EQUALIZER FOR PA SYSTEM

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BORANG PENGESAHAN STATUS TESIS♦

JUDUL: **EQUALIZER FOR PA SYSTEM**

SESI PENGAJIAN: **2007/2008**

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Date : <u>23 NOVEMBER 2007</u>

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This thesis is submitted as partial fulfillment of the requirements for the award of the Bachelor of Electrical Engineering (Hons.) (Electronics)

Faculty of Electrical & Electronics Engineering
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NOVEMBER, 2007

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Author : MOHD ARIFFUDDIN B MOKHTAR

Date : 23 NOVEMBER 2007

To my late father, Allahyarham Haji Mokhtar bin Haji Abdul Ghani,
Al-Fatihah

Dedicated this work also to my family especially my mom

ACKNOWLEDGEMENT

In the name of Allah, the most loving and the most compassionate

I would like to take this opportunity to express my sincerest appreciation to everyone that officially or none officially contributed to this thesis especially my supervisor Associate Professor Shaikh Nasir bin Shaikh Ab. Rahman. Without his guidance I will not be able to produce this thesis. Once again, thank you very much.

I also would like to thank Mr. Mohd Salmizan b Mohd Zain, our laboratory technician, because with his help I'm able to finish this project on time and not to forget my friends that always give me ideas and useful advice.

Last but not least I wish to extend my appreciation to my entire family member that always gives me a support and always stand behind me. Thank you.

ABSTRACT

In audio processing, equalization is the process of changing the frequency envelope of a sound. In passing through any channel, frequency spreading occurs. In this project I have build 9 band graphic equalizer. The graphic equalizer has become very popular in recent years. It is called graphic because, as the front-panel sliders or knob are adjusted; their positions give an approximate display of the resultant frequency response. This device divide the audible spectrum into nine frequency bands, and allow adjustments to each band via its own boost/cut control gain from 1.17 to 6. A growing number of audio enthusiasts are using equalizers to adjust their stereo system's frequency response, whether to compensate for room acoustics, for PA system or for creative recording purposes. Instead of broad adjustments of treble, bass, and maybe the midrange (sometimes called presence), we now have independent control over the low bass, mid-bass, high bass, low midrange, and so forth.

ABSTRAK

Di dalam pemprosesan audio, penyamarataan adalan proses mengubah sampul frekuensi bunyi. Bunyi akan melalui beberapa saluran dan akan di pecah kan kepada beberapa mengikut jalur frekuensi. 9 jalur penyamarata grafik akan di bina di dalam projek ini. Penyamarata grafik sebegini telah popular kebelakangan ini. Iannya di panggil penyamarata grafik kerana di panel hadapan terletaknya knob dimana kedudukan knob akan menandakan kedudukan secara kasar hasil respon frekuensi. Alat ini akan membenarkan setiap jalur mempunyai gandaan dari 1.17 hingga 6. Kebanyakan peminat di dalam pemprosesan audio menggunakan penyamarata untuk mengubah respon frekuensi stereo, sama ada untuk kegunaan bilik akustik, sistem hebahan atau juga untuk kegunaan merekod. Sebagai langkah alternatif daripada mengubah sesuatu yang besar julatnya untuk ***treble, bass***, dan juga ***midrange*** (kadang kala di panggil *presence*), kita boleh mengubah julat yang lebih kecil seperti ***low bass, mid-bass, high bass, low midrange***, dan seterusnya.

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LIST OF ABBREVIATIONS

PA	-	Public Address
PPH	-	Jabatan Pembangunan & Pengurusan Harta
EQ	-	Equalizer
f_s	-	Frequency of source
DJ	-	Disc Jockey
RTA	-	Real time analyzer
CD	-	Compact disc
PCB	-	Printed circuit board
DSP	-	Digital Signal Processing
PC	-	Personal computer

LIST OF SYMBOLS

m/s	-	Meter per second
km/h	-	Kilometer per hour
m	-	Meter
T	-	Absolute temperature
M	-	Mass of the molecular gas
R	-	Constant of gas
υ	-	Sound of speed
^{o}C	-	Celcius degree
Hz	-	Hertz
ΔL	-	Difference between 2 power level
dB	-	Decibels
I	-	Intensity
p	-	Pressure
ρ	-	Density of air
V	-	Voltage
A	-	Ampere

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CHAPTER I

INTRODUCTION

1.1 Problem Statement

In order to throw out a simple gathering or function, the current practices and procedures applied is quite intensify as one needs to fill a form and wait for turn. Yet for a small gathering or a prompt function which need to be done in such a hurry, this will cause slightly a problem.

In order to unravel this problem and make things less complicated for all personnel involved, students from Faculty of Electrical Engineering will design PA system that consist of mixer, equalizer, and power amplifier that can be used for a small function or gathering.

.

1.2 Introduction To Project

Engineering Project I, BEE 4172 is a partial requirement for Bachelor of Electrical Engineering. For this particular course, I’ve been given a task to construct, building and testing of an equalizer unit to be used as part of a public address (PA) system. The equalizer should have a minimum of five different channels for adjusting the frequency response of the input audio signal

The purpose of this project is to make the quality sound of a public address system better than before it use the equalizer. This can happen by cutting or boosting the gain of the output in the range of frequency. Equalizer (EQ) is a combination of a filter; usually adjustable, chiefly meant to compensate for the unequal frequency response of some other signal processing circuit or system. An EQ filter typically allows the user to adjust one or more parameters that determine the overall shape of the filter's transfer function. It is generally used to improve the fidelity of sound, to emphasize certain instruments, to remove undesired noises, or to create completely new and different sounds.

1.3 Objective

The objectives of this project are to;

i. To make a useable equalizer for faculty.
ii. To synchronize this device with mixer and power amplifier to make a low cost PA system for faculty.

1.4 Scope of Works

In this project I will focus on;

i. The designation of physical look of the equalizer.
ii. Fabrication of equalizer circuit.
iii. Testing the overall performance of this device.

CHAPTER 2

LITERATURE REVIEW

2.1 Introduction to Sound

Sound is wave that carries information from one point to another point as well as energy. In this thesis the focus is more on the longitudinal sound waves in air. Longitudinal means that the back and forth motion of air is in the direction of travel of the sound wave. The changing in air pressure allows us to hear which will be covered later in this chapter later on.

2.1.1 Source of Sound

There are so many sources of sound that we can find but in this thesis I will focus more on vibrating bodies and changing airflow. Vibrating bodies is a when something like guitar string or piano vibrates it displaces the air next to it and causes the air pressure to increase and decrease slightly. These air pressure fluctuations can be called as sound wave. This source of sound is the familiar source that we can find.

Changing airflow happens when we speak or sing and when our vocal folds alternately open and close so that the rate of air from our lung increases and decreases, resulting in a sound wave. We can find this situation when people use some music instruments such as like clarinet, saxophone or trumpet.

2.1.2 Noise

There are many sounds that we hear everyday but not all of that we wanted to hear. The unwanted sound we called as noise. In this thesis the discussion are more on one way to eliminate one noise that we called as acoustic feedback.

Acoustic feedback occurs when the amplified sound from any loudspeaker reenters the sound system through any open microphone and is amplified again and again and again.

There are so many way to eliminate acoustic feedback. It can be divided into two. Firstly one is by physical way including ask the talker to talk more louder, reduce the distance from the talker to the microphone, reduce the number of open microphones, and move the loud speaker farther away from the microphone.

Secondly is by using electrical way such as using an equalizer to cut the frequency band in which the feedback occurs or using frequency shifter to shift the frequency band that the feedback occurs.

2.2 Waves

As we already know that sound waves in air are longitudinal waves. Many other types of waves such as light waves and radio waves are transverse waves. Even these types of waves are different from sound waves but they have similar behavior.

The first thing about the waves is they can transport energy and also information from one place to another place through a medium.

The other properties are they can be reflected, refracted, or diffracted [1]. But they all travel with different speed in different medium. Example sound waves travel through air as a medium only 344m/s ≈ 600km/h but light can travel as fast as 3×10^8 m in 1 second.

2.2.1 Sound Waves

Sound waves can travel through many medium states such as solid, liquid, or gas. We can try to test the movement characteristic of sound waves by placing a large pipe or tube with the loudspeaker at one end. An electrical impulse to the loudspeaker causes the cone to move forward suddenly, compressing the air in front of it very slightly. This pulse of air pressure travels down the tube at a speed of about 340m/s. It may be absorbed at the far end of the tube, or it may ne reflect back toward the loudspeaker depending in what is at the end of the tube [1].

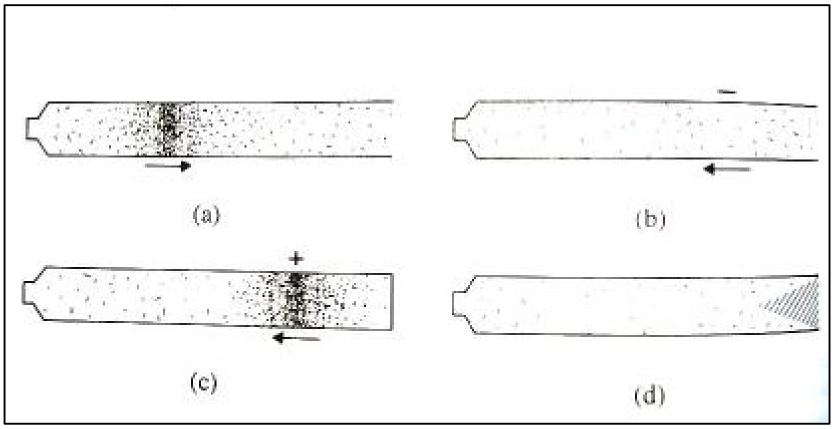


Figure 2.1: Reflection of the sound pulse in a pipe; (a) incident pulse; (b) reflection at an open end; (c) reflection at a closed end; (d) no reflection on the absorbing end.

Reflection of a sound pulse for a 3 different end condition is illustrated in Figure 2.1. If the end is open the excess pressure drops to zero, and the pulse reflects back as negative pulse of pressure as can be seen at Fig. 2.1 (b). If the end is closed the pressure build up to twice it value and the pulse reflects back as a positive pulse of pressure as can be seen in the Fig. 2.1 (c). If the end is terminated with a sound absorber, there is virtually no reflected pulse. Such a termination is called anechoic, which mean “no echo” [1].

The speed of sound can be known by using this formula

$$v = \sqrt{\frac{\gamma R T}{M}}$$

Where T is absolute temperature, M is the mass of the molecular gas and γ and R is the constant of the gas. The estimation value for M is 2.88 × 10-2, when R is 8.31 and γ is 1.4. By using all this value on that formula we can get $v = 20.1\sqrt{T}$. As it said before, T is absolute temperature by adding 273 and the Celsius scale. For an example if the temperature is 23 so T will become 300.

To make it simpler we can simplified the formula because the speed of sound increases by about 0.6m/s for each Celsius degree. So the new formula will be:

$$v = 331.3 + 0.6t\ m/s$$

Sound can travels faster in other medium such as liquid and solid. The speed of sound on several materials can be referred in Table 2.1.

Table 2.1: Speed of sound in several materials

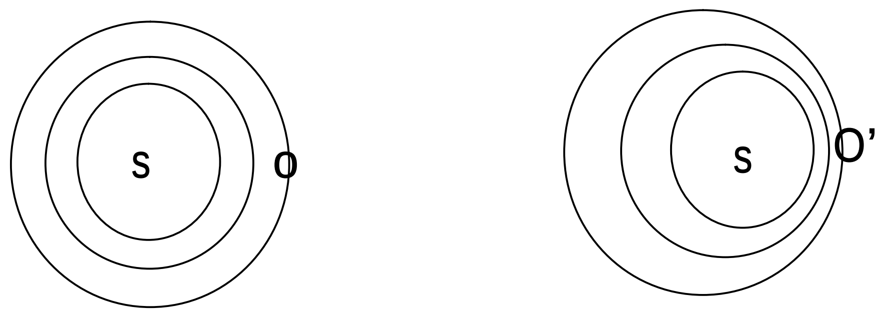
Substance	Temperature (°C)	Speed (m/s)	Speed (ft/s)
Air	0	331.3	1,087
Air	20	343	1,127
Helium	0	970	3,180
Carbon dioxide	0	258	846
Water	0	1,410	4,626
Methyl alcohol	0	1,130	3,710
Aluminum	—	5,150	16,900
Steel	—	5,100	16,700
Brass	—	3,480	11,420
Lead	—	1,210	3,970
Glass	—	3,700–5,000	12–16,000

This table is taken from [1] where all the data here is subject to the result of an experiment made by the author. This thesis only takes this information for a review.

2.2.2 Doppler Effect

When the frequency of the source, f_s reach the observe, the frequency of the sound should be same as f_s but this only happen if neither source nor observer aren't moving. If they are moving toward each other, the observed frequency is greater than f_s; if they are moving apart, the observed frequency is lower than f_s. This apparent frequency shift is called the Doppler Effect [1].

The meaning of Doppler Effect can be made clearer by with the aid of a figure. From Figure 2.2, the center of the circle is source, S, and the observer, O at rest. If the source emits 100 waves per second, an observer at rest will receive the exact amount as the source emit. However a moving observer O' will receive more than 100 waves per second because it move toward source. A moving observer that moving towards will receive more because he or she "meets" the waves as he or she moves.



(a) (b)

Figure 2.2 Doppler Effect (a) observer at rest (b) Moving observer

From the Figure 2.2 we can know the apparent frequency (the rate at which the observer meets waves). It will be

$$f' = f_s\left(\frac{v + v_o}{v}\right)$$

υ_o = the speed of the observer

υ = speed of sound

Note that if the observer passed the sound source, υ_o must be subtracted from the υ. Thus the frequency drop abruptly as the observer passes the source. There is another case if the source in motion. The observer will receive a greater rate than he or she would get from a stationary source. The apparent frequency will be

$$f = f_s\left(\frac{v}{v - v_s}\right)$$

But if the source moves directly toward observer, the frequency will drop abruptly, not gradually as the source pass by.

2.3 Frequency Response

As mentioned earlier, sound is cause by a vibration in air. This vibration will produce two things for sure. The first one is frequency and the second one is amplitude. Vibration occurs in a single wavelength and frequency is a measurement how much vibration occurs in a single second. It usually measured in Hertz (Hz). Frequency is directly correspondent to the pitch of sound. To make it clearer we can see from the Figure 2.3.

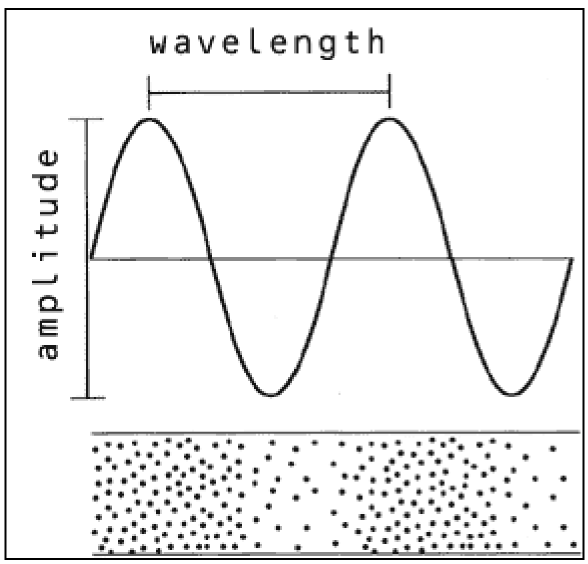


Figure 2.3: Two component in waves

Frequency response is the measure of any system's response at the output to a signal of varying frequency (but constant amplitude) at its input. It is usually referred to in connection with electronic amplifiers, loudspeakers and similar systems. The frequency response is typically characterized by the magnitude of the system's response, measured in dB, and the phase, measured in radians, versus frequency [2].

This project main focus is to have the desired frequency response by using an equalizer. The introduction to the equalizer will be discussed later on this chapter. The range of frequency that human can hear is from as low as 20 Hz till as high as 20 kHz. The desired frequency response is something like in Figure 2.4.

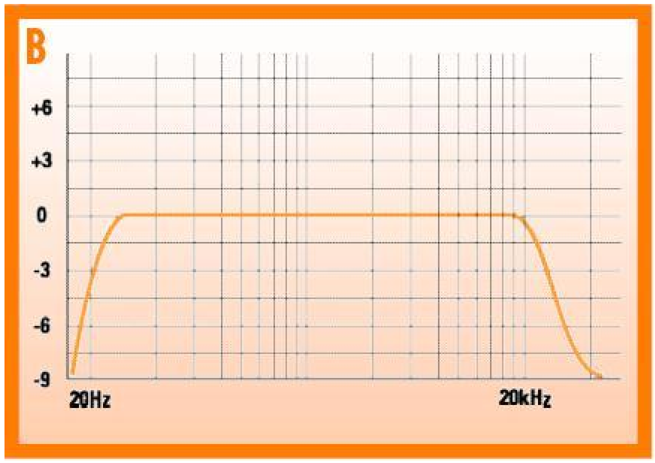


Figure 2.4: The desired Frequency Response

Figure 2.4 showed the ideal flat frequency response that mostly people want. But to achieve that in the real world is quite impossible even the flat response microphone has some deviation. More importantly, it should be noted that a flat frequency response is not always the most desirable option. In many cases a tailored frequency response is more useful. For example, a response pattern designed to emphasize the frequencies in a human voice would be well suited to picking up speech in an environment with lots of low-frequency background noise [3].

2.4 Human Auditory System

Human auditory system is a complex in structure and remarkable in function. Not only does it respond to a wide range of stimuli, but it precisely identifies the pitch and timbre (quality) of a sound and even the direction of the source. Much of the hearing function is performed by the organ named ear, but recent research has emphasized how much hearing depends on the data processing occurs in the central nervous system as well.

2.4.1 Range of Hearing

Most people will find that their hearing is most sensitive at 1-4 kHz and that is less sensitive at high and low frequency. Most of the children can hear frequency that a lot higher than what an adult can hear. That is why sometimes your children keep on complaining about sound that you cannot hear. People will loss their capability of hearing a higher frequency as they grow older. It is common for adults to have very low sensitivity for the highest frequencies.

It still debated issues about range of frequency that human can hear but it accepted worldwide that human can hear in range 20 Hz to 20 kHz. Sound below 20 is infrasonic and sound above 20 kHz is called ultrasonic. This range can be divided into 5 types called low bass, upper bass, mid-range, upper mid-range, and treble.

