V//A=2X30V 250VA B=2X15V 30VA [Toroidal]
R30,41,53,54=10 Kohms	C53=1uF 25V	
R31,45=560 ohms	C54=3.3uF 25V	
R44,45=1 Mohms	C55,56,58,59=33uF 63V	JF1=3pin male supply jack
R47=39 Kohms	C57,60=22uF 16V	JF2=Female RCA Jack
R48,50=15 Kohms	C61,62=15000uF 63V AXIAL	J1-3=2pin conn. with 2.54mm pin step
R49,51,52,55=56 Kohms	C63,64=2200uF 25V AXIAL	J2=3pin conn. with 2.54mm pin step
R56,57=3.9 Kohms	C65,66,67,68=100nF 63V MKT	J4=3pin conn. with 3.96mm pin step
R43=470 ohms 1W	Q1=BD679	J5=4pin conn. with 3.96mm pin step
TR1=47 Kohms trimmer	Q2,3=BC550	T=Tweeter 8ohms 50 until 80W
C1,22,36,23,37=1uF 63V MKT	D1,2,3,4=1N4002	W=Woofer 8ohms 50 until 100W
C2=390pF	D5=8.2V 0.5W Zener	BR1=Bridge rect. 400V 25A [Fig.2]
C3,4,7,8,14,15=100nF 63V MKT	D6=1N4148	BR2=Bridge rect. 100V 1.5A
C5,6,9,10,11,12=3.3nF 63V MKT	D7.....19=1N4148	
C13,16,17,18=3.3nF 63V MKT	D20=5mm LED [Fig.2]	

APPENDIX E

STK4042II DATASHEET

Ordering number : EN1179A

Thick Film Hybrid IC

STK4042 II

AF Power Amplifier (Split Power Supply) (80 W min, THD = 0.4%)

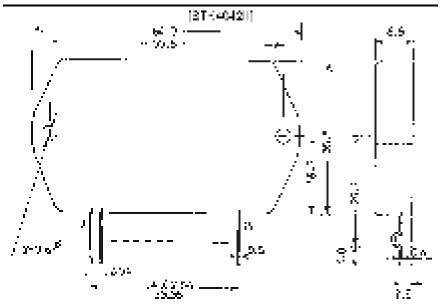
Features

- Miniature package allows audio sets to be made slimmer.
- Pin compatible amplifiers with outputs of 20 to 200 W are available.
- Facilitates thermal design of slim stereo sets by distributing the heat dissipating IC's in the set.
- The adoption of constant current circuits reduces pop noise when the power supply is turned on or off.
- Supports the design of supplementary electronic circuits (thermal shutdown, load short protection, and pop noise muting at power on and off).

Package Dimensions

unit: mm

4075



Specifications

Maximum Ratings at Ta = 25°C

Parameter	Symbol	Condition	Rating	Unit
Maximum supply voltage	V_{CC} max		±65	V
Thermal resistance	θj-c		1.2	°C/W
Junction temperature	Tj		150	°C
Operating case temperature	Tc		125	°C
Storage temperature	Tstg		[illegible] to +125	°C
Available time for load shorted	ts*	V_{CC} = ±45 V, R_L = 8 Ω, f = 50 Hz, P_O = 80 W	2	s

Note: Use a constant voltage power supply as the test power supply unless otherwise specified.
* Use the transformer power supply shown on the next page when measuring the available time for load shorted and the output noise voltage.

Recommended Operating Conditions at Ta = 25°C

Parameter	Symbol	Condition	Rating	Unit
Recommended supply voltage	V_{CC}		±45	V
Load resistance	R_L		8	Ω

Operating Characteristics at Ta = 25°C, V_{CC} = ±45 V, R_L = 8 Ω (noninductive load), Rg = 600 Ω, VG = 40 dB

Parameter	Symbol	Condition	Rating min	Rating typ	Rating max	Unit
Quiescent current	I_{CCO}	V_{CC} = ±54 V	15		120	mA
Output power	P_O	THD = 0.4%, f = 20 Hz to 20 kHz	80			W
Total harmonic distortion	THD	P_O = 1.0 W, f = 1 kHz			0.3	%
Frequency response	f_L, f_H	P_O = 1.0 W, $^{+0}_{-3}$ dB		20 to 50 k		Hz
Input resistance	r_i	P_O = 1.0 W, f = 1 kHz		55		kΩ
Output noise voltage	V_{NO}**	V_{CC} = ±54 V, Rg = 10 kΩ			1.2	mVrms
Neutral voltage	V_N	V_{CC} = ±54 V	-70	0	+70	mV

Note: Use a constant voltage power supply as the test power supply unless otherwise specified.
** The output noise voltage is the peak value measured with an averaging rms scale volt meter. The noise voltage waveform should not include pulse noise.

