DESIGN AND CONSTRUCTION OF AUDIO GRAPHIC EQUALIZER WITH LEVEL DISPLAY.

Presented By

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DEDICATION

This project is dedicated to God Almighty for His immeasurable love, divine protection and provision throughout my stay in the university.

DECLARATION

I Ijibayiwa Sehinde Moses declare that this work was done by me and has never being presented elsewhere for the award of a degree. I also hereby relinquish the copyright of the Federal University of Technology, Minna.

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CHAPTER ONE

1.0 INTRODUCTION

Few would argue the necessity of equalizer for quality sound reinforcement. Without equalizer the audio sound system is left without nearly enough controls to turn, to try ad correct for room difficulties, speaker anomalies and individual performer preferences.

Rudolf Graf's modern Dictionary of electronics defines equalizer as a devices designed to compensate for an undesired amplitude frequency and/or phase frequency characteristic of a system or component. It also defines equalization as the process of reducing the frequency and/or the phase distortion of a circuit by the introduction of network to compensate for the difference in attenuation and or time delay at the various frequencies in the transmission band.

A graphic equalizer is an audio component used to flatten the system spectral response in the audio signal band or produce other desirable effects. In the application 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 characteristic of the room, concert hall, speaker system and other factors affecting the sound. In high fidelity sound reproduction systems, the chief concern of the user is that the sound reaching the listener should confirm as precisely as possible to the supplied source signal, whether it be from a turntable, tuner, tape-deck, or other source. The difficult with the listening environment arises from the difference in its responses to different frequency sounds. Some listening environment may be quite lively, providing multiple reflections of high frequency components, whereas other may be quite dead, providing substantial damping of high frequency components. To combat the influence of the listening environment upon the fidelity of reproduction of the audio signal, it has become popular to introduce modifications in the frequency response characteristics of the audio system which compensate for the colorations introduced by the listening environment. This is generally accomplished by means of an audio signal path between the signal source and the speakers. An equalizer is often used in the amplification system to correct or produce a desired frequency response of the broadcast system and environment producing the audio program heard by the audience.

A graphic equalizer is a popular apparatus which divides the frequency band of audio signals into a range of frequency sub-bands or channels and which can vary the frequency characteristics at the respective channels, and thereby change the frequency characteristics over entire band s desired. A graphic equalizer typically will adjust the energy level of the audio data in one or more different frequency bands in order to change the characteristics of the audio data. An equalizer may also be used to add more audio energy to the lower frequency bands which will then provide more bass sounds. Speaker system employ elaborate crossover network circuitry to divide the signal among drivers whose design and composition are suited to relatively uniform performance with a frequency range. Graphic equalizers are introduced into the reproduction system to correct the signal for the variable influence of the listening room in different frequency ranges and to further compensate for speaker deficiencies, again of given amplitude whose frequency is varied.

1.1 AIMS AND OBJECTIVES

The main aim of this project is to design and construct a low cost, wide band audio graphic equalizer. The system is to function as a stand-alone unit with audio input and output terminals respectively.

To realize this goal, the following factors were considered to form the basis of the system design and development:

I. The amplitude- frequency response of the audio signal at various frequency bands falls between $100H_Z$ and $10KH_Z$, should be capable of adjustment, i.e. de-emphasis and pre-emphasis for high frequencies, as well as attenuation and boosting of low frequencies in order to achieve the desired sound effect.

II. The acquired and reproduced sound must possess high quality and fidelity.

III. It should be relatively inexpensive.

IV. I should be easy to set up, reliable and should require minimum maintenance.

This project will in no small way, enhance the sound quality, and thus the listening pleasure that people derive from high fidelity stereo equipment.

1.2 SCOPE

The audio graphic equalizer designed have comprises of five biquads which form the five bands of the audio graphic equalizer, with a frequency range of $100H_Z$, which fall well within he the audible frequency range that lies between $20H_Z$ to $20KH_Z$. this designed audio graphic equalizer have a frequency determining circuit for each narrow audio frequency band that include an adjustable component, a slide resistor, oriented vertically in such a way that the position the controls of a plurality of adjustment adjustable components for each separate frequency band gives a graphical representation of the overall frequency response of the equalizer.

This designed audio graphic equalizer also comprises of display which enable the user to view the graphic level of the input signal at a particular point in time. The equalizer made use light emitting diode (LED) as a means of display to display the graphic level of input signal, though this does not produce the best of efficiency as that of the liquid crystal display (LCD).

1.3 METHODOLOGY

The method employed in the design of this equalizer is that of the basic pass filter which pass some signal of certain frequency and reject the other frequencies that are outside those specified frequencies. A **BA3812L** IC is used in the design of equalizer. The **BA3812L** is a five point graphic equalizer that all the required function integrated into one IC. External Capacitors C_0 and C_1 of appropriate values are connected at the terminals of the BA3812L Ic to determine the centre frequencies of each band of equalizer is fed into Lm324 Ic that is connected in an open collector for the purpose of inverting the output of the equalizer. The inverted output is then fed into Lm324 IC which compares the output with a voltage that is fed into it through a variable resistor and then display the outcome using LED (Light Emitting Diode).

1.4 PROJECT OUTLINE

The theoretical background and literature review of audio graphic equalizer is treated in Chapter two.

The systematic design and development for the audio graphic equalizer is treated in chapter three. The transfer function synthesis, and frequency response analysis for each biquad of the cascaded filter network, as well as other related parameters employed in the equalizer design are also treated in detail.

Chapter Four deals with the testing, Results and Discussion of results.

In chapter five, drawn up conclusions, Recommendations for further work, as well as references are presented.

CHAPTER TWO
LITERATURE REVIEW

2.1. HISTORICAL BACKGROUND

The early forms of sound recording were considered miraculously lifelike. This was from the first successful recording. Thomas Edison's recitation of Mary had a little lamb, inscribed on a tin foil cylinder in 1877.in time; it became clear that a more faithful reproduction would increases the usefulness and impact of sound.

Little progress was made in effect to attain high fidelity in sound reproduction until the introduction of electrical recording in 1952. Amplifier of the electrical signal at all stages was the key to a vast new expansion of recording. Mathew nature and Holly spawned the first use of variable equalizer for sound improvement. Motion picture with sound brought audio play back system into theaters for the first time soon, some peoples' attention focused on the "30"s and Volkmann worked for RCA. John Volkmann was credited with being the first person to use a variable equalizer to improve reproduced sound. He applied this new tool to equalize a motion picture theater playback system. [1, 2, 3].

While Bell laboratory used fixed equalizers than this foe correcting audio transmission losses [4], Volkmann represents one of the first uses of an external variable equalizer as an add component to an installed system. Telephone application involved integrating equalization as part of the receiving electronics, as opposed to thinking of the equalizer as a separate entity.

During the same period Volkmann experiment with equalizer for reproduced sound, Hollywood found uses for them in producing sound. Langevin cinema Engineering and others created outboard operator adjustable equalizer for post-production sound effects speech enhancement [4]. Langevin model EQ-251A represents vary early use of slide controls. While not a graphic equalizer in today sense, it was the forerunners. The EQ-251A featured two slide controls, each with switched frequency points. One corner frequency choices, while other provided peaking boost/cut with four switchable centre frequencies. This passive unit looked and performed equal to anything manufactured today.

