DESIGN AND CONSTRUCTION OF A 60Hz TO 16KHz AUDIO GRAPHIC EQUALIZER

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DECLARATION

I hereby declare that this project work was wholly and solely conducted by me; Mark T. Adzege; under the supervision of Mr. Paul Attah of Electrical and Computer Engineering Department, Federal University of Technology, Minna

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24/03/2000

Date

CERTIFICATION

I hereby certify that this work has been supervised, read and approved as meeting part of the requirements for the award of the degree of Bachelor of Engineering in the Department of Electrical and Computer Engineering, Federal University of technology, Minna, Niger State.

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DEDICATION

This project is dedicated to my Mum, Mrs. Becky Adzege and my siblings; Blessing, Moses and Ngiher.

ACKNOWLEDGEMENT

I am very grateful to everybody who helped me carryout this project. Thanks to my Mum, Mrs. Becky Adzege, for her relentless support, morally and financially; Engineer Abanyam of Instrumentation Department, Benue Cement Company PLC. Gboko, for providing the much needed assistance. I also recognize the contributions of my partner on this project, Mr. Anwana Amba.

I will also like to thank my H. O. D. Dr. Y. A. Adediran, my supervisor, Mr. Paul Attah and all the academic and non-academic staff of the Department for their unexceptionable impartation of knowledge to me.

Finally, I would like to give a big shout out to my friends, Anwana G; O. G Steve; James Longkwang; Mr. Pat; Gerard Dust; Don Pedro; Rahman; Kpam; Little Joe; Daze; Lungfa; Mike Kwaplong: You guys made my sojourn in F. U. T. Minna a memorable one.

TABLE OF CONTENT

PAGE

CHAPTER ONE

INTRODUCTION

CHAPTER TWO

SYSTEM DESIGN

CHAPTER THREE

PROJECT CONSTRUCTION

CHAPTER FOUR

CONCLUSION AND RECOMMENDATION

ABSTRACT

In the design of an audio system, the optimum performance is realized when the system can record and reproduce an audio signal at high fidelity. It is important to have an audio control unit on the system which can permit a user to adjust an audio signal to a level which suits the purpose of the recording, the environment where the signal is going to be reproduced, the listening preference of a listener and to correct any irregularities like noise that may have been introduced into the signal during recording. In other to vary the signal level for the above mentioned purposes, a circuit consisting of several bandpass filters in cascade needs to be designed. Each bandpass filter circuit is designed to filter its own range of frequencies in the audio frequency range of 20Hz to 20KHz. By so doing, an audio signal can be separated into two major tones, bass and treble. Bass tones refer to the low frequencies in the signal and treble refers to the high frequencies. An electronic circuit, which is designed for this purpose, is called an Audio Graphic Equalizer.

The work contained in this thesis is an effort to give an elaborate information about the design of an audio graphic equalizer.

CHAPTER ONE

INTRODUCTION

The basic functional unit of a graphic equalizer is an active bandpass filter. How many of these filter circuits comprise an equalizer depends on the number of bands the equalizer is divided into. Equalizers or generally filters are employed in electronic circuits where the separation of signals according to their frequencies is of vital importance. In sound systems, which is our primary focus here, equalizers are necessary to adjust the systems frequency response to suit the listeners desires, the acoustic environment in the home and any deficiencies in equipment or recordings. This is achieved by isolating particular bands of frequencies for increased or decreased emphasis by the output acoustical system (amplifier, speaker etc).

Having known their essentiality in electronic circuits, it is important to understand in technical jargon what a filter is. In general terms, a filter is any combination of passive or active elements designed to select or reject a band of frequencies. Passive elements here refers to resistors, inductors and capacitors why active elements are transistors or operational amplifiers.

This gives birth to two general classes of filters;

PASSIVE FILTERS: - These filters as said before, are composed of series or parallel combination of R, L and C elements. They are not very popular in audio frequency operations because they introduce distortions in the filter characteristics, a situation, which is very undesirable. This happens as a result of the deviation of the impedance of a practical inductor from its ideal value due to inherent resistance associated with its realization. The model of the inductor shown in Figure 1.1 will help to explain how this happens.

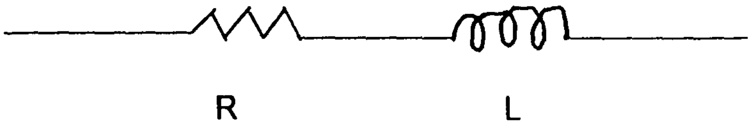


Fig. 1.1. An inductor model.

The quality factor Q_L of the inductor is given as

$$Q_L = \frac{\omega L}{R}$$

The larger the resistance R, the lower the quality factor and the further from ideal the inductor is. To minimize this deviation, inductors with very high values of Q_L have to be used. However, at frequencies above 1KHz high quality inductors tend to get bulky and expensive.

ACTIVE FILTERS: - Unlike passive filters, they are realized using resistors, capacitors and active devices which are usually operational amplifiers.