APPENDIX F

TL071 & TLO72 DATASHEET

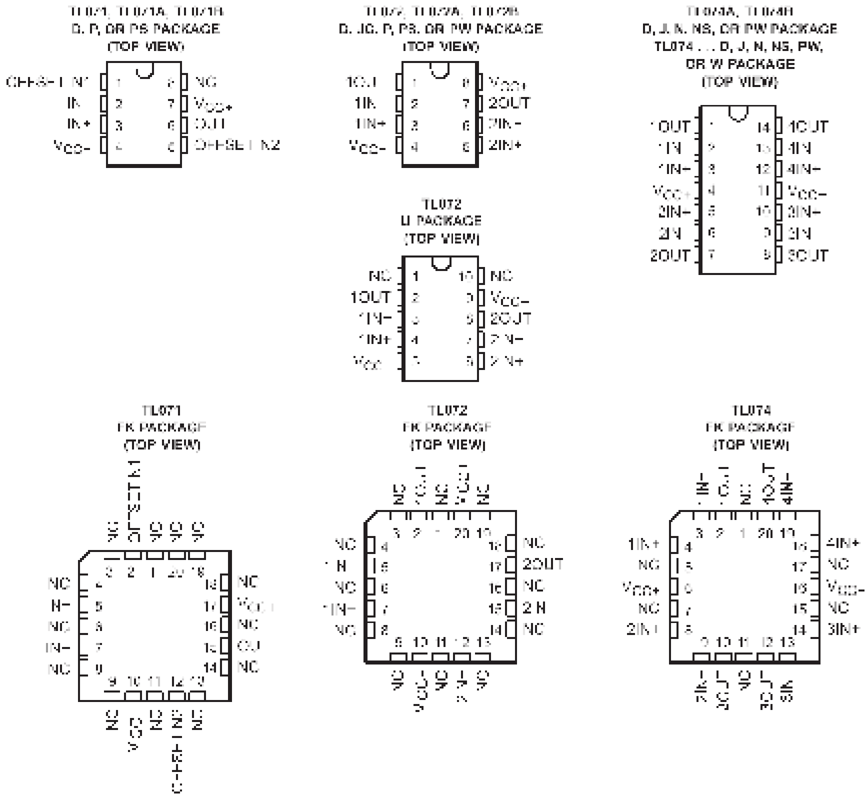
- Low Power Consumption
- Wide Common-Mode and Differential Voltage Ranges
- Low Input Bias and Offset Currents
- Output Short-Circuit Protection
- Low Total Harmonic Distortion . . . 0.003% Typ
- Low Noise
  V_n = 18 nV/√Hz Typ at f = 1 kHz
- High Input Impedance . . . JFET Input Stage
- Internal Frequency Compensation
- Latch-Up-Free Operation
- High Slew Rate . . . 13 V/μs Typ
- Common-Mode Input Voltage Range Includes V_{CC+}

description/ordering information

The JFET-input operational amplifiers in the TL07x series are similar to the TL08x series, with low input bias and offset currents and fast slew rate. The low harmonic distortion and low noise make the TL07x series ideally suited for high-fidelity and audio preamplifier applications. Each amplifier features JFET inputs (for high input impedance) coupled with bipolar output stages integrated on a single monolithic chip.

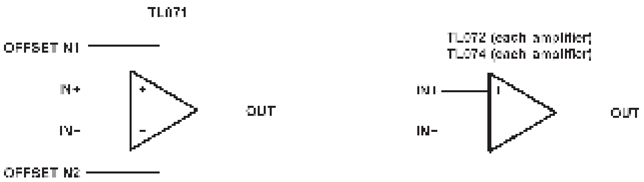
The C-suffix devices are characterized for operation from 0°C to 70°C. The I-suffix devices are characterized for operation from −40°C to 85°C. The M-suffix devices are characterized for operation over the full military temperature range of −55°C to 125°C.

Please be aware that an important notice concerning availability, standard warranty, and use in critical applications of Texas Instruments semiconductor products and disclaimers thereto appears at the end of this data sheet.



NC − No internal connection

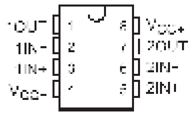
symbols



APPENDIX G
NE5532 DATASHEET

- **Equivalent Input Noise Voltage 5 nV/√Hz Typ at 1 kHz**
- **Unity-Gain Bandwidth . . . 10 MHz Typ**
- **Common-Mode Rejection Ratio . . . 100 dB Typ**
- **High dc Voltage Gain . . . 100 V/mV Typ**
- **Peak-to-Peak Output Voltage Swing 32 V Typ With $V_{CC\pm} = \pm 18$ V and $R_L = 600\ \Omega$**
- **High Slew Rate . . . 9 V/μs Typ**
- **Wide Supply-Voltage Range . . . ±3 V to ±20 V**

NE5532, NE5532A . . . D, P, OR PS PACKAGE
SA5532, SA5532A . . . D OR P PACKAGE
(TOP VIEW)



description/ordering information

The NE5532, NE5532A, SA5532, and SA5532A are high-performance operational amplifiers combining excellent dc and ac characteristics. They feature very low noise, high output-drive capability, high unity-gain and maximum-output-swing bandwidths, low distortion, high slew rate, input-protection diodes, and output short-circuit protection. These operational amplifiers are compensated internally for unity-gain operation. These devices have specified maximum limits for equivalent input noise voltage.

ORDERING INFORMATION

T_A	PACKAGE†		ORDERABLE PART NUMBER	TOP-SIDE MARKING
0°C to 70°C	PDIP – P	Tube of 50	NE5532P	NE5532P
			NE5532AP	NE5532AP
	SOIC – D	Tube of 75	NE5532D	N5532
		Reel of 2500	NE5532DR	
		Tube of 75	NE5532AD	N5532A
		Reel of 2500	NE5532ADR	
	SOP – PS	Reel of 2000	NE5532PSR	N5532
			NE5532APSR	N5532A
−40°C to 85°C	PDIP – P	Tube of 50	SA5532P	SA5532P
			SA5532AP	SA5532AP
	SOIC – D	Tube of 75	SA5532D	SA5532
		Reel of 2500	SA5532DR	
		Tube of 75	SA5532AD	SA5532A
		Reel of 2500	SA5532ADR	

† Package drawings, standard packing quantities, thermal data, symbolization, and PCB design guidelines are available at www.ti.com/sc/package.

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