For the low bass (20 to 80 Hz) includes the first two octaves. These low frequencies are associated with power and are typified by explosions, thunder, and the lowest notes of the organ, bass, tuba, and other instruments. Too much low bass results in a muddy sound [4].

For the upper bass, (80 to 320 Hz) includes the third and fourth octaves. Rhythm and support instruments such as the drum kit, cello, trombone, and bass use this range to provide a fullness or stable anchor to music. Too much upper bass results in a boomy sound [4].

For the mid-range, (320 to 2,560 Hz) it includes the fifth through seventh octaves. Much of the richness of instrumental sounds occur in this range, but if over-emphasized a tinny, fatiguing sound can be the result [4].

For the upper mid-range, (2,560 to 5,120 Hz) is the eighth octave. Our ear is very particular about sound in this range, which contributes much to the intelligibility of speech, the clarity of music, and the definition or "presence" of a sound. Too much upper mid-range is abrasive [4].

For the treble, (5,120 to 20,000 Hz) includes the ninth and tenth octaves. Frequencies in this range contribute to the brilliance or "air" of a sound, but can also emphasize noise [4].

The range of sound intensity (pressure) and the range of frequency to which the ear responds are remarkable indeed. The intensity ratio between the sounds that bring pain to our ears and the weakest sounds we can hear is more than 1 trillion (10^{12}). The frequency ratio between the highest and the lowest frequencies we can hear is nearly 10^3 (1000) times, or more than nine octave (each octave represents a doubling frequency) [1].

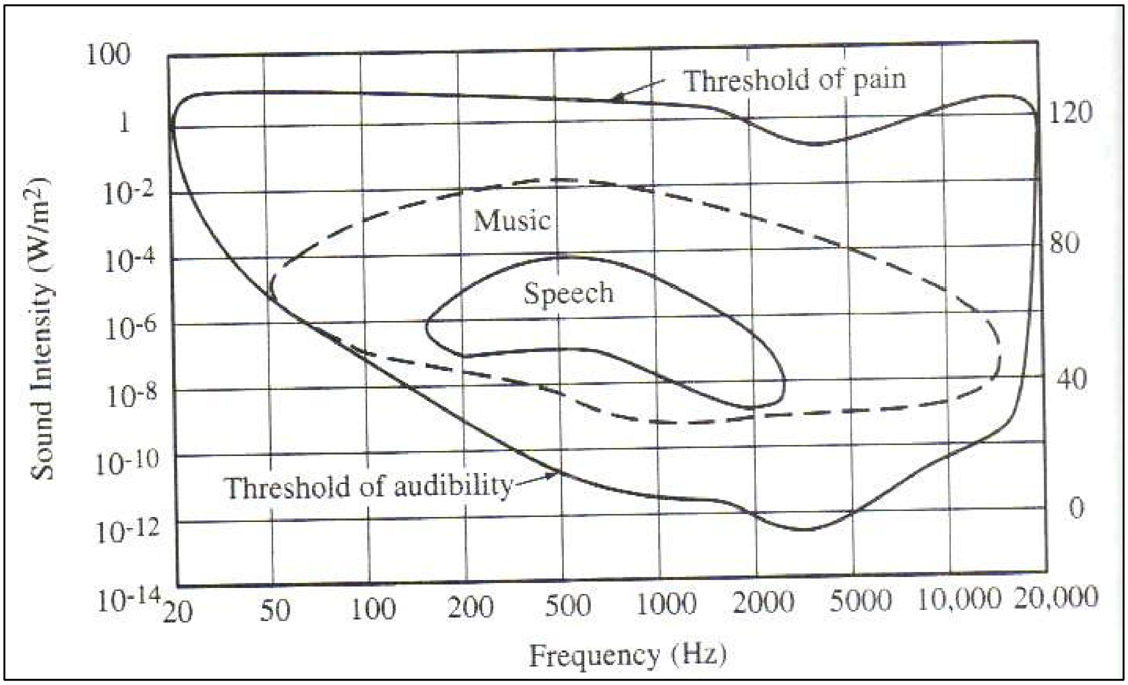


Figure 2.5: Range of frequencies and intensities to which auditory system responds [1]

From Figure 2.5 we can see how our auditory system responds to the intensity of sound. This figure is a result from experiment done by Fletcher and Munson (1933). The curve of threshold audibility demonstrates the relative intensity of the ear to sounds of low frequency at moderate to low intensity level.

2.4.2 Auditory System

As a human, our auditory system that is already known by everyone of us is our ears. Our ear can be divided into 3 parts which is first part is the outer ear, the second part is the middle ear, and the last part is the inner ear. Figure 2.6 below will show us what all that 3 parts are. This drawing is not to scale; for purposes of illustration, the middle ear and the inner ear has been enlarged.

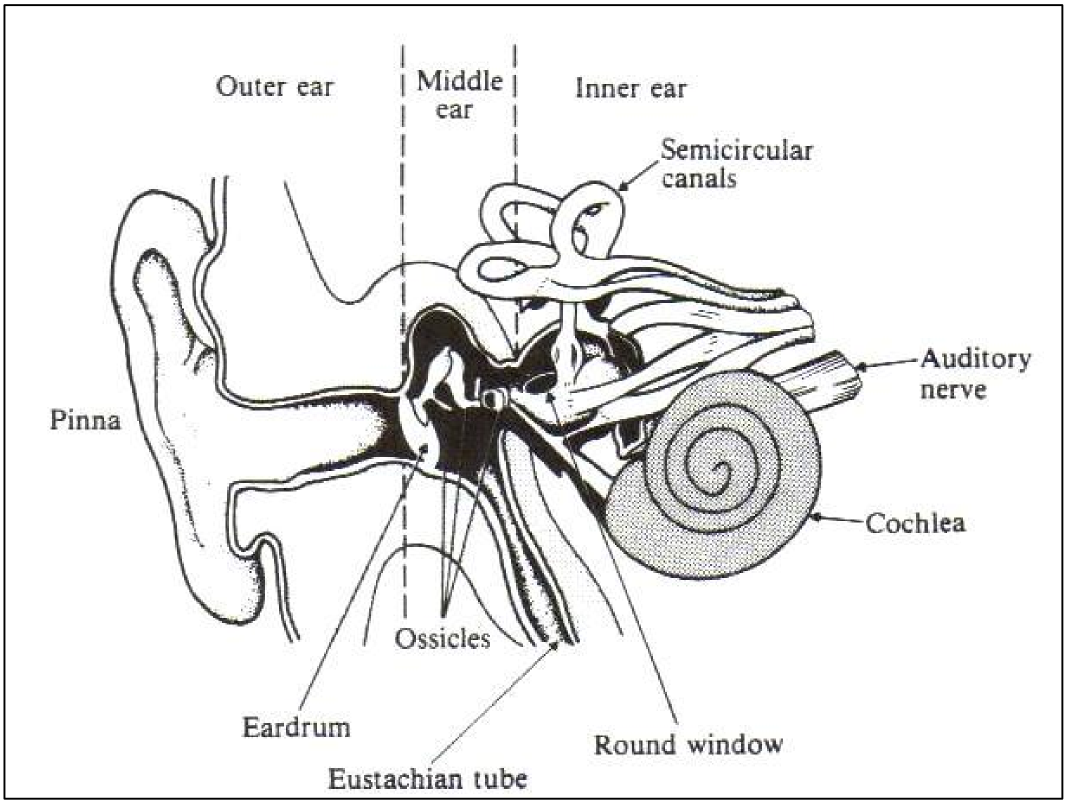


Figure 2.6: A schematic diagram of the ear, showing outer, middle, and inner regions

The outer ear consists of the external pinna and the auditory canal (meatus), which is terminated by the eardrum (tympanum). The pinna helps, to some extent, in collecting sound and contributes to our ability to determine the direction of origin of sounds of high frequency. The auditory canal acts as a pipe resonator that boosts hearing sensitivity in the range of 2000 to 5000 Hz.

The middle ear begins with eardrum, to which are attached three small bones (shaped like hammer, an anvil, and stirrup) called ossicles. The eardrum, which is composed of circular and radial fibers, is kept taut by the tensor tympani muscle. The eardrum changes the pressure a vibration of incoming sounds waves into mechanical vibration to be transmitted via the ossicles to the inner ear [1].

The function of ossicles is like a lever, which change the very small pressure to the much greater pressure up to 30 times. These 3 bones act like a mechanical transformer. The other function of ossicles is to protect the inner ear from very loud noises and sudden pressure changes. The responds to loud sounds is called acoustic reflex.

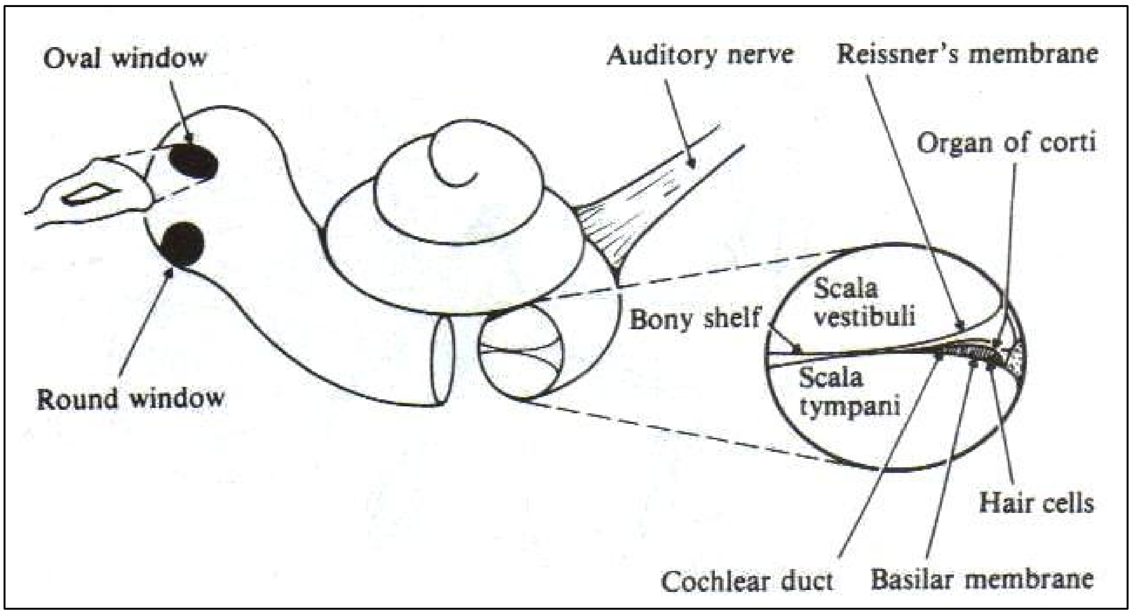


Figure 2.7: A schematic diagram of cochlea (inner ear) and a section cut of it

Inner ear as u can see from Figure 2.7 is the most complex part in the human auditory system. It contains the semicircular canals and the cochlea. Semicircular canals are for balancing of the body. It has nothing to do with the auditory system while cochlea is a crucial part of ear where it transforms pressure vibration to the properly coded neural impulse. Figure 2.8 below shows a schematic representation of the overall hearing mechanism.

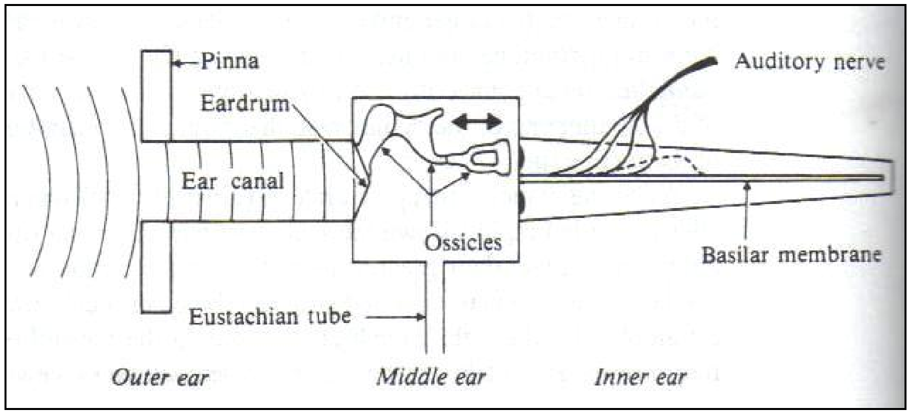


Figure 2.8: A schematic representation of the overall hearing mechanism [1].

2.4.3 Binaural Hearing and Localization

As a human we have been given 2 ears which we can have binaural hearing. The most benefit we can get from it is we can localize the source of sound. Binaural hearing really enhances our hearing ability by sense the direction of the sound. Lord Raleigh in 1876 performed an experiment and found that sounds of low frequency were more difficult to locate than those of high frequency [1].

2.5 Sound Pressure and Loudness

Most people might think it is imposible to measure the pressure and loudness of sound. However output signal of microphone usually is proportional to the sound pressure, thus sound can be measured with microphone and a voltmeter.

2.5.1 Decibels

Decibels scales are widely used to compare two quantities. We may express the power gain of an amplifier in decibels, or we may express the relative power of two sound sources. The decibels difference between two power levels, ΔL, is defined in term of their power ratio W_2/W_1:

$$\Delta L = L_2 - L_1 = 10\log\left({}^{W_2}/_{W_1}\right)$$

Although decibels scales always compare two quantities, one of these can be a fixed reference, in which case we can express another quantity in term of this reference. For example, we often express the sound power level of a source by using $W_0 = 10^{-12}$ W as a reference. Then the sound power level (in decibels) will be

$$L_W = 10 \log\left({}^{W}/_{W_0}\right)$$

2.5.2 Sound Intensity Level

Sound intensity level at a point some distance from the source can be expressed in decibels by comparing it to a reference intensity for which we generally use $I_0 = 10^{-12}$ W/m^2. Thus the sound intensity level at some location is defined as

$$L_I = 10\log\left({}^{I}/_{I_0}\right)$$

Even though sound power level and sound intensity level is measured in decibels but don't be confused with them. Sound power level is a power of a source of sound it self but sound intensity level is a measurement of sound at some point of distance from the source of sound. To further understand regardingt this, Figure 2.9 will help to show the different.

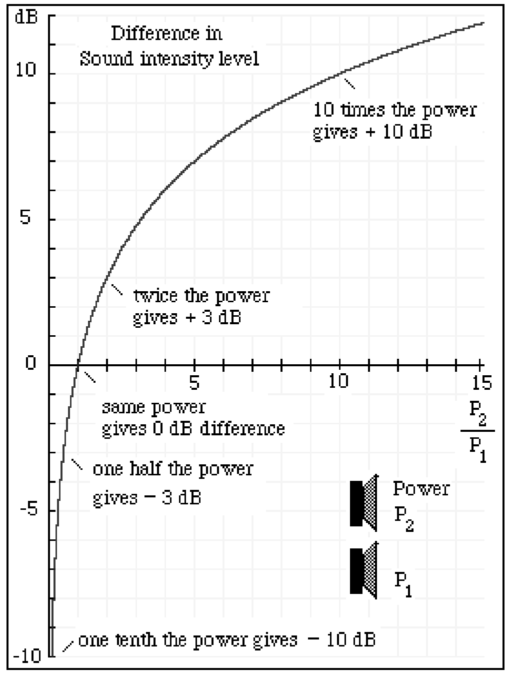


Figure 2.9: Graphically way to show sound intensity level measured in decibels

2.5.3 Sound Pressure Level

The intensity of a sound wave is proportional to the pressure squared. In other words, doubling the sound pressure quadruples the intensity. It can clearly be seen by this formula:

$$I = {p^2}/{\rho c}$$

I is the intensity of sound and p is a pressure where ρ is a density of air and c is a speed of sound. Sound pressure levels are measured by a sound-level meter, consisting of a microphone, an amplifier, and a meter that read in decibels. In the table below shows that the sound pressure levels of a number of sounds.

Table 2.2: Typical sound levels one might encounter

Jet takeoff (60 m)	120 dB	
Construction site	110 dB	Intolerable
Shout (1.5 m)	100 dB	
Heavy truck (15 m)	90 dB	Very noisy
Urban street	80 dB	
Automobile interior	70 dB	Noisy
Normal conversation (1 m)	60 dB	
Office, classroom	50 dB	Moderate
Living room	40 dB	
Bedroom at night	30 dB	Quite
Broadcast studio	20 dB	
Rustling leaves	10 dB	Barely audible
	0 dB	

Few questions might rise by looking to Table 2.2. Is it possible to have negative decibles? The answer to this question is; decibels are a ratio or difference between two quantities. So zero decibels means it is a reference level. In Table 2.2 the reference as one can see is a rustling leaves.

The sound level in dB is a measure (on a logarthmic scale) of the ratio of the sound pressure or sound intensity to this reference level. The logarithm of one is zero, so zero dB corresponds to the reference level. Numbers greater than one has positive logarithms, so positive a decibel means sound levels greater than that of the reference. Numbers smaller than 1 has negative logarithms, so negative decibel means sound levels below the reference level [5].

2.6 Speech Production

Of all creatures in this world, only human have the power of articulate speech. Speech is our chief means of communication. In addition, the human voice is our oldest musical instrument. The human voice and human ear are very well matched to each other. The ear has its' maximum sensitivity in the frequency range from 1000 to 4000 Hz, and that is the range of frequency in which the resonances of all vocal tracts occur.

2.6.1 The Vocal Organs

The aged thought that the only part of our body that related with our production of speech is our tongue and our mouth. The truth about production of speech, there is about more than ten parts of our body that contributing in production of speech. Figure 2.10 will show the overall vocal organs.