Arts Davis's company, cinema engineering, developed the first recognizable graphic equalizer [4] known as the type 7080 Graphic Equalizer, it featured 6 bands with boost/cut range of 8dB, adjustable in 1dB steps. After Art Davis moved to Altec, he designed a 7 band successor to the 7080 known as the model 9062A. A hugely successful graphic equalizer setting into the 70s. Being an active design, the 7080 allowed signal signal boosting without loss, which is a nice feature. (with passive units), boosting of signal require an initial broad band signal loss and then reducing the loss on a band-by-band basis. For example, flat might represent 16dB while a 6dB boost represented only 10dB loss. It was all a matter of reference point.

Another innovative feature of the 7080 was the first use of staggered mixing amps to aid in smooth combining of the equalized audio signal. Cinema Engineering designed 3 mixing amplifier for 6 bands. Using this approach, no amplifier mixed adjacent bands. The centre frequencies were $80H_Z$, $200H_Z$, $500H_Z$, $1.25KH_Z$(labelled 1.3KH), $3.2KH_Z$(loaded 3KH), and $8KH_Z$. The amplifiers mixed $(80+1250)H_Z$, $(200+3200)H_Z$, and $(500+8000)H_Z$ respectively, using separate amplifier to mix signals spaced 4 octaves apart, resulted in seamless recombination at the output.

2.1 OTHER IMPROVEMENT ON GRAPHIC EQUALIZER

Not much happened during the 40's and early 50s due to world war 11 and its aftermath. Most applications of variable equalizers involved post-production work. No serious success at room equalization is known. Then in 1958, Wayne Rudmose (a professor at southern Methodist University, Dallas, Texas) successfully applied new theories about equalization to the Dallas Love Field Airport. Prof. Rudmose published his monumental work [5] and sound system equalization was born.

In 1962, Texas made another major contribution to variable equalizer history. This time it was the University of Texas (Austin) and a Physics professor named C.P. Boner, DR. S. Boner and Rudmose were contemporaries and friends, having co-authored a paper 23years earlier [6]. Boner, acknowledged by many as father of acoustical equalization, built organs as a hobby. From his organ/room turning experiences and acoustical physic knowledge grew a profoundly simple theory. Boner reasoned that when feedback occurs, it did so at one precise frequency, and stop it all you had to do was install a very notch filter at that frequency. He went to one of his former students whose company make precise filters for instrumentation and ask him to design a narrow band audio filter. Gifford White agreed and launched white instruments into the new field of acoustic equalization.

Armed with white equalizers, boner established the foundation theory for acoustic feedback, room-ring modes, and room sound system equalizing technique [7, 8, 9, 10]. Expanding boner's work was a student of Wayne Rudmose named William Conner. In 1967 Conner published a concise paper [11] still considered among the best to describe the theory and methodology of sound system equalization.

Also in 1967, Art Davis, along with Jim Noble and Don Davis (not related) developed the industries first set (passive) for Alter-Lansing. Don Davis presented the paper to the Audio Engineering society in October, 1967 [12]. Dubbed the "Acousta-voice" system, it ushered in the modern age of sound system equalization and represented the ultimate in speed and convenience. The Acousta-voice system proved another path existed for the control of room-ring modes. As an alternative to Boner's narrow band notching technique, 1/3 – octave "board band" filters produced the same results.

In 1982, Reine Corporation pioneered a new type of graphic equalizer called a constant –Q Graphic equalizer to solve one of the most annoying problems that plagued all previous 1/3-octave designs. Namely the band width of the filter was function of the slider position; only at the extreme boost/cut position were the filter bandwidths only 1/3 –octave wide. At all modest boost/cut position the filter bandwidths exceeded one octave. For true "graphic" operation, and real control of a system's frequency response, this was an unacceptable design.

The constant-Q graphic equalizer circuit topology allows true 1/3-octave bandwidth control at all positions. Now, equalizers are available that re accurately "graphic" in the picture formed by conventional designs, if a single slider is boosted 3dB the only that 1/3-octave frequency band is being affected.

In 1987, Yamaha introduced the DEQ7 digital Equalizer, the first stand-alone variable equalizer based on digital processor (DSP) technology [13]. A combination of "graphic" (bad terminology since there is no graphical representation of settings) and parametric, the DEQ7 feature 30 different built-in configurations.

The audio graphic equalizer with level display designed here is a unique graphic equalizer with a graphical display of signal display of its output. The development of this 5 band audio graphic equalizer is the logical next step after reviewing and clearing understanding designs and problems of LRC equalizers and constant-Q equalizers. It's the result of applying the very best constant-Q Graphic equalizer topology. The filter sections are no totally isolated from the effects of the amplitude slide pots with respects to centre frequency and bandwidth, allowing each filter to be designed for the precise centre frequency and narrow bandwidth required. The result is unequalled freedom between bandwidth and slider position. A freedom to make subtle adjustment a reality without resulting to racks of parameter or being forced to 1/6-octave graphic overkill. When using a 1/3-octave analyzer, a constant–Q equalizer gives the best, most accurate results, and truly delivers "graphic" representation of the equalization curve with the front panel sliders.

CHAPTER THREE

3.1 SYSTEM DESIGN AND IMPLEMENTATION

The audio graphic equalizer designed here is 5-band audio graphic equalizer with level display, and frequency range between $100H_z$ and $10KH_z$. This range falls well within the audible frequency range that lies between $20H_z$ to $20KH_z$. It is thus used to modify strictly the amplitude frequency response characteristics of audio signals as the human ear is insensitive to phase variations. It can therefore be adjusted at various frequency bands to give a "flat" or linear response (i.e. all frequencies reproduced equally), for speech waves that are affected in advertently by electronics circuit performance deficiencies or limitations.

Furthermore, it will compensate for the tendency of magnetic tape or discs used in high fidelity equipment to be more sensitive two one range frequencies, by electronic modification of them respective audio output.

3.2 BLOCK DIAGRAM OF AUDIO GRAPHIC EQUALIZER WITH LEVEL DISPLAY

Fig 3.1 shows each block representing a stage in the circuit of audio graphic equalizer with level display. There are basically three (3) modules or stage namely;

- The audio graphic equalizer circuitry.
- The display unit.
- The power unit.