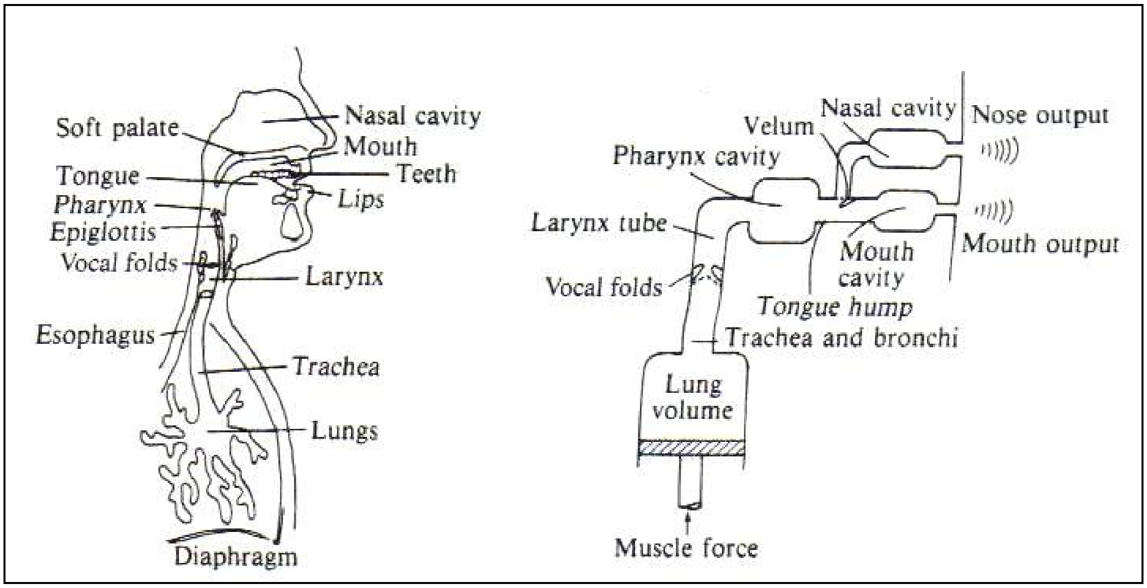


Figure 2.10: Human vocal organs and a representation of their main acoustical features.

The lungs serve as both reservoir of air and energy source. Whether in speaking or in exhaling, air is forced to lungs through larynx into three main cavities of the vocal tracts. These cavities are called pharynx, nasal and oral cavities. From the nasal and oral cavities, the air exits through the nose and mouth respectively.

In order to produce sound, the flow of air is interrupted by the vocal cords or by constriction in the vocal tract (made with tongue or lips). The sounds from the interrupted flow are appropriately modified by various cavities in the vocal tract and are eventually radiated as speech from the mouth and in some cases, the nose [1].

2.6.2 Articulation Speech

The articulation of English speech sounds also called phonemes is a basic thing to the speech. One or more phonemes combine to form a syllable and one or more syllable combine to form a word. It can be divided into two: vowels and consonants. Vowel sounds are produced with the vocal folds in vibration. Consonant may be either voiced or unvoiced. Table 2.3 will show some example of the vowels.

Table 2.3: The vowels of Great American English

Pure vowels					
ee	heat	/i/	aw	call	/ɔ/
i	hit	/ɪ/	ú	put	/ʊ/
e	head	/ϵ/	oo	cool	/u/
ae	had	/æ/	ŭ	ton	/ʌ/
uh	the	/ə/	er	bird	/ɜ/
ah	father	/ɑ/			

Various speech scientist lists from 12 to 21 different vowel sounds used in English language. This discrepancy in number comes about partly because of a difference of opinion as to what constitute a pure vowel sound rather than a diphthong. Figure2.11 below shows the approximate tongue position for articulating these vowels. Number 1-8 are the eight cardinal vowels, which serve as standard of comparison between languages [1].

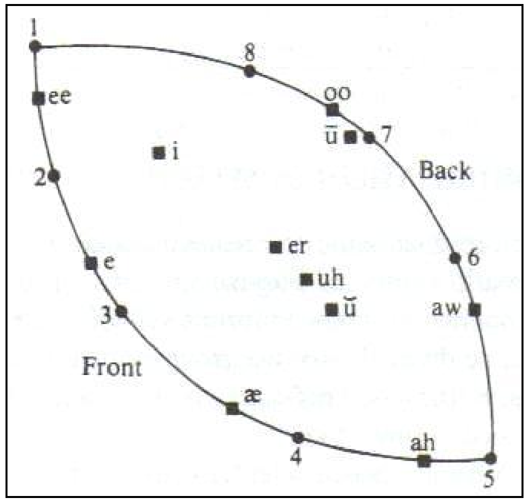


Figure 2.11: Approximate tongue positions for articulating vowels.

2.7 Equalizer

Equalizer (EQ) basiclly is a combination of many filters. These filters will be arranged according to the frequency band that it will able to control. There are many kind of equalizer that can be found in the market nowadays.

There are many kinds of EQ. Each one has a different pattern of attenuation or boost. A peaking equaliser raises or lowers a range of frequencies around a central point in a bell shape. A peaking equalizer with controls to adjust the level (Gain), bandwidth (Q) and center frequency is called a parametric equalizer. If there is no control for the bandwidth (it is fixed by the designer) then it is called a quasi-parametric or semi-parametric equalizer.

A pass filter attenuates either high or low frequencies while allowing other frequencies to pass unfiltered. A high-pass filter modifies a signal only by taking out low frequencies; a low-pass filter only modifies the audio signal by taking out high frequencies. A pass filter is described by its cut-off point and slope. The cut-off point is the frequency where high or low-frequencies will be removed. The slope, given in decibels per octave, describes how quickly the filter attenuates frequencies past the cut-off point. A band-pass filter is simply a combination of one high-pass filter and one low-pass filter which together allow only a band of frequencies to pass, attenuating both high and low frequencies past certain cut-off points.

Shelving-type equalizers increase or attenuate the level of a wide range of frequencies by a fixed amount. A low shelf will affect low frequencies up to a certain point and then above that point will have little effect. A high shelf affects the level of high frequencies, while below a certain point, the low frequencies are unaffected.

One common type of equalizer is the graphic equalizer, which consists of a bank of sliders for boosting and cutting different bands (or frequencies ranges) of sound. Normally, these bands are tight enough to give at least 3 dB or 6 dB maximum effects for neighboring bands, and cover the range from around 20 Hz to 20 kHz (which is approximately the range of human hearing). A simple equalizer might have bands at 20 Hz, 200 Hz, 2 kHz and 20 kHz, and might be referred to as a 4-band equalizer. A typical equalizer for live sound reinforcement might have as many as 24 or 31 bands. A typical 31-band equalizer is also called a 1/3-octave equalizer because the center frequencies of sliders are spaced one third of an octave apart [2].

2.7.1 Filter

To be able to construct equalizer, the most important part is the filter. The basic rule for equalizer is using filters. Filters can be divided to 4 types. The first one is high pass filter. The simple high pass filter is as shown in Figure 2.12.

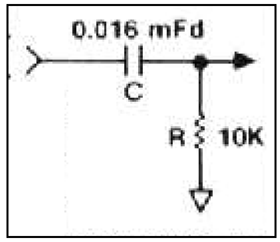


Figure 2.12: High Pass Filter

At some specified frequency, the capacitor's impedance is equal to that of the resistor, and it is here that the response will be 3 dB down. If this frequency is 1 KHz, then the filter is referred to as a 1 KHz high-pass [6]. It attenuates or rejects all frequency below f_c or in this case we took 1 KHz as example.

The second one is called Low pass filter. It is significantly different with high pass filter. It’s constructed by transposing the capacitor and resistor place.

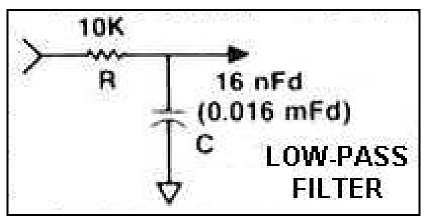


Figure 2.13: Low Pass Filter

From Figure 2.13; it shows how the passive low pass filter been constructed. It will allow a lower frequency than the f_c and the example here is 1 KHz.

The third one is bandpass filter. To make it easier to understand, bandpass filter consist of low pass filter and high pass filter which low pass filter has higher value of critical frequency, f_c and the high pass filter has a lower one. It can be constructed by cascading the two filters with the specification mentioned earlier.

The last one is a bandstop filter. Just like bandpass filter, this kind of filter also constructed by cascading the low pass and high pass filter but the characteristic will be different in which the critical frequency of high pass filter will be higher than low pass filter.

All the four filters mentioned are passive filter. Active filter can be achieved by using an active element. One example is to apply the op-amp 741. The op-amp provides gain, so the signal is not attenuated as it passes through the filters. The high impedance of the op-amp prevent excessive loading driving source, and the low output impedance of the op-amp prevent the filter from being effected by the load that it is driving. It's also easy to adjust.

The designing process of filter will be easier with the help of op-amp. By adding the op-amp the output can be boosted, unlike the passive filter, the highest gain for passive is unity gain (1). Nevertheless, the circuit can be designed as desired.

CHAPTER 3

METHODOLOGY

3.1 Basic Idea

This project basically stress on improving the quality of the sound system which is used during the public address system. There are so many devices that can be used to archive this target. One of the devices is called equalizer. Equalizer is an audio component used to flatten the system spectral response in the audio signal band or to produce other desirable effects. In the amplification and broadcast of music or other performances, either live or from recordings, the tonal content of the broadcast audio program can be distorted by frequency dependent attenuation or reinforcement from characteristics of the room, concert hall, speaker system, or other factors affecting the sound. The job of an equalizer is t treat all the sound equally thus the output will be same as the input.

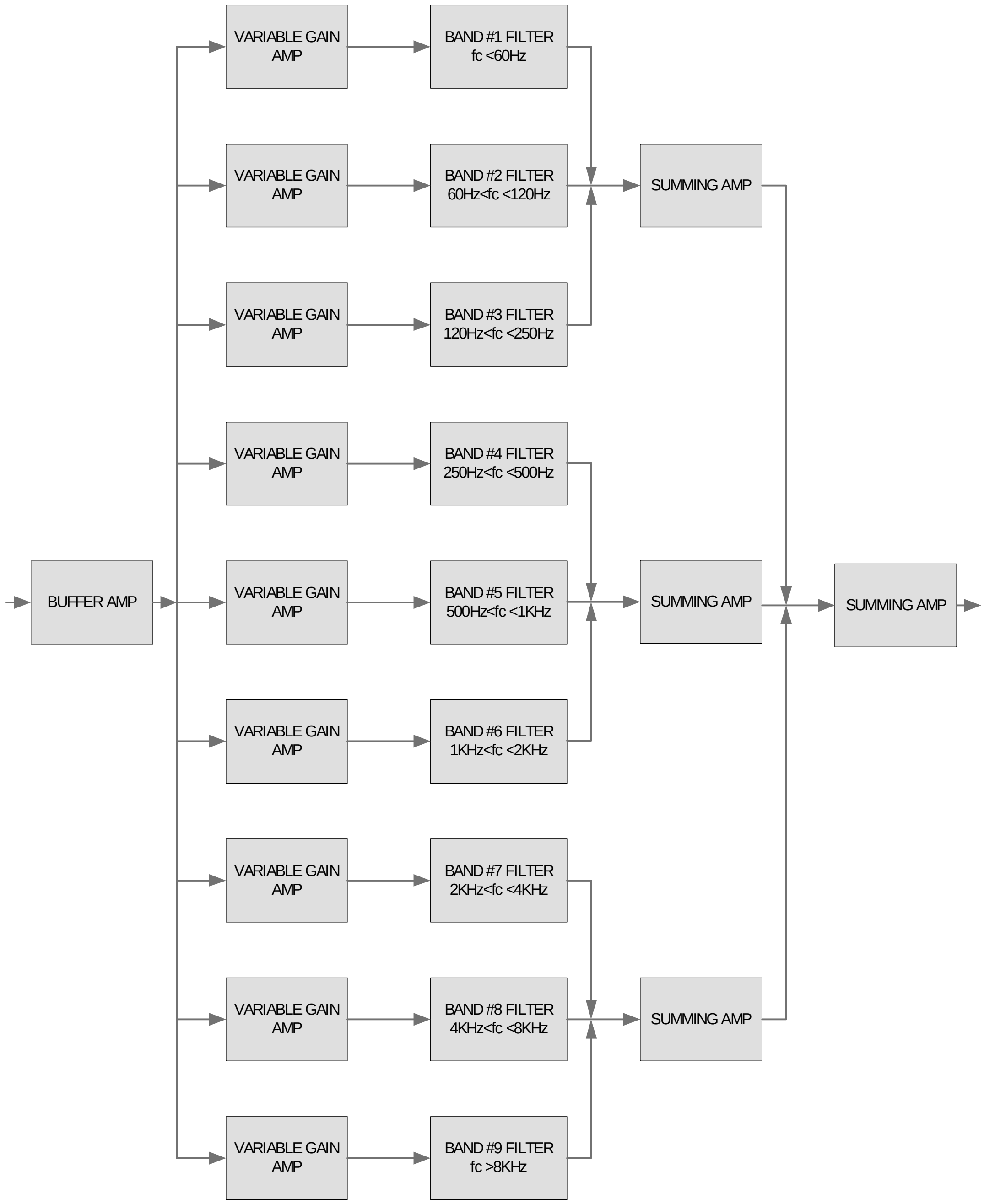


Figure 3.1 Block diagram of the equalizer [7]

Figure 3.1 shows the block diagram of the equalizer which is designed in this project. However, the power supply must be built. The purpose of the power supply is to supply +15 V and -15 V for the LM741. Figure 3.2 indicates the design of the power supply.

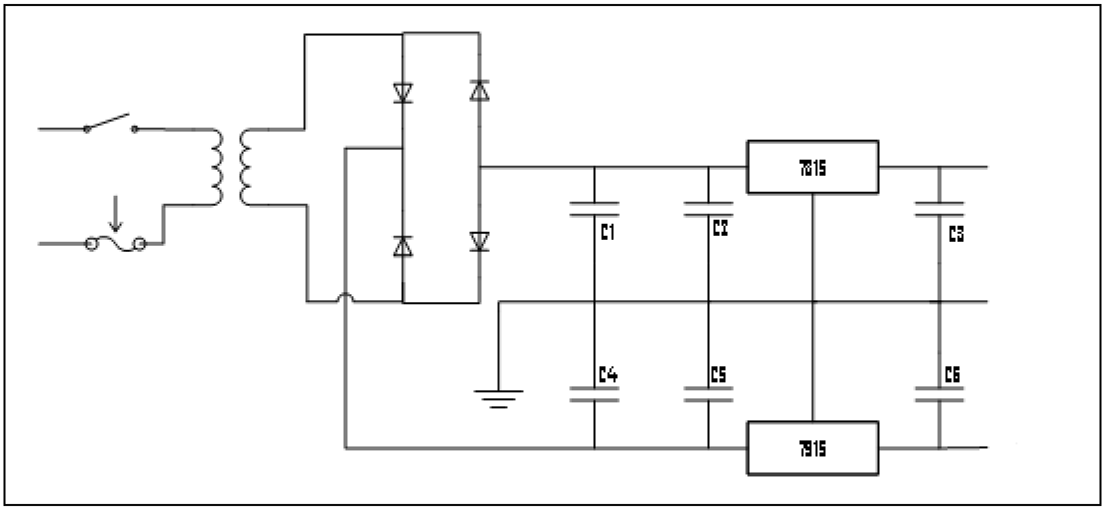


Figure 3.2: The design of power supply

From the Figure 3.2, 7815 is an IC that will supply +15 V and 7915, also an IC that will have an output of -15 V. The input reading will be 240 Vrms from the household line voltage. Fuse will protect the equalizer circuits. Transformer T1 steps down the alternating household line voltage to a lower alternating voltage. The diode bridge, which consists of diodes D1-D4, rectifies the alternating low voltage into a pulsating DC voltage. Capacitors C1 and C4 smooth out the pulsating DC voltage into a DC voltage with a ripple or AC component. The AC component of the DC voltage is reduced positive voltage regulator of U1 and negative voltage regulator U2. Capacitors C2 and C5 are required if the voltage regulator are located more than a few inches from capacitor C1 and C4. Capacitors C3 and C6 improve the transient responses of the positive and negative voltage regulator, respectively [7].

3.2 Buffer Amplifier

Buffer amplifier will provide electrical impedance transformation from one circuit to another circuit. By having high output impedance and low input impedance it prevent the second circuit from loading the first circuit unacceptably and interfering with it desired operation.

In this particular project, this equalizer will receive input from the mixer and in between of mixer and equalizer there will be buffer amplifier. Figure 3.3 below simplifies the above description.

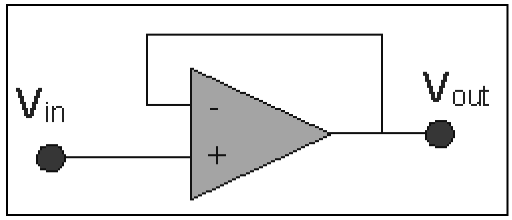


Figure 3.3 Buffer amplifier

The circuit essentially acts as a copy - at the output - of the input voltage, $V_{\mathbf{in}}$. It does that without drawing any current from wherever the input voltage terminal is attached. However, at the output terminal, whatever amount of current the operational amplifier can supply can be drew. In this project the current that operational amplifier can supply is almost 1 A.

3.3 Operational Amplifier

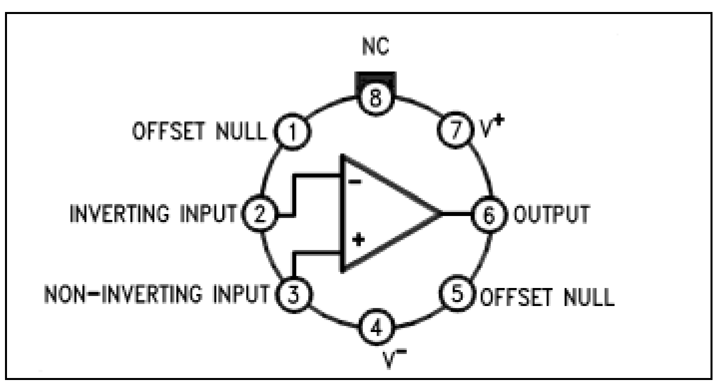


Figure 3.4 LM741 connection diagram

Operational amplifier (Op-amp) is a device use to be integrator, summing amplifier, and general feedback application. LM741 as u can see from the Figure 3.4 is a basic Op-amp in the market. Because the lack of space, this could not be translated in this project.

As a result quad operational amplifiers are used. The chosen quad operational amplifiers are LM324. It is significantly lower in cost and can be operated at the supply voltage vary from 3 - 32 V. The common mode input range also includes the negative supply. The pin connections for the LM324 are shown in Figure 3.5 below.

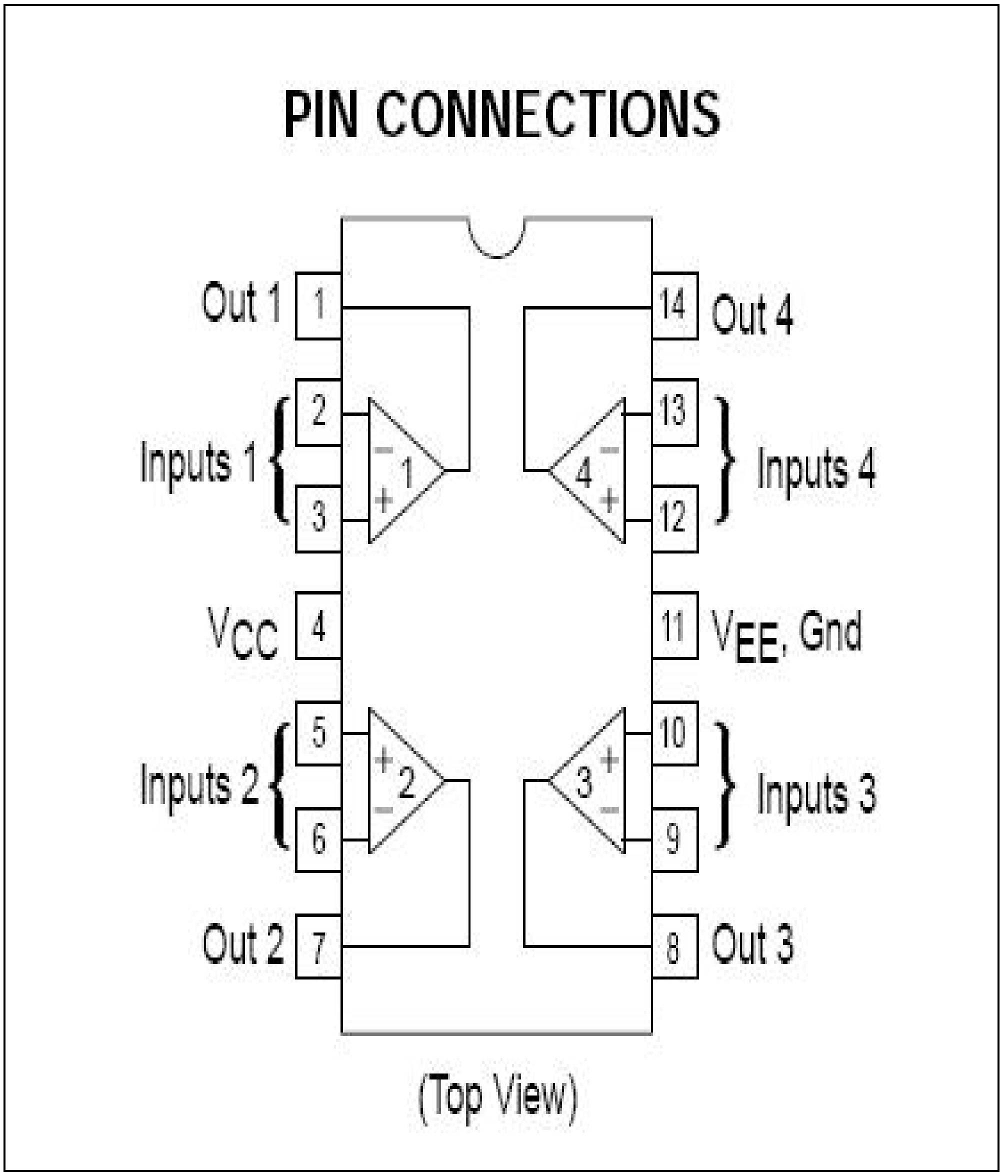


Figure 3.5 Pin connection for the LM324

3.4 Variable Gain Amplifier

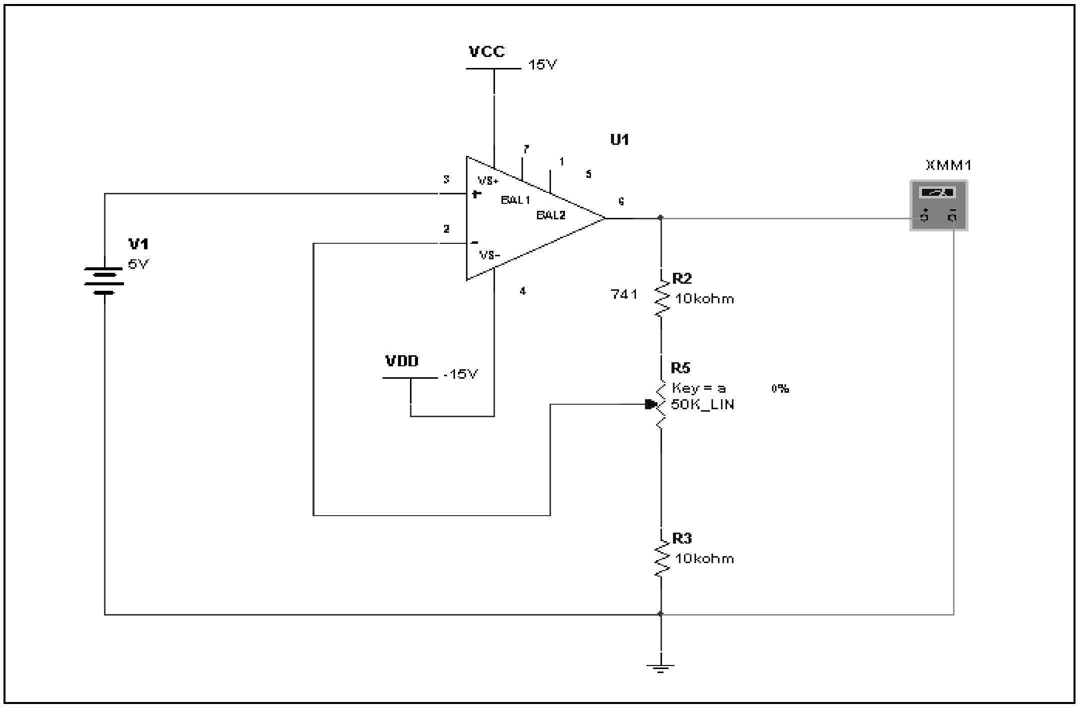


Figure 3.6 Variable Gain Amplifier simulation using Multisim

Figure 3.6 above shows the connection of the variable gain amplifier. This connection has their input from the buffer amplifier and the output of this connection will go straightly to the filter. This variable gain amplifier will determine the value of gain in each frequency band that we desire.

This circuit is a main circuit to control the frequency response of output of equalizer. Even there is other circuit as important as variable gain amp but with this circuit we can control the gain which is the main usage of the equalizer.

3.5 Filter Design

Equalizer made from combination from types of filter. First thing to be considered before making the equalizer is what is the range of frequency that desired to be controlled. In this project the range of frequency are; 60 Hz, 60Hz$<f_c<$120 Hz, 120Hz<f_c< 250 Hz, 250Hz<f_c<500 Hz, 500Hz<f_c<1 kHz, 1 kHz<f_c<2 kHz, 2 kHz<f_c<4 kHz and 4 kHz<f_c<8 kHz.

The first frequency to be controlled is going to use high pass filter. Bear in mind that the arrangement of the frequency is in ascending way. This project will only have an octave frequency which meant only one frequency will be covered in one octave.

Filters can be divided into 4.

a) Low pass filter
b) High pass filter
c) Band pass filter (combination of high pass and low pass filter)
d) Band stop filter (will not be use in this project)

First frequency that to be controlled the gain will use low pass filter. Low pass filter as describes in the Figure 3.7, is a filter that only let frequency that tower than the frequency of the filter. The arrangement of the low pass filter can be seen at Figure 18.

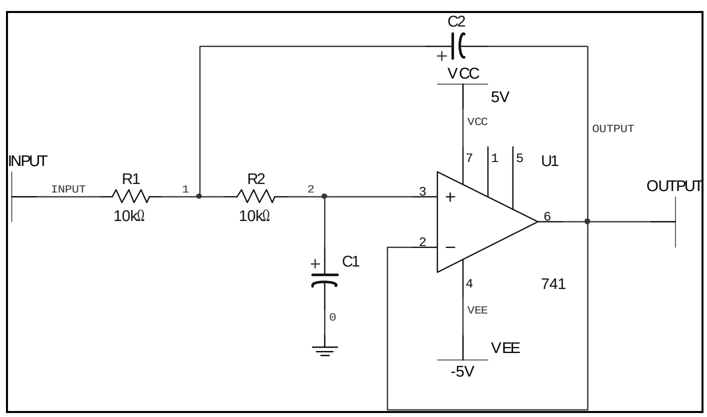


Figure 3.7: Low pass filter design

For the second until the eight bands, band pass filter will be used. As mentioned before, band pass is a combination of a high pass and low pass filter. So, it only let a frequency in the middle of low pass and high pass filter to get through them. The design of this type of filter can be see in the Figure 3.8.

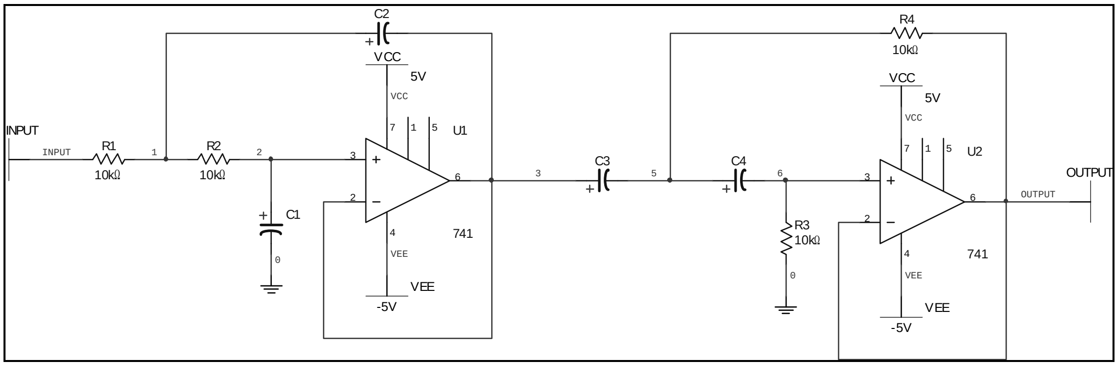


Figure 3.8 The arrangement of band pass filter

By comparison between low pass filter and band pass filter one easily differentiate how the high pass filter arrangement.

3.6 Summing Amplifier

As nine bands represent nine frequencies that can be controlled and only one output sufficient for each channel (one for the right channel and one for the left channel). Summing amplifier can be used to satisfy the need of this project.

For the summing amplifier the gain of high pass should be considered, band pass and the low pass filters. For the low and high pass filters, the gain is a unity gain thus the input and the output will merely the same value. But for the band pass filters, the gain is 1/3. So the value for the output will only be 1/3 the value of input.

3.7 Simulation

In this project there are two things that will be used. Firstly, simulation to forecast the result and secondly is analyzer to see the result and testing the complete hardware.

3.7.1 Multisim

Simulation processes are adapted at the beginning of this project by Multism software. Multisim was choose because it is easier has wide variety of component. The versions used in this project are Multisim 8 Power Pro.

The only problem faced is to forecast the result in this software for filter design. Although Multisim did have spectrum analyzer, up to the end of this project I did not know how to use that spectrum analyzer.

3.7.2 Analyzer

To achieve result we have to use spectrum analyzer. Unfortunately, in the laboratory we only have spectrum analyzer that will respond frequency range 150 kHz to 2 MHz. In order to solve this problem, Real Time Analyzer (RTA) was used. Real Time Analyzer (RTA), which is a spectrum analyzer that will respond to the audio spectrum (20 Hz – 50 kHz) known as TrueRTA. This RTA will be discussed later on the next chapter.

3.8 Physical

Three mm plywood is used as body cover. For the front panel 0.5 cm woods is used. There will be 10 holes in the front panel of equalizer used for knob, which represent the parameters.

Figure 3.9 : The physical look of the equalizer

Figure 3.9 shows the plan on how this equalizer will look alike from the front panel. The dimension for this box is 30cmx35cmx10cm.

CHAPTER 4

RESULT AND ANALYSIS

4.1 TrueRTA [8]

Back in the late 1970's audio engineers started using Real Time Analyzers, or RTAs, to provide a live display showing the frequency spectrum of audio signals. These early analyzers worked by using a collection of electronic bandpass filters. Each band was typically one-octave wide and the bands were distributed on center frequencies spaced in one-octave intervals.

The most popular octave center frequencies are: 62.5 Hz, 125 Hz, 250 Hz, 500 Hz, 1 kHz, 2 kHz, 4 kHz, 8 kHz and 16 kHz. One full range audio signal was input to the bank of bandpass filters with the output of each filter representing the signal level for that frequency band. Then, the output of each filter would be fed into something equivalent to a VU meter, usually an LED display, to show the loudness of the signal in each frequency band. By placing the level displays side by side they formed a live graph of the audio signal and we were able to see the bass versus the treble energy, or perhaps a peak in the midrange all in real time as the music played.

In those early days of RTAs the popular analyzers had an LED display with a bar graph that displayed 10 bars as an octave equalizer. Each bar was driven by the output of one of the bandpass filters. In search of better resolution, engineers then divided the frequency bands into even more narrow bands. From one-octave frequency bands we advanced to 1/3-octave bands, going from 10 bands to 30 bands across the audible spectrum. Today even finer bands are used for audio analysis.

More recently, digital technology has given us a new way to achieve the same results. Today, once we have a signal in the digital domain, such as a PC with a sound card, we can bring to bear the awesome power of Digital Signal Processing or DSP. Modern personal computers now have enough power to do a significant amount of processing of live audio signals.

By using powerful DSP methods, True***RTA*** can show you up to 1/24th octave frequency bands for a total of about 240 bands spanning the full audio spectrum! The display is very smooth and reveals even the finest detail of the program material under analysis, whether it is music or a test tone.

But in this project I will use only 1 octave of TrueRTA since this is the free version and my project is an octave equalizer. Figure 4.1 shows the front panel for this analyzer.

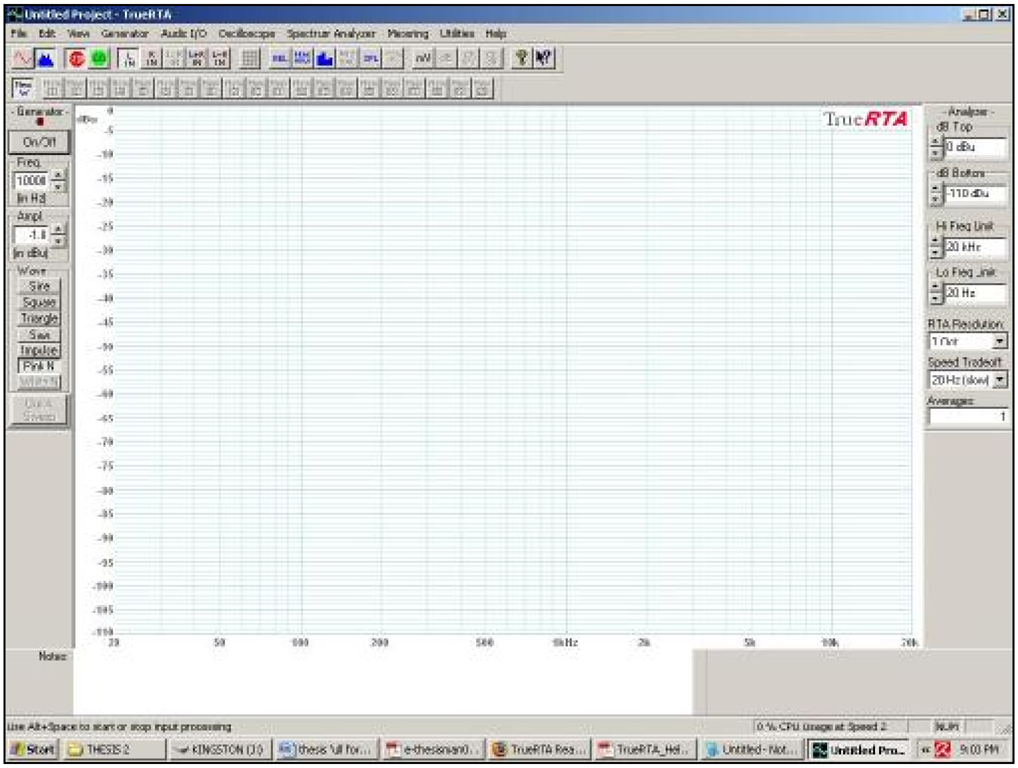


Figure 4.1 Front Panel of TrueRTA

This TrueRTA also has its own built in generator. This generator can also generate signal on some freqeuncy with some amplitude. This Figure 4.2 below will make the usage of this generator easier to understand.

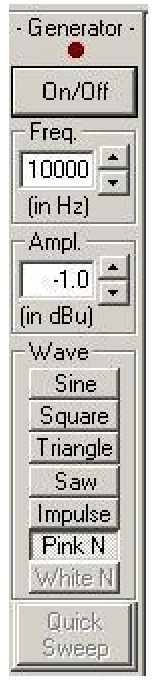


Figure 4.2 Generator panel on the left side of the TrueRTA

Figure 4.2 shows the panel for generator. From this panel we can see the on off button. The red dot shows us whether the generator is switch on or not. The frequency (Freq.) shows the frequency of the signal that we try to generate in Hertz unit. The amplitude can also be set in the Ampl section together with the type of signal that we want.

Before you can use TrueRTA productively, you will need to understand how to use the Windows audio mixers (and perhaps your sound system's custom mixers) in order to select among the various input signals that are available to your PC sound system. The signal that serves as the input to the analyzer is selected at the Windows Record Control mixer (or the Record section of your custom mixer). The descriptions below show how to access the Windows Record Control to select the desired signal source:

1) Open the Windows Volume Control:

· From the Windows task bar, right click on Volume and select "Open

Volume Controls" (Alternately, from the Start button select: Programs/Accessories/Entertainment/Volume Control.)

2) Switch to the Recording Mixer:

· From the Volume Control's Options menu select Properties.

· At the Properties window's "Adjust volume for" section select "Recording".

· Click on the OK button to close the window and bring up the Record Control mixer.