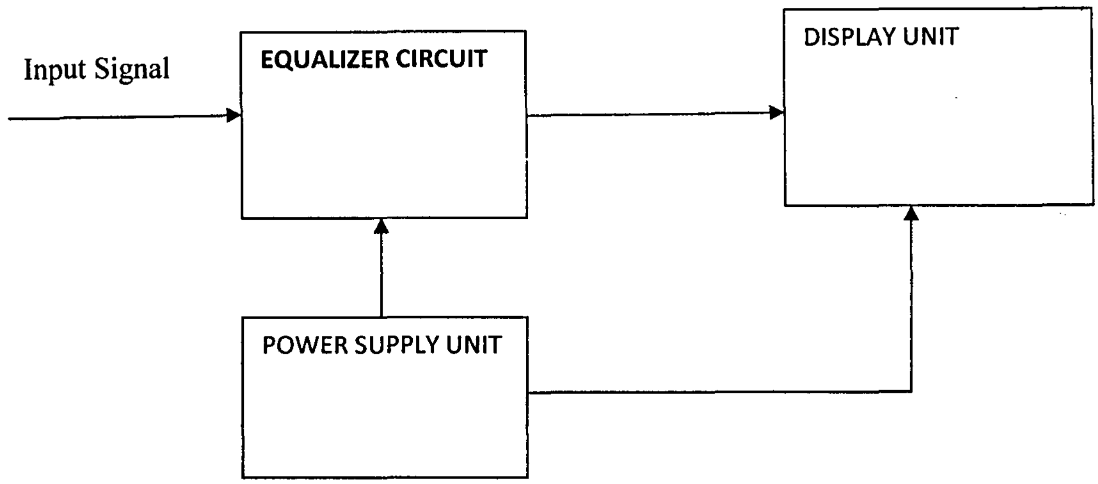
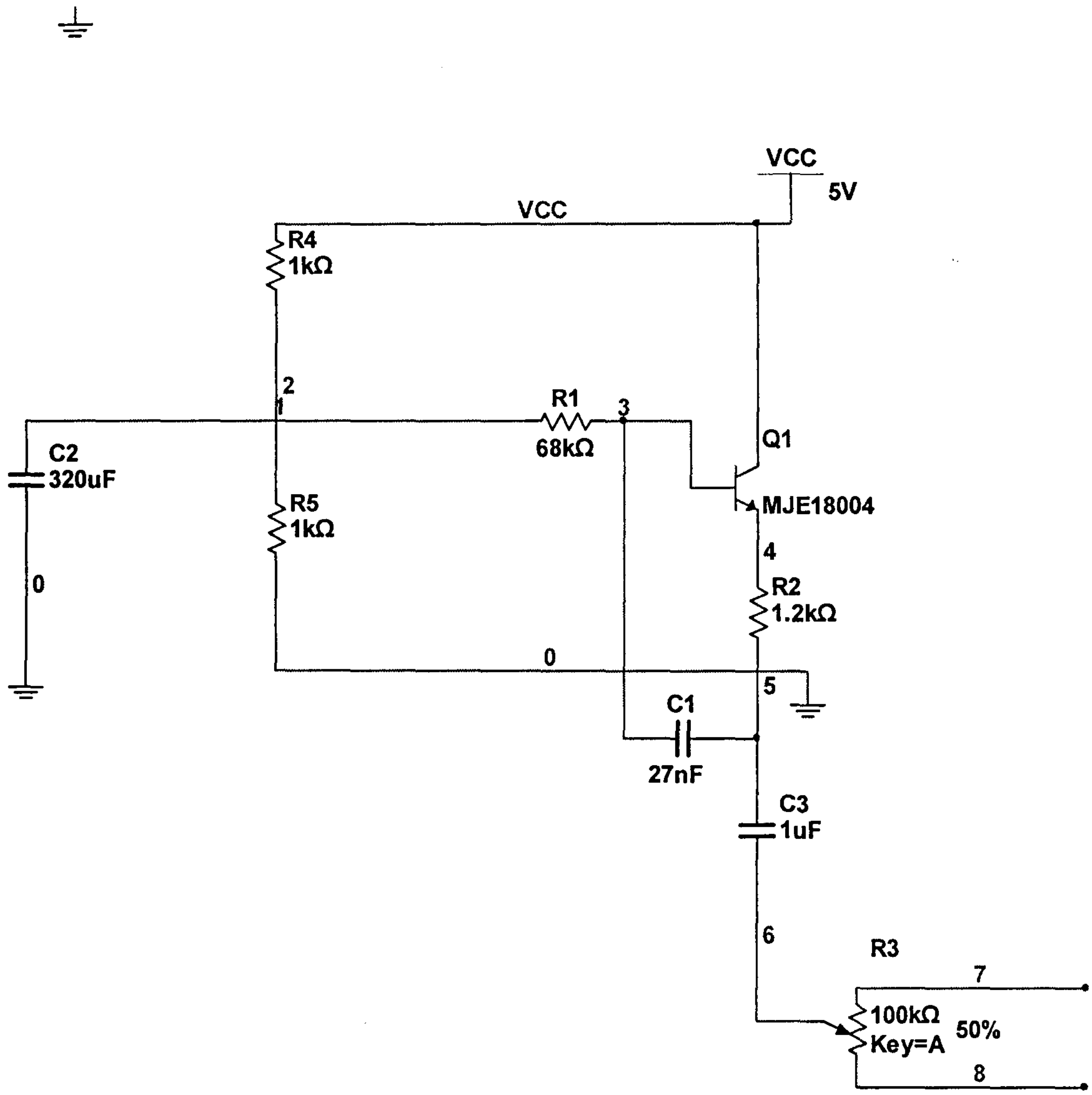


Fig 3.1 Block Diagram of Audio Graphic Equalizer with Level Display

3.3 THE AUDIO GRAPHIC EQUALIZER SYSTEM DESIGN

The graphic equalizer circuit module (stage), is the stage where the filtering of the sound and determining of the centre frequency take place. The design process employs a cascaded arrangement of the following as the basic building blocks;

- A filter network.
- And it's respective turning circuit (the output of the filter)



.Fig 3.2 Diagram of a Filter Network

The design of the audio graphic equalizer circuitry made use of IC known as BA3812L. The BA3812L is a five point equalizer IC that has most of the filtering and amplification. Components integrated into one IC. The equalizer circuit is divided into five bands and each band is arranged as shown above in Fig 3.2. The design of each band is such that when a signal is fed into the circuit, it passes through a bypass capacitor that blocks the D.C signal and allow A.C signal to go through the capacitor into the circuit. The signal is fed into the transistor that is configured in a common collector mode, this gives a reasonable current gain but provide less than or nearly equal to unity voltage gain. The output current gain equation is given as: for the diagram shown above in Fig 3.2

$$\frac{I_E}{I_B} = \frac{I_E}{I_C} \times \frac{I_C}{B} = \frac{\beta}{\alpha} = \frac{\beta}{\beta/(1+\beta)} = (1+\beta) \ldots\ldots\ldots\ldots\ldots\ldots (1.1)$$

The current flow in various paths in the circuit is given as:

$$I_E = I_B + I_C = I_B + \beta\, I_B = (1+\beta) I_B \ldots\ldots\ldots\ldots\ldots\ldots\ldots (1.2)$$

Therefore, output current = $(1+\ \beta)$ × Input current.

The resistor $R_o\, and\ R_1$ act as biasing resistor to the transistor in order for it to operate properly and to be able to maintain constant voltage gain of unity. The two externally connected capacitors C_o *and* C_1*, are connected* in parallel and they form a filtering network and frequency determining circuit with $R_o\ and\ R_1$ and thereby determine the centre frequency of the band.

The band by-pass filters are practically the two-pole active RC type with main purpose of isolating specific frequency ranges for the modification of amplitude response characteristic.