The typical input signal selections that is available at the Recording Mixer are microphone input selection, line input selection, wave signal selection, CD audio signal selection, and "what u hear" signal selection.

4.1.1 Microphone Input Selection

At the Record Control mixer select the Microphone input when you want to analyze the signal from a microphone plugged into the sound system's "Mic In" jack. The fader for the mic input also has to make sure to be raised. Inexpensive multimedia microphones (such as those built into many laptop computers) are usually accurate enough for the casual analysis of live sound sources. (If you hear the mic through your speakers then lower the speaker volume at the task bar.) For precision acoustical analysis you will most likely not be using the PC's mic input but rather you will use a professional mic and microphone preamplifier to feed the PC's "Line Input" jack.

4.1.2 Line Input Selection

To use the Line Input, Line-In at the Windows Record Control mixer and raise the Line-In fader need to be selected. For precise acoustical measurements the PC's Line

Input must be feed from the output of a microphone preamplifier used with a calibrated measurement microphone. The line input will also be used for most electronic testing where the output from the unit under test is connected to the PC’s “Line In” jack. Power amps should probably be connected to the Line-In only by using an in-line attenuator of at least 20 dB. (10 k Ohm series resistor with 1 k Ohm across the Line-In jack)

4.1.3 Wave Signal Selection

If you are analyzing a signal that originates within the computer (such as a .wav file or the direct digital output of TrueRTA’s signal generator) you will need to select “Wave” at the Windows Record Control mixer and raise the Wave fader.

4.1.4 CD Audio Signal Selection

Audio can directly be analyzed from a CD by selecting CD Audio at the Windows Record Control mixer and raising the CD Audio fader to maximum. Depending on how the computer is configured, CD Digital can also be selected and use the direct digital output of the CD as your input signal for analysis.

4.1.5 “What U Hear” Signal Selection

When the analyzer input want to be heard the same as whatever you are listening discard the calibration, “What U Hear” mode can be selected at the Windows Record Control mixer and raise the “What U Hear” fader. Then the input to the analyzer will be the same signal as selected at the Windows Play Control mixer.

4.2 Pink Noise Analysis

From the TrueRTA we can generate the pink noise signal generator. The signal of pink noise treat all the frequencies same. In other words, the amplitude of all frequency will be same or will be significantly to each other.

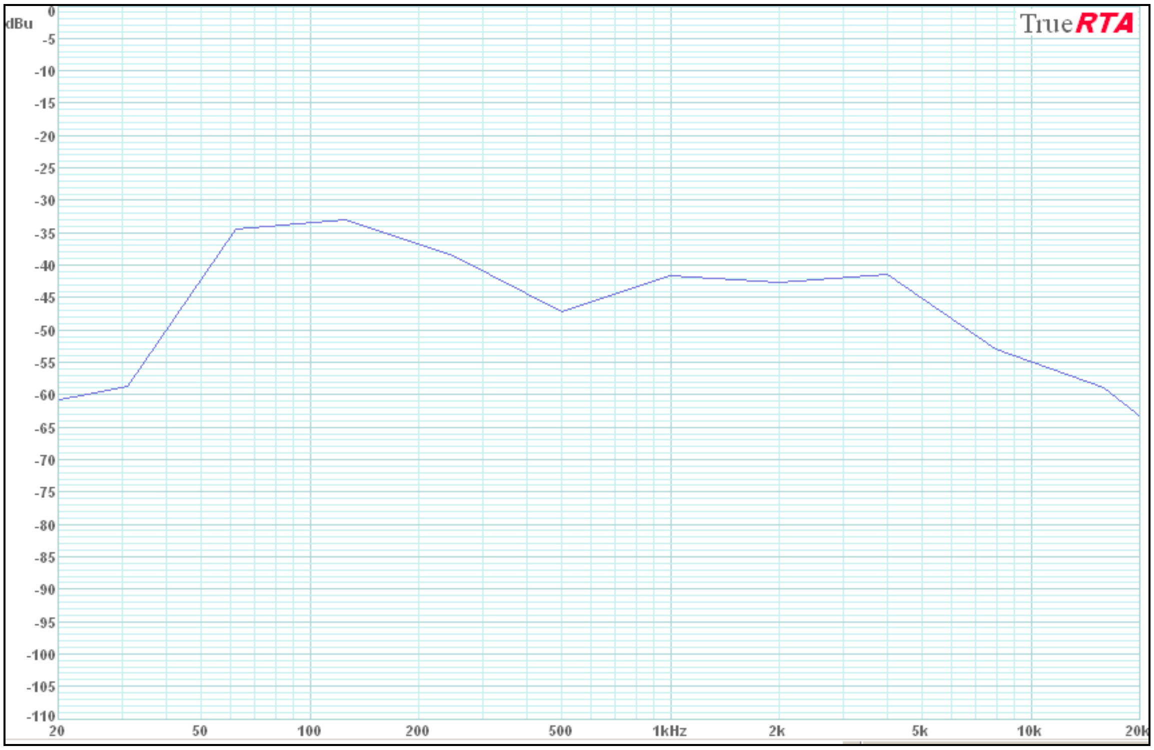


Figure 4.3: Pink noise signal

Figure 4.3 shows the pink signal noise before any attempt to increase the gain of the equalizer, or in other words, all equalizer parameter is set to the minimum. Thus, will make a unity gain. In order to show that this equalizer able to control a wide range of frequency gain (from 60 Hz to 8 kHz) we must set the parameter of frequency band to the maximum 1 by 1.

The purpose of equalizer is not only to gain, but in some condition such as when feedback occurs we have to cut the gain so the gain at frequency which feedback occurs is low and people cannot hear it enough.

4.3 60 Hz - 120 Hz Analysis

This paragrapht will show how the second frequency band (it will be 3rd knob in the front panel of the equalizer, refer to apendix A for more information). We only show the result for the second frequency band since this TrueRTA only produce pink noise in frequency range from 50 – 5 kHz (this is an approximate based on inspection method). The result for this analysis can be seen from the Figure 4.4.

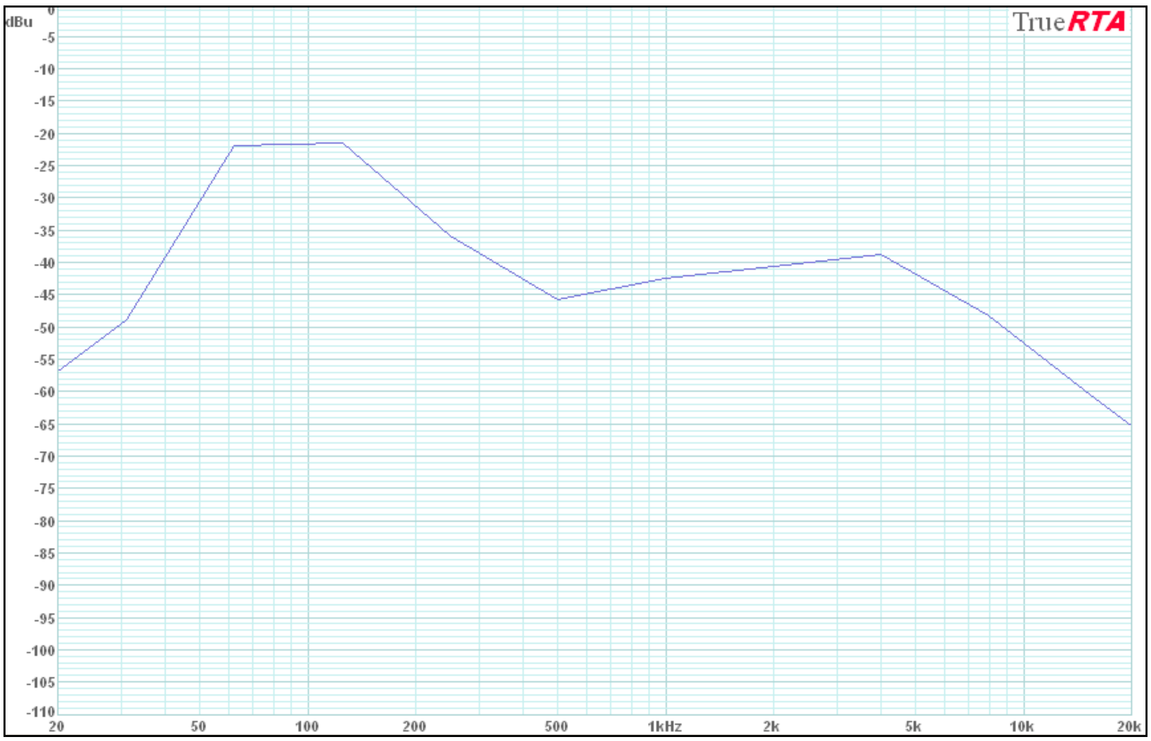


Figure 4.4: Analysis on second band

Since this analyzer is based on octave, the first band didn't really give an effect to the result. The bell curve need to considered, which the change in gain at some frequency also has an effect to the nearby frequency.

From the response obtained, it can be compared with the original pink noise. The gain is somehow close to 15dB.

To know how to chage gain to dB gain refer to sub topic 2.5.1 The discussion about decibels is in general. But in that sub topic the formula that given is the difference between 2 powers, however, in this case it should be changed to the gain.

$$L_W = 10\ log\, {}^{W}/_{W_0}$$

In this case the power ratio can be changed to the gain ratio. W_0 is set as the reference or 1.17 since the output of the equalizer if all the parameter set to minimum is 1.17. For the ratio, we let W = 36 since the mazimum output for every each of band is 6 and for the volume we also set it to 6. The gain for cascaded amplifier is a product multiply of those amplifier gains.

$$L_W = 10\ \log {}^{36}/_{1.17}\ dB$$

$$L_W = 14.88\ \approx 15\ dB$$

The only thing that will affect the result is the room condition and surrounding noise. This error occurs because TrueRTA will capture any sound. In UMP there is no room that suitable for us to get the perfect result.

4.4 120 Hz – 250 Hz Analysis

The second octave or frequency band of my equalizer will be 120 – 250 Hz. To see or to analyze the result we must maintain the setting in section 4.2 and set the parameter for the third band to the maximum. The result for this analysis can be seen from the Figure 4.5.

Figure 4.5: Analysis on third band

Just like the analysis before there is gain for the third band but due to the bell curve and the limitation of this TrueRTA the frequency response did not give an ideal result. This thing can be solved by using the higher precision of analyzer such as 1/3 octave or more.

4.5 250 Hz – 500 Hz Analysis

As the analysis goes to the fourth band the setting will be exactly the same as before. The entire band is remained as exactly as before after that changes the parameter for the fourth band until the maximum. The result that we will get is like the Figure 4.6 below.

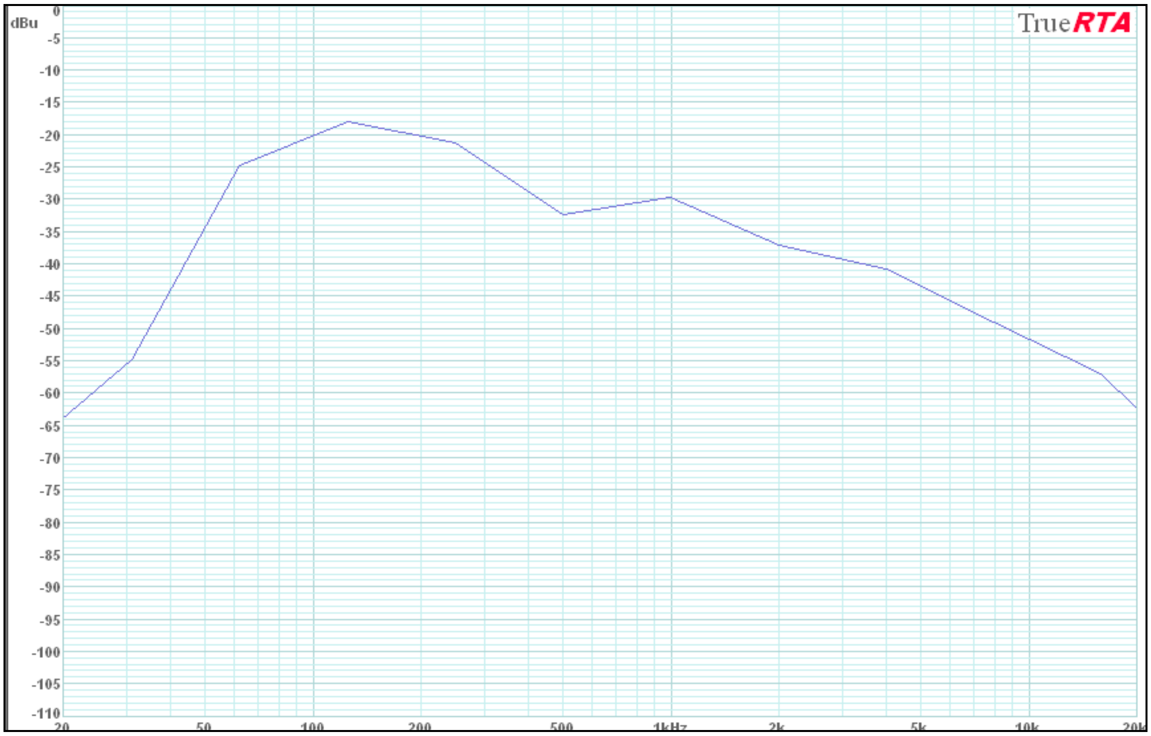


Figure 4.6: Analysis on fourth band

There is a difference between Figure 4.5 and Figure 4.6. The difference is at the frequency between 200 – 500 Hz. This result is based on the type of band that used. Even though there was an error, but the caused of all the errors has been discuss in the earlier chapter.

4.6 500 Hz – 1000 Hz Analysis

There will be no different in setting the parameter between this analysis and the previous one only for the fifth band; we set it to the maximum. We have to compare the result here with the previous one to know whats the different and what this fifth band can really do. The result for this analysis can be seen from the Figure 4.7.

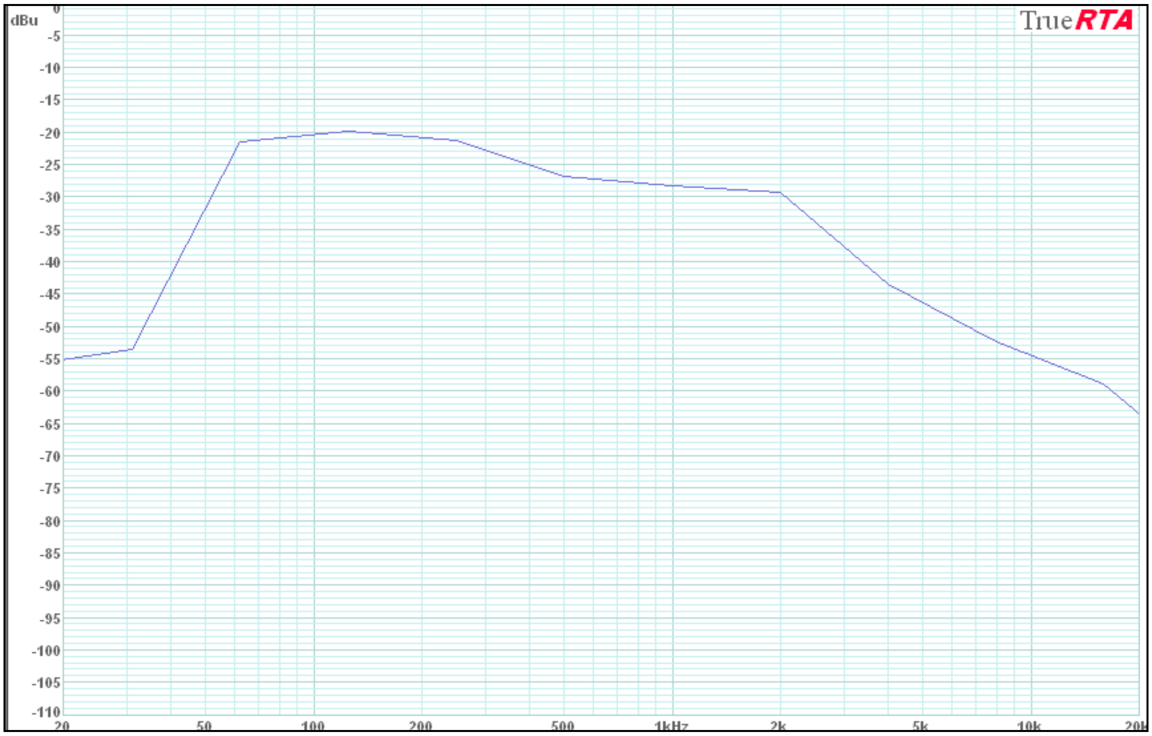


Figure 4.7: Analysis on fifth band

From the Figure, it shows how the bell curve affects the result. The frequency response that we get on the 500 – 1 kHz is something that we expected. According to the formula back in section 2.5.1, we can get the expected result and the real result is similar. Because of the volume is already set to the maximum on the previous analysis so the gain for this will be 6. If we use the formula we get 7.78 dB for the gain and according to the result that we get is 6 dB.

4.7 1000 Hz – 2000 Hz Analysis

In this analysis we set all the parameter to the maximum except for the seventh, eighth, and ninth band are set to minimum. We also expect that there will be visible changes on the frequency response. The result for this analysis can be seen from the Figure 4.8.