A common parameter employed in the design of filter related network is the voltage transfer function, where the general from of a network function is given as:

$$H_{(s)} = \frac{a_n s^n + a_{n-1} s^{n-1} + a_{n-2} s^{n-2} + \cdots \ldots + a_0}{b_m s^m + b_{m-1} s^{n-1} + b_{m-2} s^{m-2} + \cdots \ldots + b_0} \qquad \ldots\ldots\ldots\ldots\ldots\ldots (1.3)$$

Where $a_n \neq 0, b_m \neq 0$ and all the co-efficient a_i *and* b_i are real.

The basic filter network used is based on single amplifier biquad topologies. The negative feedback topology employed is so called because the RC network is associated with the operation amplifier provides a feedback path to the negative input terminal of the operational amplifier, the bearing on inverting amplifier structure.

Hence the function $^{v_{0(s)}}/_{v_{0(s)}}$ (voltage gain) of the biquad from nodal analysis is given as:

$$T_{(s)} = \frac{V_0}{V_m} \frac{-\frac{1}{R_0C_1}s}{S^2 + s + \left(\frac{1}{R_0C_0} + \frac{1}{R_1C_0}\right) + \frac{1}{R_0R_1C_0C_1}} \ldots\ldots\ldots\ldots\ldots (1.4)$$

Where the general form of the transfer function of a second order band-pass filter is given as

$$T_{(s)} = K\frac{s}{s^2 + as + b} \ldots\ldots\ldots\ldots\ldots (1.5)$$

$$T_s = K\frac{s}{s^2 + \frac{wp}{qp} \times s + w_p^2} \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots (1.5_b)$$

Where w_p is the pole frequency and $Q_{p,}$ the pole quality factor.

In addition, the steady-state transfer function $H(jw)$of any two-pole band pass filter can be expressed as

$$H(jw) = \frac{M_0}{1 + JQ\left(\frac{F}{F_0} - \frac{F_0}{F}\right)} \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots (1.6)$$

Where $M_0$the maximum is gain within the band, and $F_0$is called geometric centre frequency. (In passive RCL circuits, F_0 is called resonant frequency). The parameter Q,(known ass the quantity factor) is a measure of selectivity or sharpness of the filter.

The amplitude response, $m_{(w)}$ can be determined as

$$M_{(w)} = \frac{M_0}{\sqrt{1 + Q^2\left(\frac{F}{F_0} - \frac{F_0}{F}\right)^2}} \dots\dots\dots\dots\dots\dots\dots (1.7)$$

The relative decibel response $M_{dB}(w)$ can be expressive as

$$M_{dB}(w) = 20\log_{10}\frac{M_{(w)}}{M_0} = 20\log_{10}\frac{1}{\sqrt{1 + Q^2\left(\frac{F}{F_0} - \frac{F_0}{F}\right)^2}} \dots\dots\dots\dots\dots (1.8)$$

OR

$$M_{dB}(w) = -10\log\left(1 + Q^2\left(\frac{F}{F_0} - \frac{F_0}{F}\right)^2\right) \dots\dots\dots\dots\dots (1.8_b)$$

In specifications of the filtering requirements, the concept of attenuation is often used. Assume that the maximum pass-band amplitude response for a certain filter isM_0, and the amplitude response at some arbitrary frequency w is$M_{(w)}$. The relative attenuation, $\alpha_{dB}(w)$ measured in decibels (dB) is defined as:

$$\alpha_{dB}(w) = 20\log_{10}\left(\frac{M_0}{M_w}\right) \dots\dots\dots\dots\dots\dots\dots\dots\dots (1.9)$$

OR

$$\alpha_{dB}(w) = -20\log_{10}\left(\frac{M_0}{M_w}\right) \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots (1.9_b)$$

Note that this attenuation is related to the maximum level of transmission in the pass-band.

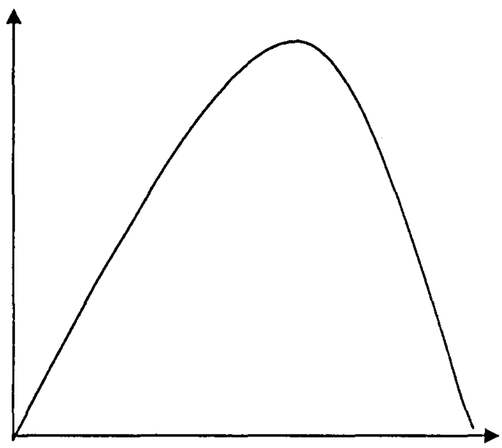


Fig 3.2 the illustrations of band pass filter response.

Let $F_L and F_h$ represent frequencies on low and high sides, i.e. lower and upper frequencies respectively, at which the response is $1/\sqrt{2}$ times the peak response (-3.01 dB down). The band width is defined as:

$$B = F_h \quad F_L \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots (2.0)$$

The frequency F_0 is the geometric centre frequency

$$F_0 = \frac{1}{2\pi\sqrt{R_0 R_1 C_0 C_1}} \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots (2.1)$$

The frequency Q is known as the quality factor of the filter.

$$Q = \sqrt{\frac{C_1 R_1}{C_0 R_0}} \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots (2.2)$$

The upper and lower frequency can be expressed as:

$$F_L = \frac{1}{2\pi R_1 C_1} \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots (2.3)$$

$$F_h = \frac{1}{2\pi R_0 C_0} \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots (2.4)$$

Where R_1 *and* $C_1$the resistor and capacitor on the low are pass side, R_0 *and* C_0 being the resistor and capacitor on the pass side respectively.

3.3.1 THE DESIGN SPECIFICATIONS

During the process , the given frequency range of $86H_z$ to $13KH_z$ was systematically divided into five bands, thus requiring five cascaded two-pole active band-pass filters.

In order to ensure that there are no rejected bands within the frequency range, the respective amplitude response from the band-pass filter virtually overlap at the upper and lower ends of their respective pass bands within the spectrum.

The lower frequency of the first filter at the input of the cascaded network is fixed at $86H_z$, and the upper frequency of the fifth filters fixed at approximately $13KH_z$, thus meeting he desired range.

Hence, appropriate resistor and capacitor values were used in obtaining the upper and lower frequencies for each biquad in the network. From equation 2.3 and 2.4, we have

<u>First Band-pass Filter</u>

$$F_L = \frac{1}{2\pi R_1 C_1} = \frac{1}{2\pi \times 68 \times 10^3 \times 0.027 \times 10^{-6}} = 86.69 H_z \approx 86 H_z$$

$$F_h = \frac{1}{2\pi R_0 C_0} = \frac{1}{2\pi \times 1.2 \times 10^3 \times 1 \times 10^{-6}} = 132.63 H_z$$

<u>Second Band-Pass Filter</u>

$$F_L = \frac{1}{2\pi R_1 C_1} = \frac{1}{2\pi \times 68 \times 10^3 \times 0.0082 \times 10^{-6}} = 285.43 H_z$$

$$F_h = \frac{1}{2\pi R_0 C_0} = \frac{1}{2\pi \times 1.2 \times 10^3 \times 0.33 \times 10^{-6}} = 401.9 H_z$$

<u>Third Band-Pass Filter</u>

$$F_L = \frac{1}{2\pi R_1 C_1} = \frac{1}{2\pi \times 68 \times 10^3 \times 0.002710^{-6}} = 866.86 H_z$$

$$F_h = \frac{1}{2\pi R_0 C_0} = \frac{1}{2\pi \times 12 \times 10^3 \times 0.110^{-6}} = 1326.29 H_z$$

<u>Forth Band-Pass Filter</u>

$$F_L = \frac{1}{2\pi R_1 C_1} = \frac{31}{2\pi \times 68 \times 10^3 \times 820 \times 10^{-12}} = 2854.29 H_z$$

$$F_h = \frac{1}{2\pi R_0 C_0} = \frac{1}{2\pi \times 1.2 \times 10^3 \times 0.03310^{-6}} = 4019.06 H_z$$

<u>Fifth Band-Pass Filter</u>

$B = F_h - F_L = 401.9 - 285.43$

$= 116.47H_z$

$$F_0 = \frac{1}{2\pi\sqrt{R_0 R_1 C_0 C_1}}$$

$= \frac{1}{2\pi\sqrt{68\times10^3\times1.20\times10^3\times0.0082\times10^{-6}\times0.33\times10^{-6}}} = 338.69H_z$

$Q = \sqrt{\frac{C_1 R_1}{C_0 R_0}}$

$= \sqrt{\frac{68\times10^3\times0.0082\times10^{-6}}{12\times10^3\times0.33\times10^{-6}}} = 1.187$

Third Filter

$B = F_h - F_L = 1326.29 - 866.86$

$= 459.43H_z$

$F_0 = \frac{1}{2\pi\sqrt{68\times10^3\times1.2\times10^3\times0.1\times10^{-6}\times0.002\times10^{-6}}}$

$= 1072.24H_z$

$Q = \sqrt{\frac{68\times10^3\times0.0027\times10^{-6}}{1.2\times10^3\times0.1\times10^{-6}}}$

$= 1.24$

Fourth Filter

$B = 4019.06 - 284.29$

$= 1164.77 H_z$

$$F_0 = \frac{1}{2\pi\sqrt{68\times10^3\times1.2\times10^3\times0.033\times10^{-6}\times820\times10^{-12}}}$$

$= 33386.97 H_z$

$$Q = \sqrt{\frac{68\times10^3\times820\times10^{-12}}{1.2\times10^3\times0.033\times10^{-6}}}$$

$= 1.187$

Fifth Filter

$B = 13262 - 8668.57$

$= 4594.33 H_z$

$$F_0 = \frac{1}{2\pi\sqrt{68\times10^3\times1.2\times10^3\times0.01\times10^{-6}\times270\times10^{-12}}}$$

$= 10722.48 H_z$

$$Q = \sqrt{\frac{68\times10^3\times270\times10^{-12}}{1.2\times10^3\times0.01\times10^{-6}}}$$

$= 1.24$

Similarly, the actual transfer function for each of the biquads can be from obtaining the function $V_0(s)/V_{in}(s)$ from the nodal analysis and substituting for the capacitance and resistance component values.

Therefore,

For The First Band

$$T_{(s)} = \frac{V_0}{V_{IN}} = -\frac{\frac{1}{R_1 C_2} S}{S^2 + S\left[\frac{1}{R_1 C_1} + \frac{1}{R_2 C_1}\right] + \frac{1}{R_1 R_2 C_1 C_2}}$$

$$= \frac{\frac{1}{1.2\times10^3\times100\times10^{-12}} S}{S^2 + S + \left[\frac{1}{1.2\times10^3\times1\times10^{-5}} + \frac{1}{6.8\times10^3\times1\times10^{-6}}\right] + \frac{1}{1.2\times10^3\times6.8\times10^3\times100\times10^{-12}\times1\times10^{-6}}}$$

$$= \frac{-833333.33S}{s^2 + 980S + 1.25\times10^8}$$

From equation $1.5_a + 1.5_b$

The gain $|K| = 833333.33$

And the pole frequency $W_p - \sqrt{1.25 \times 10^8} = 11180$radians

The pole Q, $Q_p = \frac{11180}{980} = 11.4$

Second Band

$$T_{(s)} - \frac{V_0}{V_{IN}} - \frac{-\frac{1}{R_1 C_2} S}{S^2 + S\left(\frac{1}{R_1 C_1} + \frac{1}{R_2 C_1}\right) + \frac{1}{R_1 R_2 C_1 C_2}}$$

$$= \frac{-\frac{1}{1.2\times10^3\times1000\times10^{-12}} S}{S^2 + S\left(\frac{1}{1.2\times10^3\times0.33\times10^{-6}} + \frac{1}{6.8\times10^3\times0.33\times10^{-6}}\right) + \frac{1}{1.2\times10^3\times5.8\times10^5\times1000\times10^{-12}\times0.33\times10^{-6}}}$$

$$= \frac{-833333335S}{S^2 + 2970.88S + 3.7136\times10^8}$$

The gain, $|K| = 833333.33$

And the pole frequency $W_{p=\sqrt{3.7136\times10^8}} = 19270.7$ radians

The pole Q, $Q_p = \frac{19271.7}{2970.88} = 6.5$

Third Band

$$T_{(s)} = \frac{V_0}{V_{IN}} = \frac{\frac{1}{R_1 C_2} S}{S^2 + S\left(\frac{1}{R_1 C_1} + \frac{1}{R_2 C_1}\right) + \frac{1}{R_1 R_2 C_1 C_2}}$$

$$= \frac{-\frac{1}{1.2\times10^3\times1000\times10^{-12}}}{S^2+\left(\frac{1}{1.2\times10^3\times0.088\times10^{-6}}+\frac{1}{6.8\times10^3\times0.088\times10^{-6}}\right)S+\frac{1}{1.2\times10^3\times6.8\times10^3\times1000\times10^{-12}\times0.088\times10^{-6}}}$$

$$= \frac{-833333.33S}{S^2+9803.9S+2255\times10^9}$$

The gain, $|K| = 833333.33$

And the pole frequency $W_p = \sqrt{1.22\times10^9} = 35007$ radians

The pole Q, $Q_p = \frac{35007}{9803.9} = 3.57$

Fourth Band

$$T_{(s)} = \frac{V_0}{V_{IN}} = \frac{-\frac{1}{R_1 C_2} S}{S^2+\left(\frac{1}{R_1 C_1}+\frac{1}{R_2 C_1}\right)S+\frac{1}{R_1 R_2 C_1 C_2}}$$

$$= \frac{-\frac{}{1.2\times10^3\times1000\times10^{-12}}S}{S^2+\left(\frac{1}{1.2\times10^3\times0.033\times10^{-6}}+\frac{1}{6.0\times10^3\times0.033\times10^{-6}}\right)S+\frac{1}{1.2\times10^3\times6.0\times10^3\times1000\times10^{12}\times0.033\times10^{-6}}}$$

$$= \frac{-833333.33S}{S^2+29708.85S+3.71361\times10^9}$$

The gain, $|K| = 833333.33$

And the pole frequency, $W_p = \sqrt{3.71361 \times 10^9} = 60939.39$ radians

The pole Q, $Q_p = \frac{60939.39}{29708.85} = 2.05$

Fifth Band

$$T_{(s)} = \frac{V}{V_{IN}} = \frac{-\frac{1}{R_1 C_2} S}{S^2 + \left(\frac{1}{R_1 C_1} + \frac{1}{R_2 C_1}\right) S + \frac{1}{R_1 R_2 C_1 C_2}}$$

$$= \frac{-\frac{1}{1.2\times10^3\times1000\times10^{-12}} S}{S^2 + \left(\frac{1}{1.2\times10^3\times0.01\times10^{-6}} + \frac{1}{6.8\times10^3\times0.01\times10^{-6}}\right) S + \frac{1}{6.8\times10^3\times1.2\times10^3\times0.01\times10^{-6}\times1000\times10^{-12}}}$$

$$= \frac{-833333.33S}{S^2 + 98039.22S + 1.225490196\times10^{10}}$$

The gain, $|K| = 833333.33$

And the pole frequency $w_p = \sqrt{1.225490196 \times 10^{10}} = 110701.86$ radians