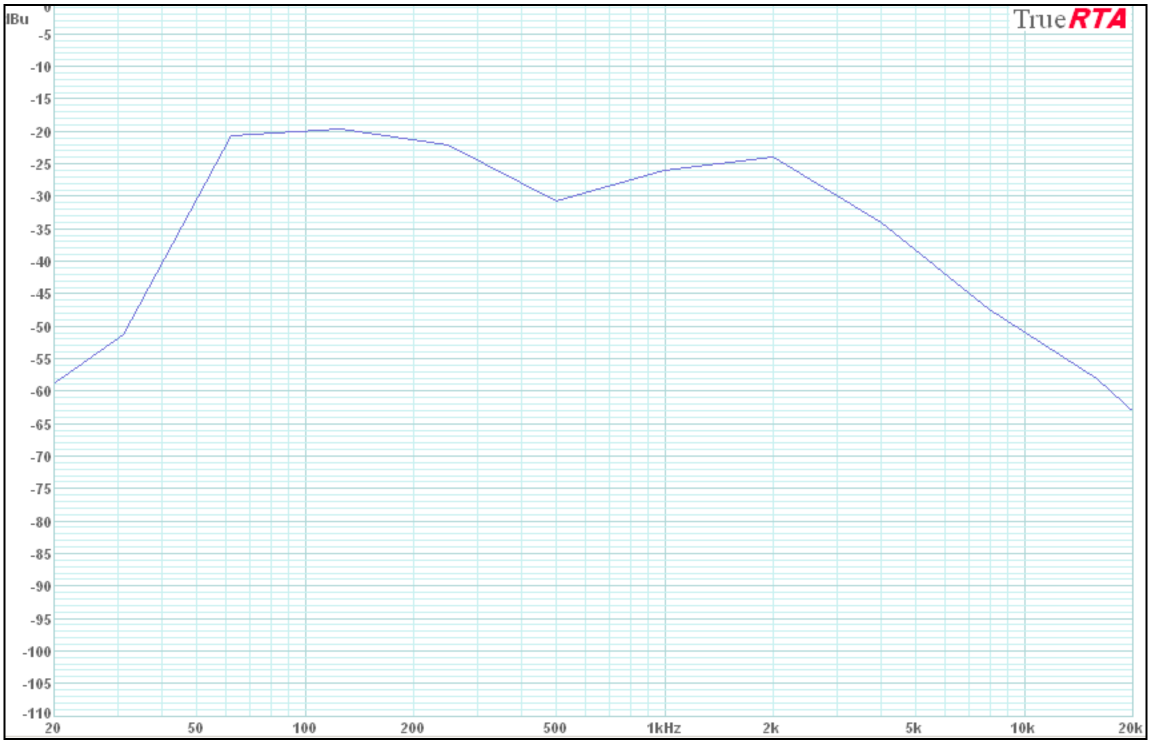


Figure 4.8: Analysis on sixth band

As we can see the gain in dB for the selected band is around 6 dB. As the same as before this happen resulting on the effect of bell curve and the limitation of the analyzer.

4.8 2000 Hz – 4000 Hz Analysis

The setting for the analysis for the sixth band basicly same as before except for the parameter for seventh is set to maximum and the remaining parameter is same as previously analysis. This band are crucial since in this frequency lies all the inteligence of the speech. Human frequencies of speech only occur around here. The result for this analysis can be seen from the Figure 4.9.

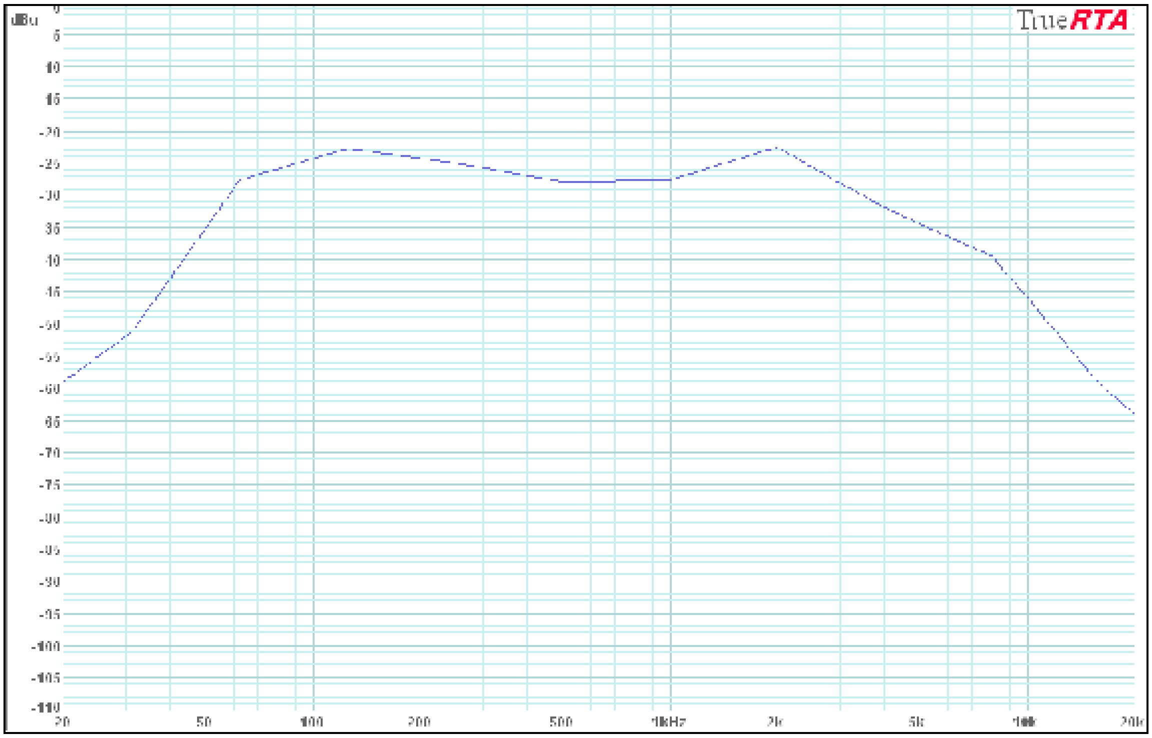


Figure 4.9: Analysis on seventh band

For this analysis we can see much the different on the shape because the range of 2000 Hz to 40000 Hz is not included in octave of the TrueRTA. The only things that we can see the effect of the bell curve for it. But the changes can be seen obviously.

4.9 4000 Hz – 8000 Hz Analysis

We can again doing the same procedure like the previous analysis but with the parameter for the eighth that are going to change. We set the parameter for eighth band to maximum and for the other remaining like the previous analysis.

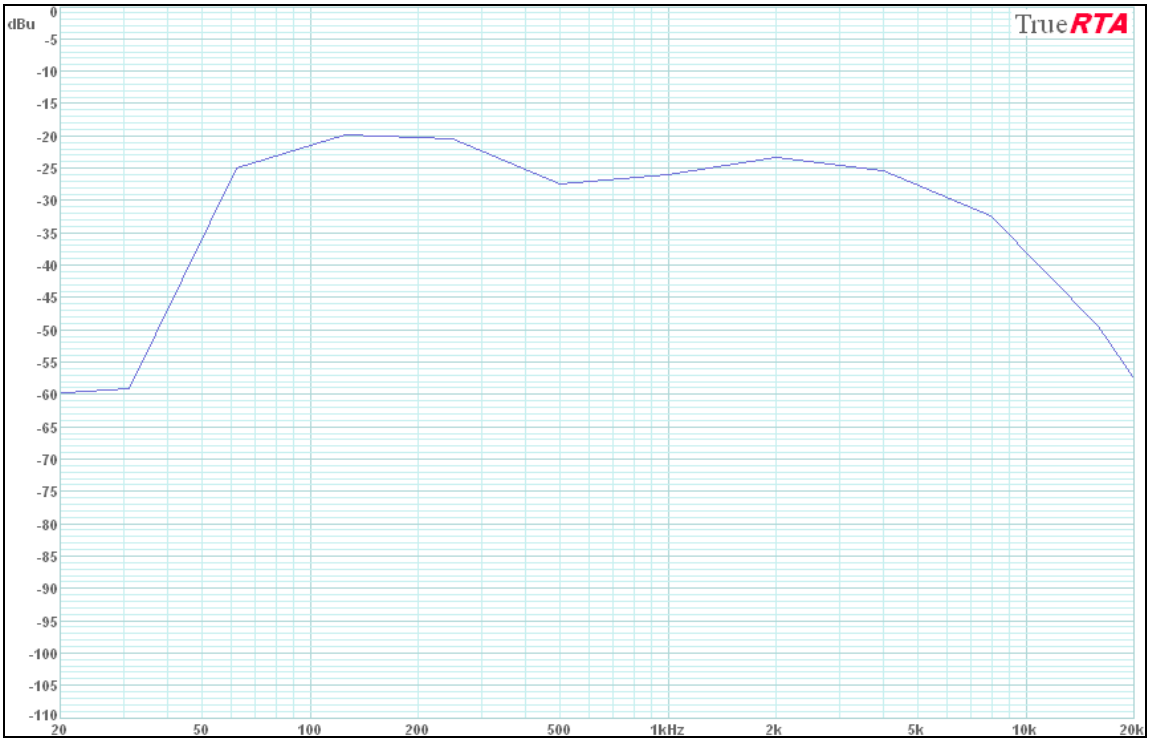


Figure 4.10: Analysis on eighth band

As we can see from the Figure 4.10 the gain in dB for the selected band is around 6 dB. As the same as before this happen resulting on the effect of bell curve and the limitation of the analyzer.

4.10 > 8000 Hz Analysis

There are not much that we can expect from this analysis since the pink noise that this TrueRTA can produce is only from 50 Hz – 10 kHz. Eventhough 10 kHz is more than 8 kHz but the limitation for the analyzer which is its only give an octave response will make the desired changes are not big enough. The result for this analysis can be seen from the Figure 4.11.

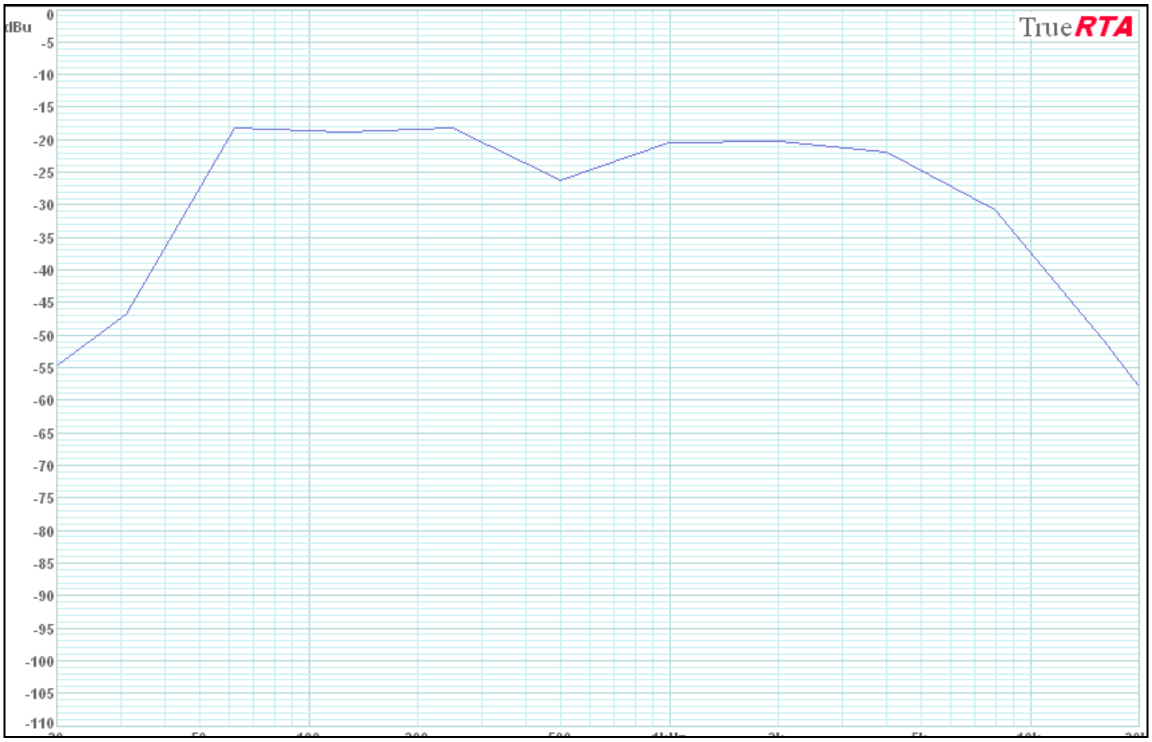


Figure 4.11: Analysis on ninth band

The only thing that we can say here is the bell curve gives a small effect to the result because the frequency response that we get here are slightly different with the previously.

4.11 Limitation

Eventhough I able to test the result and make an analysis on the result but there are some obstacles and limitations that obviously affect the overall result. The first obstacle is how to find the suitable analyer since the only analyzer that we have in laboratory is GSP-810 which only gives a response to the 150 kHz till 2 MHz.

But I'm able to solve the problem stated above by using TrueRTA but here other problems occur again. First this analyzer a octave analyzer. Eventhough we can get the 1/3 octave or even more powerfull analyzer up to 1/24 octave but for that kind of analyzer we need to buy it. This octave analyzer wills give not the exact result but it is acceptable.

Other problem is the surrounding noise. Since this TrueRTA is so sensitive, it capture all sound in the room where the analysis been made. The developper of this TrueRTA had suggested using measurement microphones and mic preamplifiers. Eventhough there is solution but these kind of microphones cost a lot of money.

CHAPTER 5

CONCLUSION AND RECOMMENDATION

5.1 Conclusion

Some testing upon doing this project has been done, and to conclude, overall output level of a sound reinforcement system can be increased by reducing the system's output in the frequency bands at which feedback occurs. The naturalness or intelligibility of a sound reinforcement system can also be improved by emphasizing the frequency ranges most critical for speech.

Other than that, this particular project can achieve a range of frequency response of 60 Hz to >8 kHz. This device can give an approximate 15 dB of gain and also worked well with mono or stereo input. Bear in mind that the important of equalizer in PA system is to reduce feedback, and this equalizer can be reliable on that.

In term of the accomplishment of this device, it can be concluded as a succesfull project because at the end, it achieves everything that already been state as scopes of this project. The equalizer also can do approximately what other available equalizer in the market can perform.

However if this equalizer is used for recording purpose, it may not perform well since the range of frequency that this device cover is from 60 Hz to 8 kHz. This happen because as the recording progress, all other sound from broad ranges of frequency will also be recorded. This is one fault of this device.

For a Public Adress (PA) System, this device can work well with the mixer, power amplifier, and loud speaker as well. This device accomplished the first objective in building this equalizer, which is to make a useable equalizer for the faculty.

The final product for this project can be seen from the Figure 5.1 and the circuit inside the product also shown here in Figure 5.2.

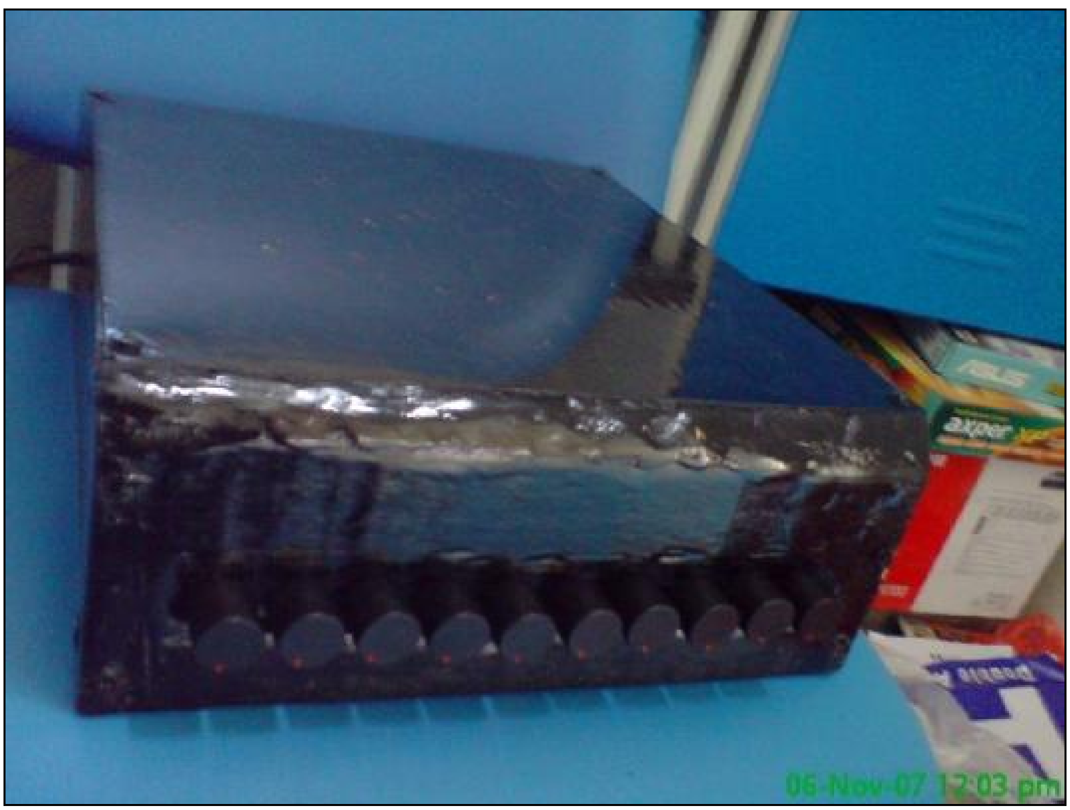


Figure 5.1: The final product

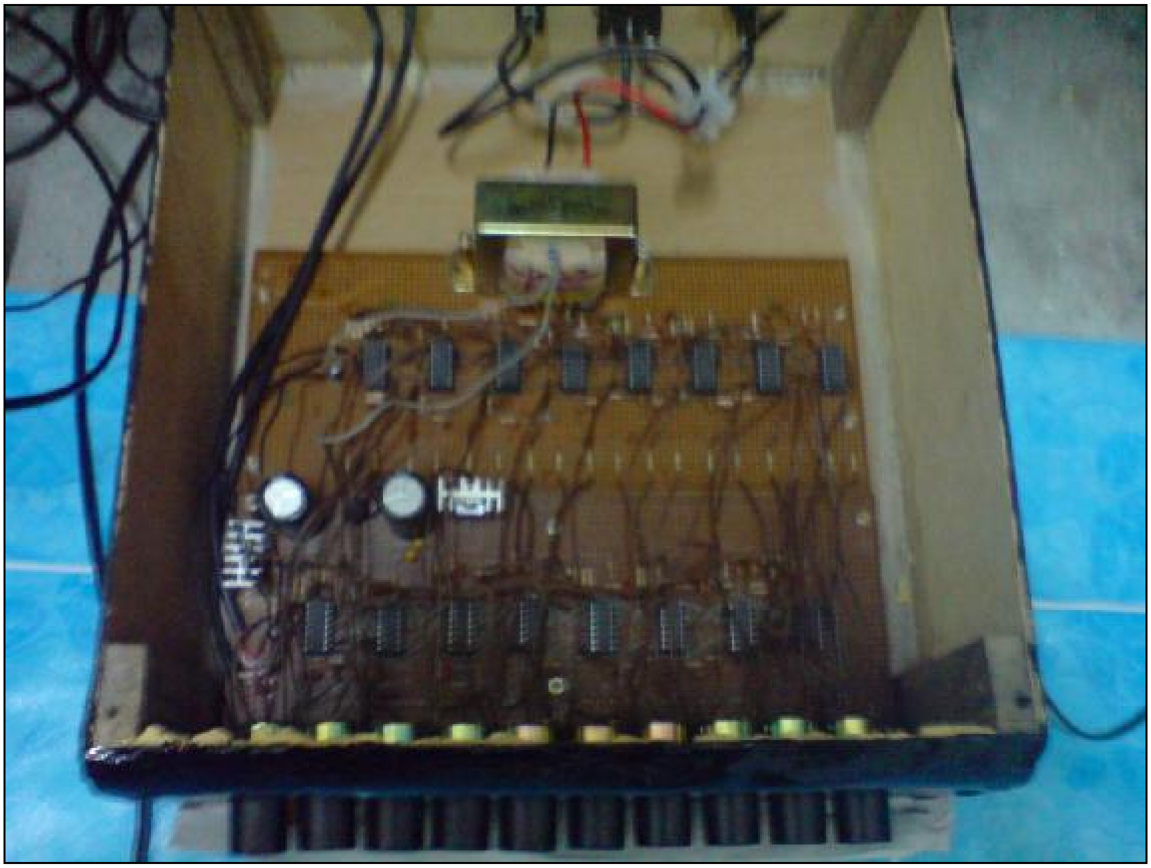


Figure 5.2: Inside the product

5.2 Recommendation

Although all objectives stated earlier have been achieved, some of it can still be improve in the future, such as if this equalizer is in the PA system, we can combine this equalizer with other device that will also can be used in the system as well.