The pole Q, $Q_p = \frac{11071.86}{98039.22} = 1.13$

The gain value is constant because only one amplifier is use to amplify the signals at various frequency band. The large gain value obtained from the computation bear close correspondence to the general assumption that an ideal operation al amplifier possesses an infinite gain.

3.3.2 The Turning Circuit

The turning circuit is connected in series with the capacitor C_0 which value dependence on the various value of centre frequency F_0 of the filter networks as shown in fig3.2. It consists of a potentiometer of relatively high resistance. The manual movement downwards of the sliding contact of potentiometer increases the resistance of the filter output, and thus reduces the amplitude of the output gain from the network. However, when the contact is subsequently moved upwards, the resistance to the network's output is reduced and hence an increase in the amplitude response is obtained.

The out put of this tone control circuit is then fed into the output amplifier of the equalizer circuit, so as to amplify the output signal before it goes out.

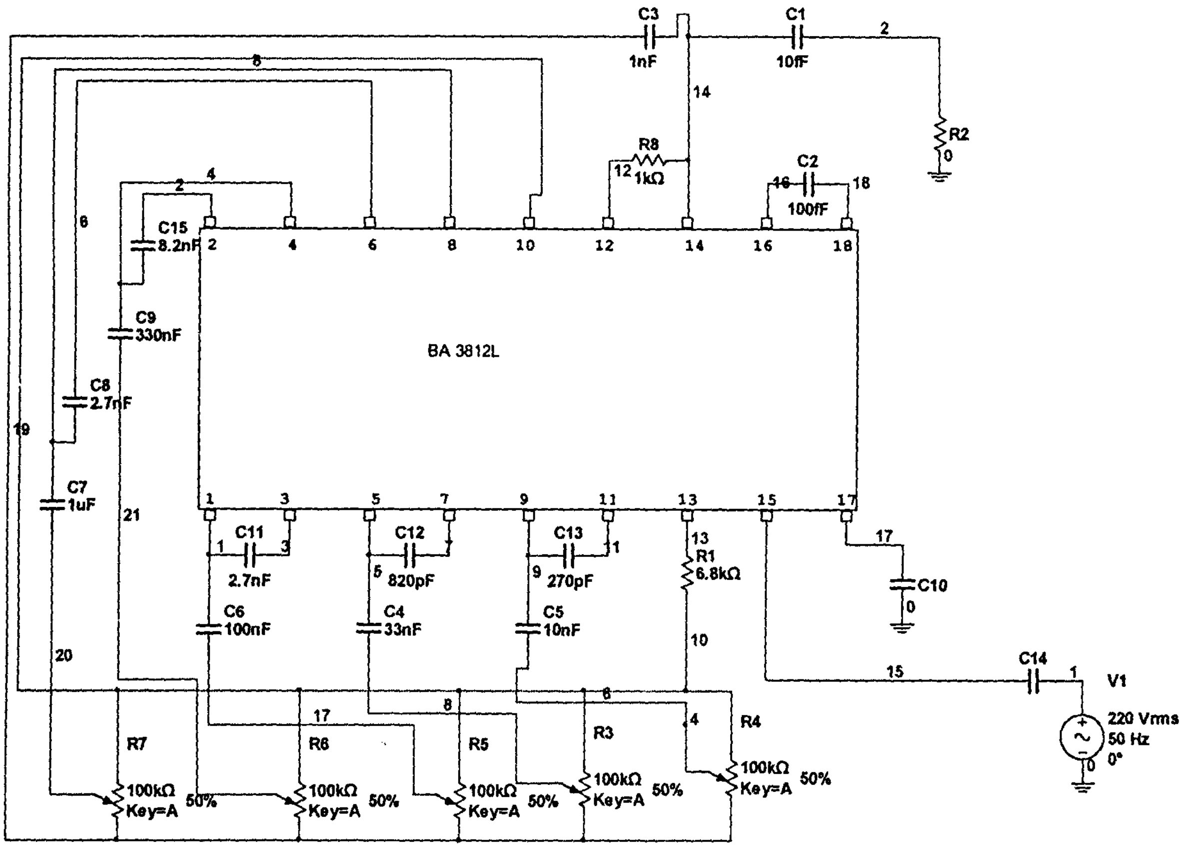


Fig3.4.The diagram of an Audio Graphic Equalizer Circuit

3.3.3 The Display Unit

The display unit of the audio equalizer is a module that is design to show graphic representation of the amplitude response of the equalizer circuits output. This consist LM339 IC, which is configured in an open collector mode. It inverts the signal that is fed into it from the equalizer circuit. The inverted signal is passed on to LM324 IC which is used as comparator to compare the inverted signal with the signal that is regulated using a variable resistor signal that is regulated using a variable resistor (potentiometer). When the result of the comparison shows, that the inverted signal is greater than the regulated signal, the light emitting diode comes on.

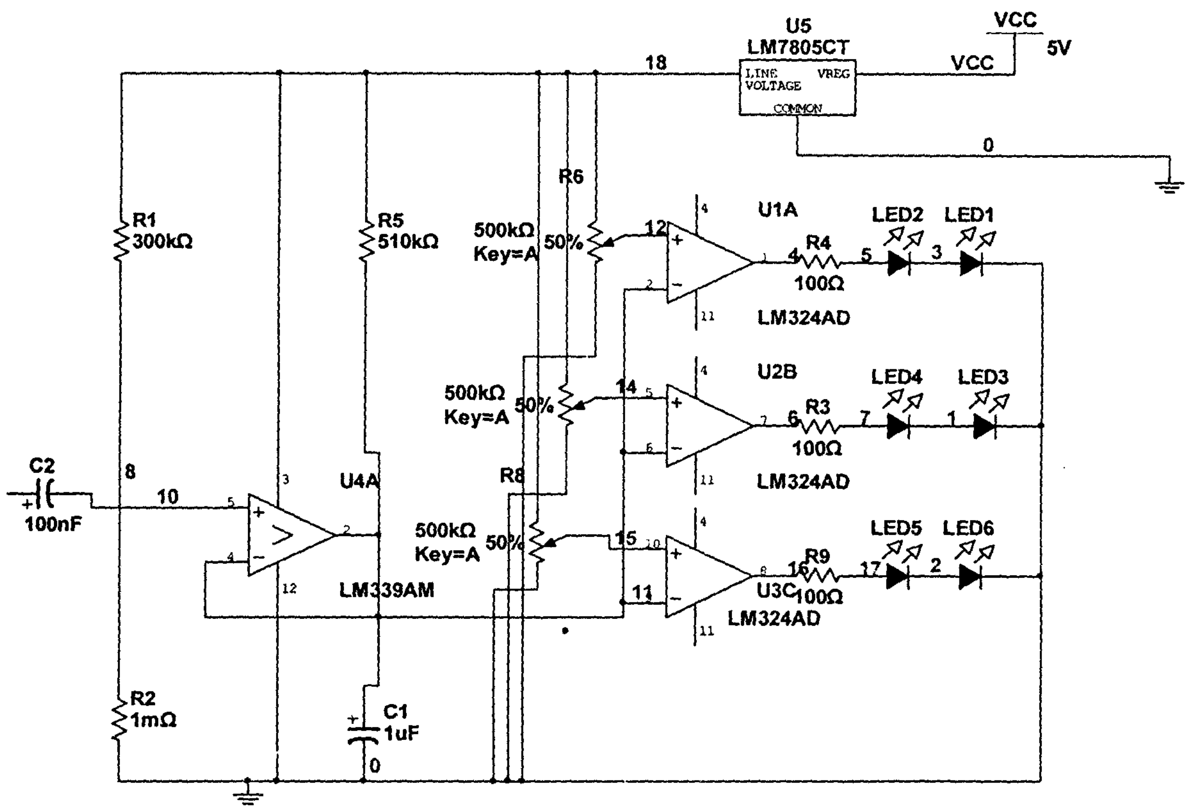


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Fig 3.5.Circuit Diagram of a Display Unit

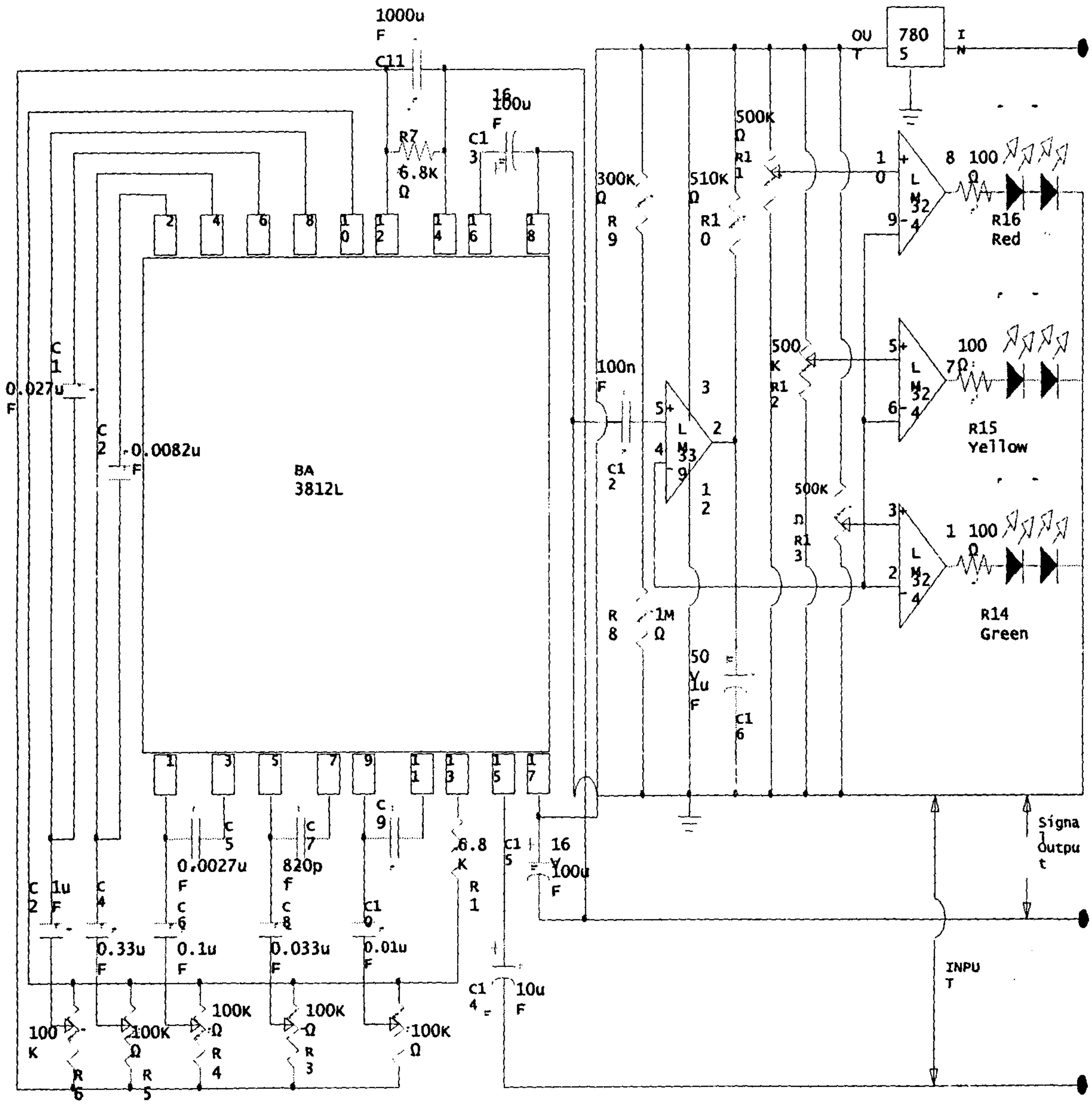

Fig 3.6. The Complete Circuit Diagram of Audio Graphic Equalizer with Level Display.

Table 3.1 List of components

S/NO	Resistors		Capacitors		ICs
1	R1	6.8KΩ	C1	0.027μF	BA3812L
2	R2	100K Ω	C2	1μF	LM339
3	R3	100KΩ	C3	0.0082μF	LM324
4	R4	100KΩ	C4	0.33μF	LM7805
5	R5	100KΩ	C5	0.0027μF	
6	R6	100KΩ	C6	0.1μF	
7	R7	6.8KΩ	C7	820pF	
8	R8	1MΩ	C8	0.033μF	
9	R9	300KΩ	C9	270pF	
10	R10	510KΩ	C10	0.01μF	
11	R11	500KΩ	C11	1000pF	
12	R12	500KΩ	C12	16V 10μF	
13	R13	500KΩ	C13	100μF	
14	R14	100Ω	C14	10μF	
15	R15	100Ω	C15	100μF	
16	R16	100Ω	C16	50V 1μF	

4 The Power Supply Unit

The ICs in the audio graphic equalizer circuit and the display circuit are fed with 5v dc from power supply unit. The unit consists of a 240v to 12v step down transformer with the output connected to a bridge rectifier to provide full wave rectification. Parallel connected capacitors serve as smoothening filters that reduce the ripple effect of the rectified output waveform. The 12v output of the bridge rectifier is regulated to 5v dc with the use of 7805 IC voltage regulator.