The common used is combination of mixer and equalizer. The disadvantage of this combination is in the term of the space limitation in modelling the product, however the advantage of this kind of device is it is much easier for PA system set up.

This equalizer can also be upgraded to be 1/3 octave. This kind of equalizer will have around thirty or more frequency band thus the usage of potentiometer is not effective. Another disadvantage is such as space limitation. This can be solved if the designation of the 1/3 octave equalizer is using printed circuit board (PCB).

5.2.1 Cost and Commercialization

The purpose of this project is to build and to construct equalizer for faculty usage. But to make available it in the market, it will be slightly a problem since this equalizer is only an octave equalizer. There are many types of advance equalizer available in the market such as 1/3 octave equalizer.

Eventhough it is cheaper than an octave equalizer available in the market we must bear in mind that this is only a prototype project. As a pioneer, it is the only model available, thus the cost to produce will be more expensive than the commercialized cost. The cost to produce this prototype is about RM 190 including the labor cost; it will be around RM 220.

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APENDIX A

USER MANUAL

User Manual for Equalizer for PA system

1. Plug in the power socket and switch on the switch at the back of equalizer.
2. Return to the flat setting (minimun gain) for the entire frequencies band.

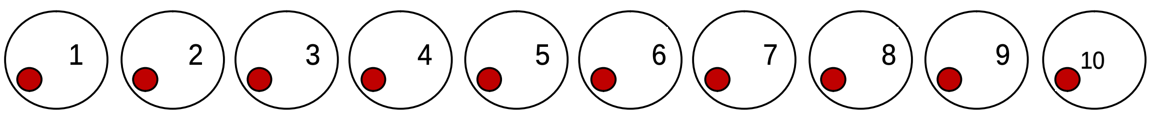


Figure A.1: Front panel of equalizer

3. From the front panel, knob 1 is for volume, knob 2 is for frequency <60Hz, knob 3 is for frequency 60<fc<120Hz, knob 4 is for frequency 120<fc<250Hz, knob 5 is for frequency 250<fc<500Hz, knob 6 is for frequency 500<fc<1000Hz, knob 7 is for frequency 1000<fc<2000Hz, knob 8 is for frequency 2000<fc<4000Hz, knob 9 is for frequency 4000<fc<8000Hz, and knob 10 is for frequency >8000Hz.
4. Even though the name of this equalizer is for PA system it can be use for other function. If this equalizer is use to make the sound of music instrument sound better, make side-by-side comparisons against commercial releases of similar types of music; this will help you in judging overall blend.
5. You can tailor the sound of an instrument only so far without losing its identity; every instrument can't be full, deep, bright, sparkly, etc. all at once. Leave some room for contrast.
6. Take a break once in a while. Critical listening tends to numb one's senses after a while, especially if you listen at high volume levels. Sounds may appear very different to you the next morning.

7. Don't be afraid to experiment. If you can't find just what you want with equalization, try moving the mike a little; if that doesn't work, move the instrument. But, most of all, keep trying.
8. If this equalizer use for a PA system, set all knob for the frequency to the highest gain. Knob for volume will remain to be on the flat setting.

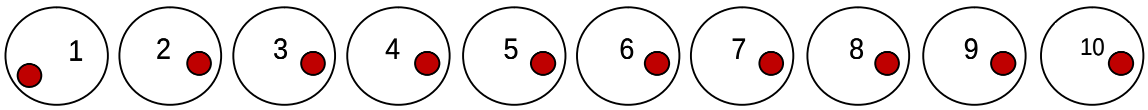


Figure A.2: The maximum gain for all the frequency and flat setting for volume.

9. Request the talker to speak louder to the microphone.
10. Reduce the distance between the talkers to the microphone. Each time this distance is halved the sound system output will increase by 6dB.
11. Reduce the number of open microphones. Each time this number is halved, the sound system output can be increased by 3dB.
12. Move the loudspeaker farther away from the microphone. Each time this distance is doubled, the sound system output can be increased by 6dB.
13. Move the loudspeaker closer to the listener. Each time this distance is halved, the sound system output will increase by 6dB.
14. Use an equalizer to cut the frequency bands in which the feedback occurs by reducing the gain in which frequency feedback occurs (this step need to do by try and error).
15. Approach gently and slowly! After every adjustment, listen carefully to the resulting sound. The goal is to improve sound quality as well as increase the gain before feedback.
16. This manual should be taken as guidelines rather than prescriptions, because every situation is different and every listener has his own desire on how the sound should be. A few general hints may contribute to the effective use of equalization.

APPENDIX B

DATASHEET FOR LM324

Order this document by LM324/D

MOTOROLA

LM324, LM324A, LM224, LM2902, LM2902V

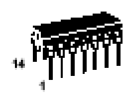
Quad Low Power Operational Amplifiers

The LM324 series are low–cost, quad operational amplifiers with true differential inputs. They have several distinct advantages over standard operational amplifier types in single supply applications. The quad amplifier can operate at supply voltages as low as 3.0 V or as high as 32 V with quiescent currents about one–fifth of those associated with the MC1741 (on a per amplifier basis). The common mode input range includes the negative supply, thereby eliminating the necessity for external biasing components in many applications. The output voltage range also includes the negative power supply voltage.

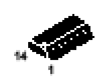
- Short Circuited Protected Outputs
- True Differential Input Stage
- Single Supply Operation: 3.0 V to 32 V
- Low Input Bias Currents: 100 nA Maximum (LM324A)
- Four Amplifiers Per Package
- Internally Compensated
- Common Mode Range Extends to Negative Supply
- Industry Standard Pinouts
- ESD Clamps on the Inputs Increase Ruggedness without Affecting Device Operation

QUAD DIFFERENTIAL INPUT OPERATIONAL AMPLIFIERS

SEMICONDUCTOR TECHNICAL DATA

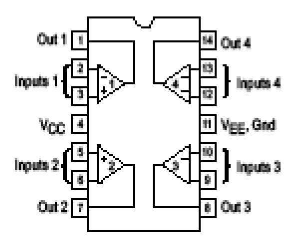


N SUFFIX
PLASTIC PACKAGE
CASE 646
(LM224, LM324, LM2902 Only)



D SUFFIX
PLASTIC PACKAGE
CASE 751A
(SO–14)

PIN CONNECTIONS



(Top View)

MAXIMUM RATINGS (T_A = +25 C, unless otherwise noted.)

Rating	Symbol	LM224 LM324, LM324A	LM2902, LM2902V	Unit
Power Supply Voltages Single Supply Split Supplies	 V_{CC} V_{CC}, V_{EE}	 32 –16	 26 +13	Vdc
Input Differential Voltage Range (See Note 1)	V_{IDR}	±32	±26	Vdc
Input Common Mode Voltage Range	V_{ICR}	–0.3 to 32	–0.3 to 26	Vdc
Output Short Circuit Duration	t_{SC}	Continuous		
Junction Temperature	T_J	150		C
Storage Temperature Range	T_{stg}	–65 to +150		C
Operating Ambient Temperature Range	T_A	–25 to +85 0 to +70	–40 to +105 –40 to +125	°C

NOTE: 1. Split Power Supplies.

ORDERING INFORMATION

Device	Operating Temperature Range	Package
LM2902D	T_A = –40° to +105°C	SO–14
LM2902N		Plastic DIP
LM2902VD	T_A = –40 to +125°C	SO–14
LM2902VN		Plastic DIP
LM224D	T_A = –25 to +85°C	SO–14
LM224N		Plastic DIP
LM324AD	T_A = 0° to +70 C	SO–14
LM324AN		Plastic DIP
LM324D		SO–14
LM324N		Plastic DIP

 Rev 3

LM324, LM324A, LM224, LM2902, LM2902V

ELECTRICAL CHARACTERISTICS (V_{CC} = 5.0 V, V_{EE} = Gnd, T_A = 25°C, unless otherwise noted.)

Characteristics	Symbol	LM224			LM324A			LM324			LM2902			LM2902V			Unit
		Min	Typ	Max	Min	Typ	Max	Min	Typ	Max	Min	Typ	Max	Min	Typ	Max	
Input Offset Voltage V_{CC} = 5.0 V to 30 V (26 V for LM2902, V), V_{ICR} = 0 V to V_{CC} −1.7 V, V_O = 1.4 V, R_S = 0 Ω	V_{IO}																mV
T_A = 25°C		–	2.0	5.0	–	2.0	3.0	–	2.0	7.0	–	2.0	7.0	–	2.0	7.0	
$T_A = T_{high}$ (1)		–	–	7.0	–	–	5.0	–	–	9.0	–	–	10	–	–	13	
$T_A = T_{low}$ (1)		–	–	7.0	–	–	5.0	–	–	9.0	–	–	10	–	–	10	
Average Temperature Coefficient of Input Offset Voltage $T_A = T_{high}$ to T_{low} (1)	$\Delta V_{IO}/\Delta T$	–	7.0	–	–	7.0	30	–	7.0	–	–	7.0	–	–	7.0	–	μV/°C
Input Offset Current	I_{IO}	–	3.0	30	–	5.0	30	–	5.0	50	–	5.0	50	–	5.0	50	nA
$T_A = T_{high}$ to T_{low} (1)		–	–	100	–	–	75	–	–	150	–	–	200	–	–	200	
Average Temperature Coefficient of Input Offset Current $T_A = T_{high}$ to T_{low} (1)	$\Delta I_{IO}/\Delta T$	–	10	–	–	10	300	–	10	–	–	10	–	–	10	–	pA/°C
Input Bias Current	I_{IB}	–	−90	−150	–	−45	−100	–	−90	−250	–	−90	−250	–	−90	−250	nA
$T_A = T_{high}$ to T_{low} (1)		–	–	−300	–	–	−200	–	–	−500	–	–	−500	–	–	−500	
Input Common Mode Voltage Range (2)	V_{ICR}																V
V_{CC} = 30 V (26 V for LM2902, V)		0	–	28.3	0	–	28.3	0	–	28.3	0	–	24.3	0	–	24.3	
V_{CC} = 30 V (26 V for LM2902, V), T_A = Thigh to Tlow		0	–	28	0	–	28	0	–	28	0	–	24	0	–	24	
Differential Input Voltage Range	V_{IDR}	–	–	V_{CC}	–	–	V_{CC}	–	–	V_{CC}	–	–	V_{CC}	–	–	V_{CC}	V
Large Signal Open Loop Voltage Gain	A_{VOL}																V/mV
R_L = 2.0 kΩ, V_{CC} = 15 V, for Large V_O Swing,		50	100	–	25	100	–	25	100	–	25	100	–	25	100	–	
$T_A = T_{high}$ to T_{low} (1)		25	–	–	15	–	–	15	–	–	15	–	–	15	–	–	
Channel Separation 10 kHz ≤ f ≤ 20 kHz, Input Referenced	CS	–	−120	–	–	−120	–	–	−120	–	–	−120	–	–	−120	–	dB
Common Mode Rejection, R_S ≤ 10 kΩ	CMR	70	85	–	65	70	–	65	70	–	50	70	–	50	70	–	dB
Power Supply Rejection	PSR	65	100	–	65	100	–	65	100	–	50	100	–	50	100	–	dB
Output Voltage–High Limit (T_A = Thigh to T_{low}) (1)	V_{OH}																V
V_{CC} = 5.0 V, R_L = 2.0 kΩ, T_A = 25°C		3.3	3.5	–	3.3	3.5	–	3.3	3.5	–	3.3	3.5	–	3.3	3.5	–	
V_{CC} = 30 V (26 V for LM2902, V), R_L = 2.0 kΩ		26	–	–	26	–	–	26	–	–	22	–	–	22	–	–	
V_{CC} = 30 V (26 V for LM2902, V), R_L = 10 kΩ		27	28	–	27	28	–	27	28	–	23	24	–	23	24	–	

NOTES: 1. T_{low} = −25°C for LM224; = 0°C for LM324, A; = −40°C for LM2902; = −40°C for LM2902V
T_{high} = +85°C for LM224; = +70°C for LM324, A; = +105°C for LM2902; = +125°C for LM2902V

2. The input common mode voltage or either input signal voltage should not be allowed to go negative by more than 0.3 V. The upper end of the common mode voltage range is V_{CC} −1.7 V.

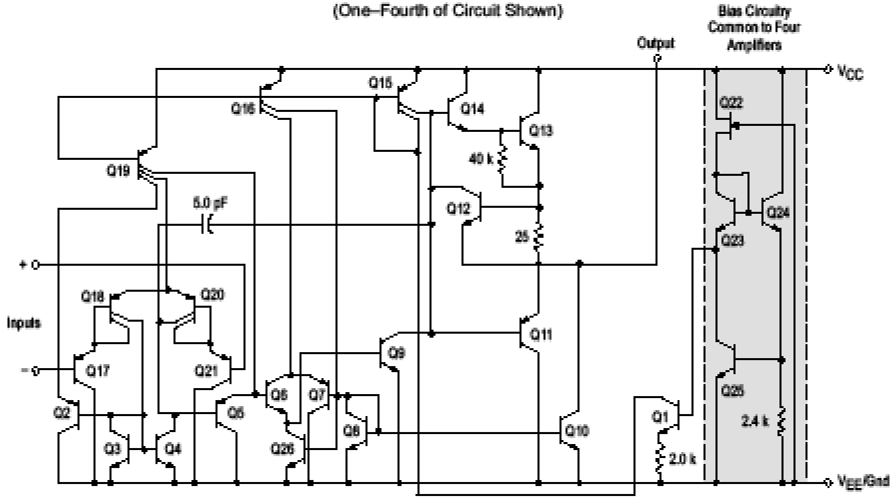
LM324, LM324A, LM224, LM2902, LM2902V

ELECTRICAL CHARACTERISTICS (V_{CC} = 5.0 V, V_{EE} = Gnd, T_A = 25 C, unless otherwise noted.)

Characteristics	Symbol	LM224			LM324A			LM324			LM2902			LM2902V			Unit
		Min	Typ	Max	Min	Typ	Max	Min	Typ	Max	Min	Typ	Max	Min	Typ	Max	
Output Voltage – Low Limit, V_{CC} = 5.0 V, R_L = 10 kΩ, T_A = T_{high} to T_{low} (1)	V_{OL}	–	5.0	20	–	5.0	20	–	5.0	20	–	5.0	100	–	5.0	100	mV
Output Source Current (V_{ID} = +1.0 V, V_{CC} = 15 V)	I_O +																mA
T_A = 25 C		20	40	–	20	40	–	20	40	–	20	40	–	20	40	–	
T_A = T_{high} to T_{low} (1)		10	20	–	10	20	–	10	20	–	10	20	–	10	20	–	
Output Sink Current (V_{ID} = –1.0 V, V_{CC} = 15 V) T_A = 25 C	I_O –	10	20	–	10	20	–	10	20	–	10	20	–	10	20	–	mA
T_A = T_{high} to T_{low} (1)		5.0	8.0	–	5.0	8.0	–	5.0	8.0	–	5.0	8.0	–	5.0	8.0	–	
(V_{ID} = –1.0 V, V_O = 200 mV, T_A = 25 C)		12	50	–	12	50	–	12	50	–	–	–	–	–	–	–	μA
Output Short Circuit to Ground (3)	I_{SC}	–	40	60	–	40	60	–	40	60	–	40	60	–	40	60	mA
Power Supply Current (T_A = T_{high} to T_{low}) (1)	I_{CC}																mA
V_{CC} = 30 V (26 V for LM2902, V), V_O = 0 V, R_L = •		–	–	3.0	–	1.4	3.0	–	–	3.0	–	–	3.0	–	–	3.0	
V_{CC} = 5.0 V, V_O = 0 V, R_L = ·		–	–	1.2	–	0.7	1.2	–	–	1.2	–	–	1.2	–	–	1.2	

NOTES: 1. T_{low} = –25 C for LM224; = 0 C for LM324, A; = –40 C for LM2902; = –40 C for LM2902V
T_{high} = +85 C for LM224; = +70 C for LM324, A; = +105 C for LM2902; = +125 C for LM2902V
2. The input common mode voltage or either input signal voltage should not be allowed to go negative by more than 0.3 V. The upper end of the common mode voltage range is V_{CC} –1.7 V.