3.4.1. Ideal Transformer

Q is the flux in the core.

is the primary voltage.

is the secondary voltage.

is the primary e.m.f. and

is the secondary e.m.f.

Let N1 and N2 represent the number of the turns of the primary coils and the secondary coils respectively.

Then

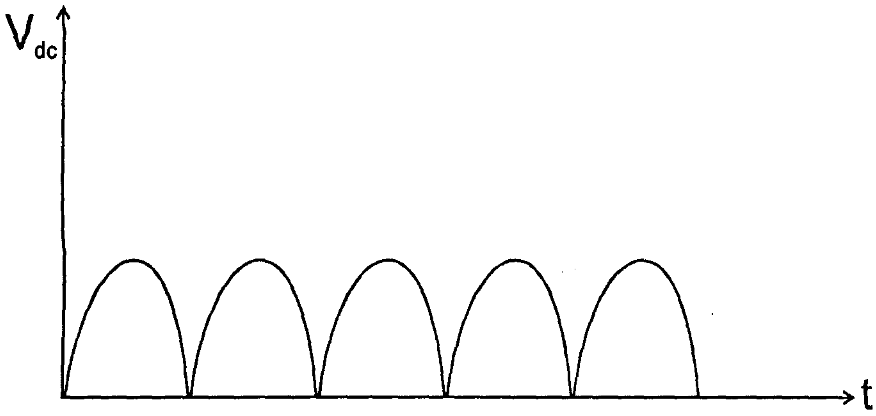

Fig 3.8 The rectifier output waveform

The full wave rectification efficiency can be calculated using

$$\eta = \frac{0.812\,R_l}{r_d + R_l}$$

Where R_l = load resistance and

r_d = diode resistance.

After full wave rectification, the relation below gives the D.C voltage with peak amplitude

Vdc = (Vrms√2 – 1.4)

For 15V rms input, the peak D.C voltage is therefore. Vdc = (12√2 – 1.4) =15V.

3.4.3 Filtering and Smoothing

The output waveform of fig 3.2 comprises of both A.C components.

The A.C component in a D.C power supply, a ripple voltage, is removed through filtering or smothered project work; the D.C voltage was smothered by 16V 1000μF and16V 10μF capacitors and fed in to the charger circuit.

3.4.4 Regulating Circuit

The regulating circuit enables the power supply unit to supply constant output voltage under varying input voltage or varying load current condition. An IC voltage regulator was employed to provide the regulated power supply. Table output voltage of 5V and 12V were obtained using 7805 with permissible load current of 1A respectively.

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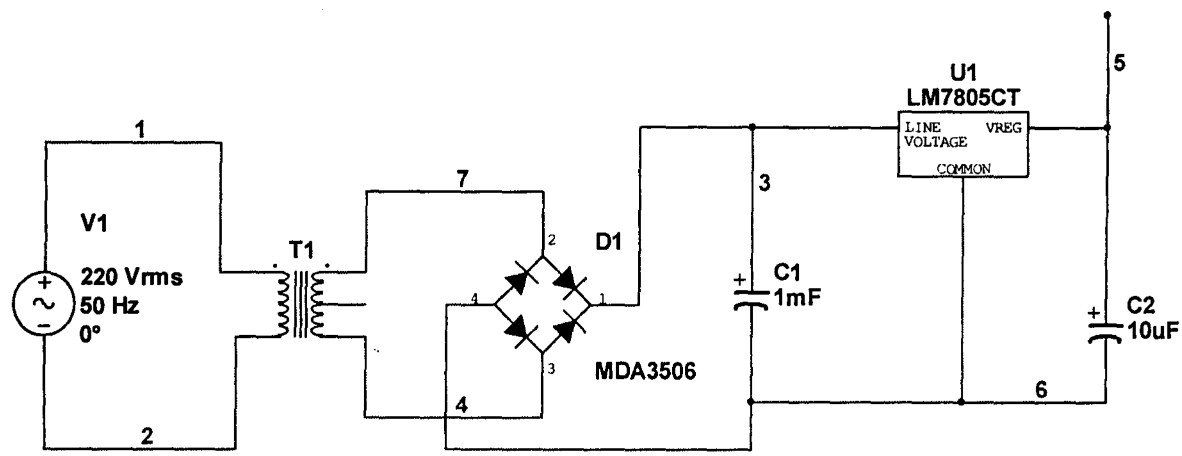

Fig 3.9. The Circuit Diagram of a Power Supply Unit

CHAPTER FOUR

4.1 CONSTRUCTION

The construction of this project was done in stages. In the first stage, the components were laid on the bread board to ensure that it is practicable and later transferred on a Vero board. The components were then assembled on the board and soldered according to circuit diagram. The power supply unit consists of a transformer, rectifier and a filter capacitor. All these except the transformer were placed on Vero board, while the transformer because of its weight was coupled to the casing of the system, and cables were used to take power from it to other component on the Vero board.

The equalizing stage and the display unit were all mounted on the Vero board. Flexible wires were used to supply voltage to various parts as demanded. The second stage was the coupling of the entire project to the casing. The circuit was cased with plastic casing with perforation to allow ventilation. Appropriate audio input and output terminals were provided, as well as an extension plug to plug into an A.C supply socket.

The low distortion, low noise and wide dynamic range BA3812L IC, LM339 and LM324 IC's constitute the "heart" of the system to which the various components of the main circuit e.g. Resistors and capacitors were connected to.

4.2 Test and Result

Test on the power supply were carried out using a digital multimeter. Each D.C output voltage level tested yielded satisfactory results. The two D.C levels obtained confirmed to acceptable tolerance levels. Accordingly they were: $+12.34/+12$V and $-12.40/-12$V.

A fidelity test for the audio output from the graphic equalizer, with an audio input from a compact disc player yielded acceptable result. The reproduced sound being faithful to the original sound was accompanied with reduced noise. In addition, the sliding contacts or turners for each of the centre frequencies (100Hz, 300H_Z, 1KH_Z, 3KH_Z and 10KH_Z) were adjusted accordingly. The high frequency ranges i.e. from 1KH_Z to 10KH_Z, were observed to be audibly pre-emphasized and de-emphasized when the contacts were slide upwards and downwards respectively. Hence the problem of high frequency background noise has been adequately taken care of. Furthermore, it was observed that on the manual adjustment of the sliding contacts upwards and downwards respectively, for the lower frequencies i.e. from 100H_Z to 300 H_Z, the bass like audio output was accordingly boosted and attenuated in a satisfactory manner.

CHAPTER FIVE

CONCLUSIONS AND RECOMMENDATION.

5.1. CONCLUSIONS

I have presented a straight-forward design of graphic equalizers with minimum-phase behaviour based on higher-order band shelving filters. Thanks to the high filter order, the inter-band influence is very small, although no special care has been taken to design filters with complementary edges except for a suitable definition of the cut-off frequencies, the resulting amplitude deviation in the transitional region between the bands is very low. Despite a slight increase at high frequencies, the amplitude ripple should be sufficiently low for most applications.

5.2. RECOMMENDATION

For future modification of this project, I recommend that Liquid Crystal Display (LCD) should be used for display of amplitude of the output signal in order to give proper graphical representation of the amplitude signal.

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