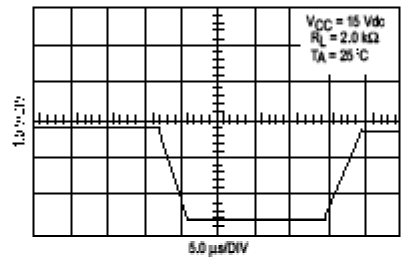
Representative Circuit Diagram
(One–Fourth of Circuit Shown)



CIRCUIT DESCRIPTION

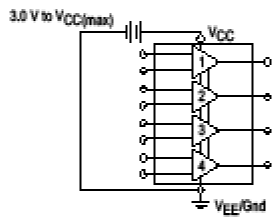
The LM324 series is made using four internally compensated, two–stage operational amplifiers. The first stage of each consists of differential input devices Q20 and Q18 with input buffer transistors Q21 and Q17 and the differential to single ended converter Q3 and Q4. The first stage performs not only the first stage gain function but also performs the level shifting and transconductance reduction functions. By reducing the transconductance, a smaller compensation capacitor (only 5.0 pF) can be employed, thus saving chip area. The transconductance reduction is accomplished by splitting the collectors of Q20 and Q18. Another feature of this input stage is that the input common mode range can include the negative supply or ground, in single supply operation, without saturating either the input devices or the differential to single–ended converter. The second stage consists of a standard current source load amplifier stage.

Large Signal Voltage Follower Response

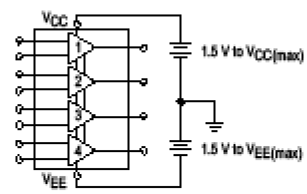


Each amplifier is biased from an internal–voltage regulator which has a low temperature coefficient thus giving each amplifier good temperature characteristics as well as excellent power supply rejection.

Single Supply



Split Supplies



LM324, LM324A, LM224, LM2902, LM2902V

Figure 1. Input Voltage Range

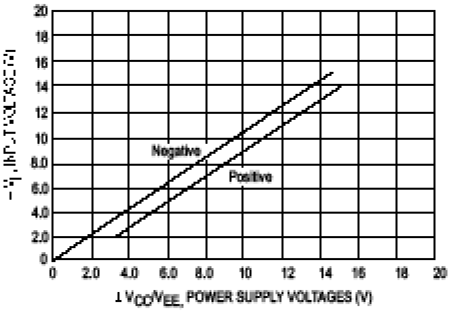


Figure 2. Open Loop Frequency

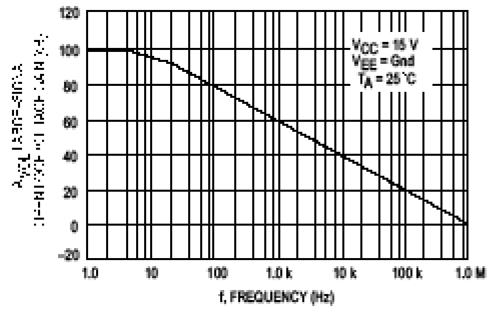


Figure 3. Large–Signal Frequency Response

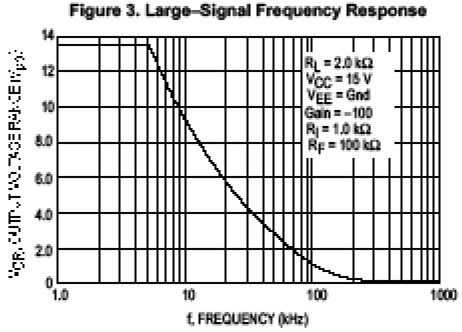


Figure 4. Small–Signal Voltage Follower Pulse Response (Noninverting)

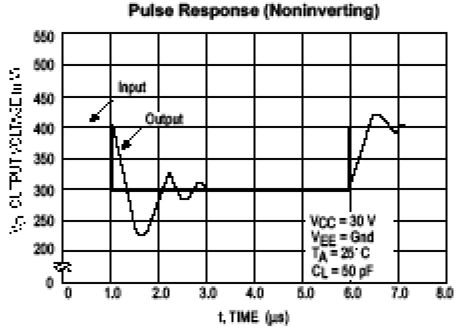


Figure 5. Power Supply Current versus Power Supply Voltage

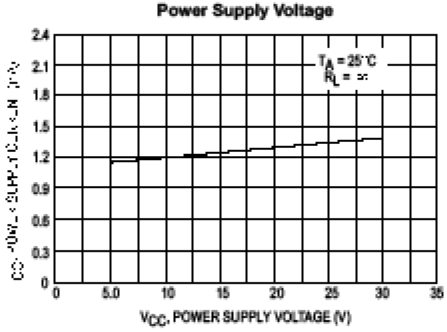


Figure 6. Input Bias Current versus Power Supply Voltage

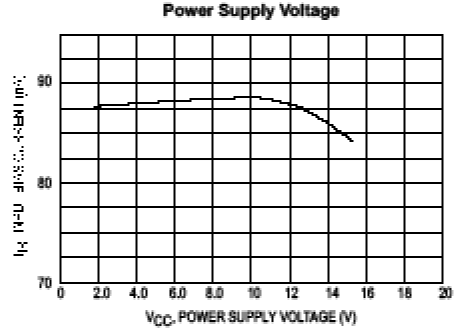
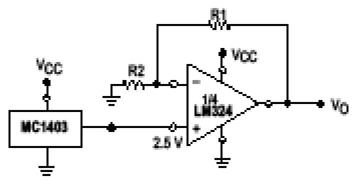


Figure 7. Voltage Reference



$$V_O = 2.5\,V\left(1 + \frac{R1}{R2}\right)$$

Figure 8. Wien Bridge Oscillator

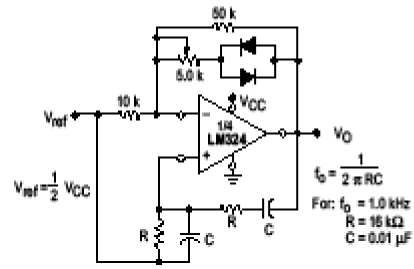
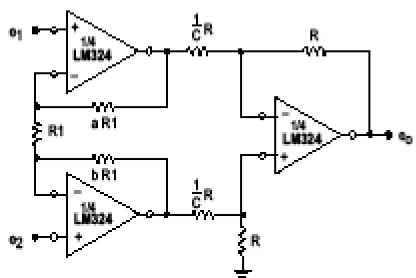
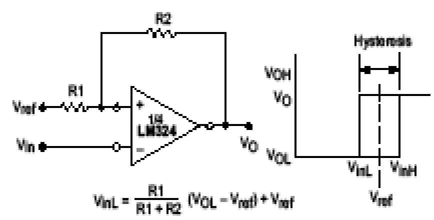


Figure 9. High Impedance Differential Amplifier



$$e_o = C\,(1 + a + b)\,(e_2 - e_1)$$

Figure 10. Comparator with Hysteresis



$$V_{inL} = \frac{R1}{R1 + R2}\,(V_{OL} - V_{ref}) + V_{ref}$$

$$V_{inH} = \frac{R1}{R1 + R2}\,(V_{OH} - V_{ref}) + V_{ref}$$

$$H = \frac{R1}{R1 + R2}\,(V_{OH} - V_{OL})$$

Figure 11. Bi–Quad Filter

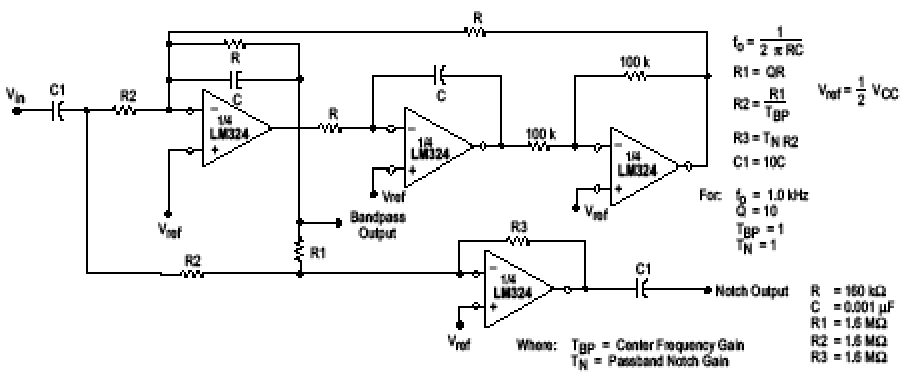
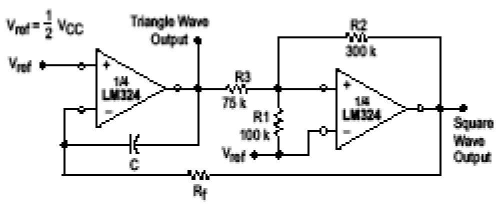
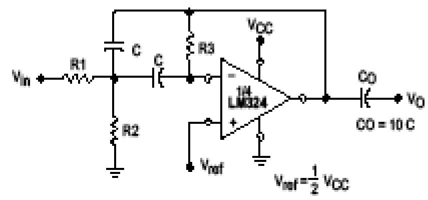


Figure 12. Function Generator



$$f = \frac{R1 + R_C}{4\,CR_f\,R1} \quad \text{if } R3 = \frac{R2\,R1}{R2 + R1}$$

Figure 13. Multiple Feedback Bandpass Filter



Given: f_o = center frequency
$A(f_o)$ = gain at center frequency

Choose value f_o, C

Then: $$R3 = \frac{Q}{\pi f_o C}$$

$$R1 = \frac{R3}{2\,A(f_o)}$$

$$R2 = \frac{R1\,R3}{4Q^2\,R1 - R3}$$

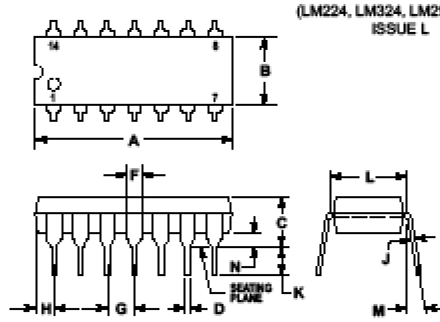
For less than 10% error from operational amplifier, $\frac{Q_o f_o}{BW} < 0.1$

where f_o and BW are expressed in Hz.

If source impedance varies, filter may be preceded with voltage follower buffer to stabilize filter parameters.

OUTLINE DIMENSIONS

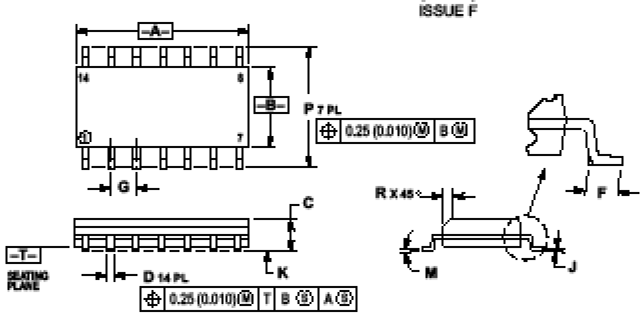
N SUFFIX
PLASTIC PACKAGE
CASE 646–06
(LM224, LM324, LM2902 Only)
ISSUE L



NOTES:
1. LEADS WITHIN 0.13 (0.005) RADIUS OF TRUE POSITION AT SEATING PLANE AT MAXIMUM MATERIAL CONDITION.
2. DIMENSION L TO CENTER OF LEADS WHEN FORMED PARALLEL.
3. DIMENSION B DOES NOT INCLUDE MOLD FLASH.
4. ROUNDED CORNERS OPTIONAL.

DIM	INCHES MIN	INCHES MAX	MILLIMETERS MIN	MILLIMETERS MAX
A	0.715	0.770	18.16	19.56
B	0.240	0.260	6.10	6.60
C	0.145	0.185	3.69	4.69
D	0.015	0.021	0.38	0.53
F	0.040	0.070	1.02	1.78
G	0.100 BSC		2.54 BSC	
H	0.052	0.095	1.32	2.41
J	0.008	0.015	0.20	0.38
K	0.115	0.135	2.92	3.43
L	0.300 BSC		7.62 BSC	
M	0°	10°	0°	10°
N	0.015	0.039	0.39	1.01

D SUFFIX
PLASTIC PACKAGE
CASE 751A–03
(SO–14)
ISSUE F



NOTES:
1. DIMENSIONING AND TOLERANCING PER ANSI Y14.5M, 1982.
2. CONTROLLING DIMENSION: MILLIMETER.
3. DIMENSIONS A AND B DO NOT INCLUDE MOLD PROTRUSION.
4. MAXIMUM MOLD PROTRUSION 0.15 (0.006) PER SIDE.
5. DIMENSION D DOES NOT INCLUDE DAMBAR PROTRUSION. ALLOWABLE DAMBAR PROTRUSION SHALL BE 0.127 (0.005) TOTAL IN EXCESS OF THE D DIMENSION AT MAXIMUM MATERIAL CONDITION.

DIM	MILLIMETERS MIN	MILLIMETERS MAX	INCHES MIN	INCHES MAX
A	8.55	8.75	0.337	0.344
B	3.80	4.00	0.150	0.157
C	1.35	1.75	0.054	0.068
D	0.35	0.49	0.014	0.019
F	0.40	1.25	0.016	0.049
G	1.27 BSC		0.050 BSC	
J	0.19	0.25	0.008	0.009
K	0.10	0.25	0.004	0.009
M	0°	7°	0°	7°
P	5.80	6.20	0.228	0.244
R	0.25	0.50	0.010	0.019

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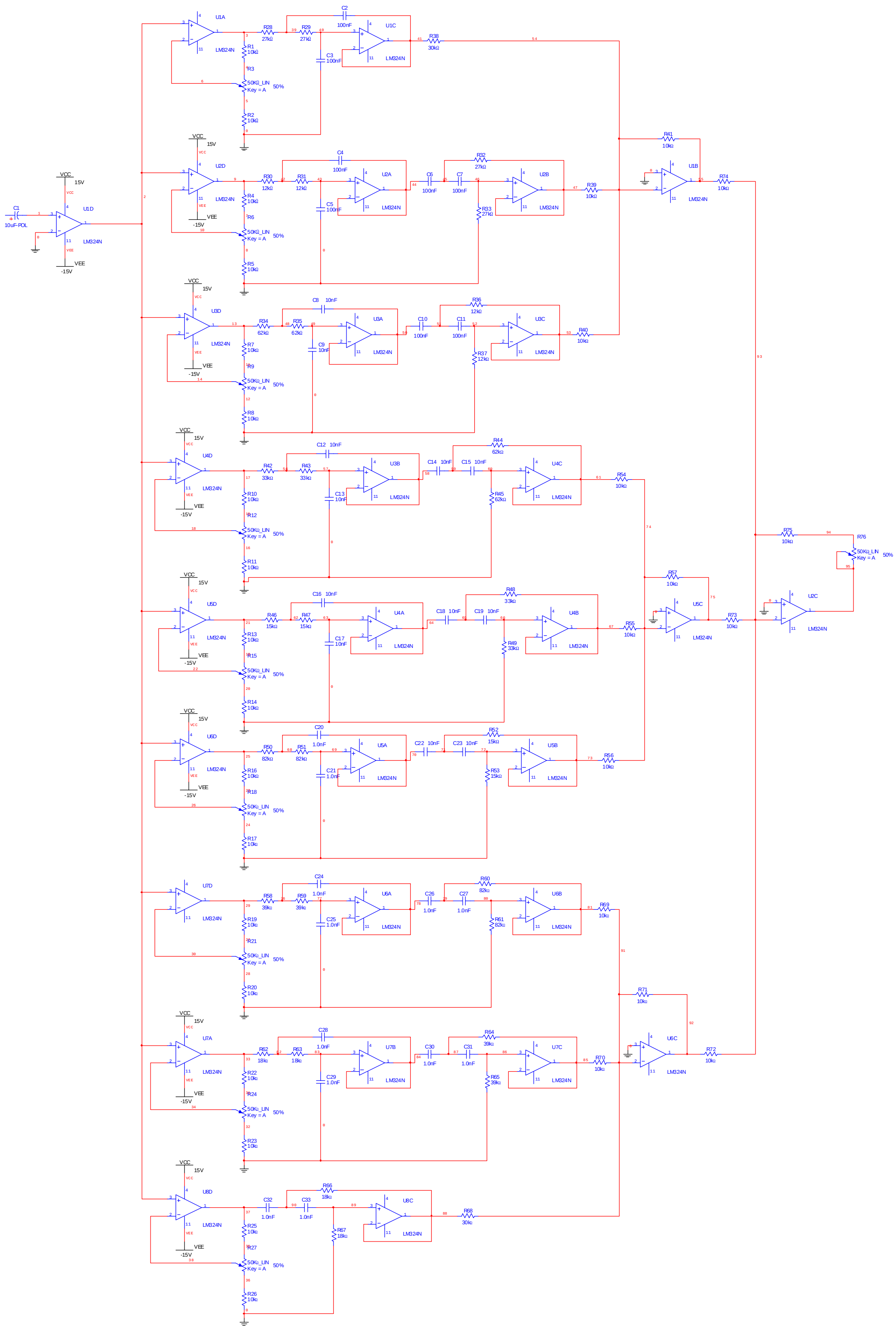
MFAX: RMFAX0@email.sps.mot.com – TOUCHTONE 602–244–6609
INTERNET: http://Design–NET.com

ASIA/PACIFIC: Motorola Semiconductors H.K. Ltd.; 8B Tai Ping Industrial Park, 51 Ting Kok Road, Tai Po, N.T., Hong Kong. 852–26629298

APPENDIX C

COMPLETE CIRCUIT FOR EQUALIZER



APPENDIX D

LIST OF COMPONENTS

Bil	Components	Spesification	Quantity
1	resistor	10k (1/4W 5%)	64
2	resistor	27k (1/4W 5%)	8
3	resistor	12k (1/4W 5%)	8
4	resistor	62k (1/4W 5%)	8
5	resistor	33k (1/4W 5%)	8
6	resistor	15k (1/4W 5%)	8
7	resistor	82k (1/4W 5%)	8
8	resistor	39k (1/4W 5%)	8
9	resistor	18k (1/4W 5%)	8
10	resistor	30k (1/4W 5%)	8
11	capacitor	100nF	16
12	capacitor	10nF	18
13	capacitor	1nF	24
14	capacitor	10uF 16V	2
15	capacitor	1000uF 35V	2
16	capacitor	1uF 16V tantalum	2
17	Dual potentiometer	50k	10
18	diode	1N4004	4
19	Op amp + base	LM324	20
20	regulator	MC7815	1
21	regulator	MC7915	1
22	transformer	15V center taped	1

23	Fuse	1A slow blow fuse	1
24	switch	2pin toggle switch	1
25	plug	Stereo plug	1
26	Knob	Circle	10
27	Cable	Stereo cable	2 meter
28	Supply jack	3 pin Male	1