These devices are usually integrated thereby allowing active filters to provide the following advantages: -

A reduction in size and weight.
Increased circuit reliability.
Reduction in production cost.
Improvement in performance.
A reduction in parasitics and simpler design process

1.1 AIMS AND OBJECTIVES

The objective in any practical design is to obtain the most economical product that will meet the given specifications. The requirements for a filter are usually gain versus frequency specifications that must be met over a prescribed temperature range and for a given period of time. These are known as the end of life specifications.

Manufacturing tolerances and physical characteristics of the elements are selected so as to minimize the cost of the filter while still meeting the end of life specifications of the filter.

Some of the specifications of the 4 – band graphic equalizer designed here are discussed below: -

DYNAMIC RANGE: -As the input to an op-amp active filter is increased, the output continues to increase until eventually the output waveform becomes clipped as illustrated in figure 1.2 The op-amp is then said to be saturated.

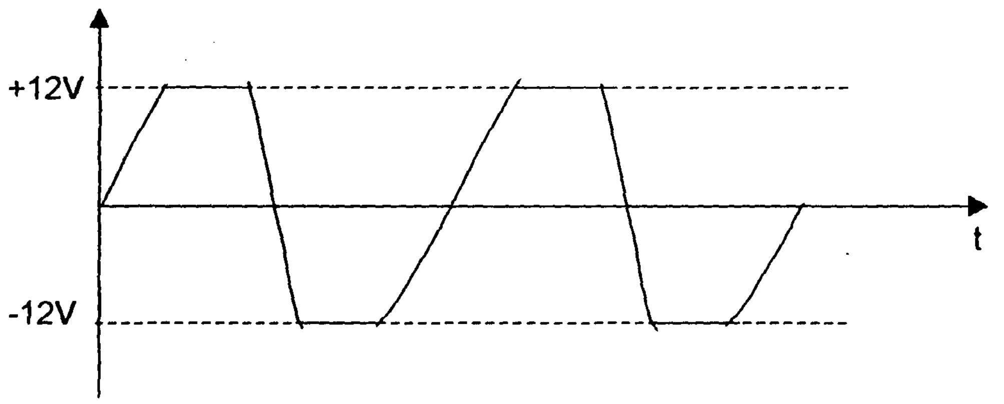


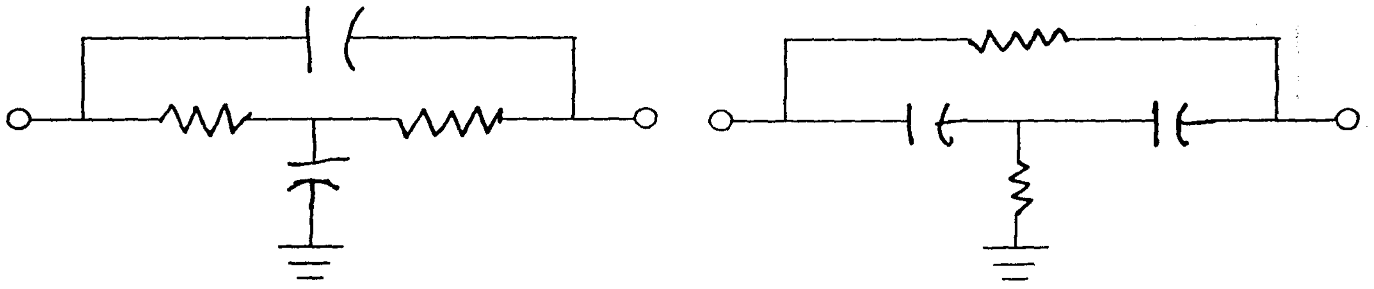
Fig 1.2 Clipping of a sine waves when the op-amp saturates.

The reason for the clipping is that the output voltage can not swing beyond the d.c power supply voltage. For instance if the power supply voltage is ± 12 volts, the output is certainly constrained to lie between + 12 and –12 volts. For a good filter, this type of distortion must be prevented by limiting the maximum input amplitude and keeping the minimum input signal level large enough to maintain the signal levels in the op-amp well above the internally generated noise voltages. The op-amp characteristic describing this limitation is known as the dynamic range. This is defined as the ratio of the maximum usable output to the noise output voltage. Now, since the output voltage depends on the filter function, dynamic voltage depends not only on the noise voltages and power supply voltage but also on the frequency characteristic of the active filter. For most active filters the dynamic range lies between 70dB and 100dB. In order to obtain maximum dynamic range, the minimum output voltage level of each of the biquads is kept the same.

The design of this is also aimed at keeping the frequency response between 60Hz and 16KHz with a reasonable flat response. This can be achieved by carefully selecting a suitable biquad topology and choosing elements to minimize circuit sensitivity.

1.2 METHODOLOGY

To realize the design of this band-pass filter circuit, a negative feed back biquad circuit has been chosen simply because it can be designed to provide a sensitivity which is among the lowest of all active RC biquad circuits. The circuit is realized by selecting one of a class of RC networks called bridge-T networks, examples of which are shown in Fig 1.3



(a) (b)

Fig 1.3 Bridge – T RC circuits.

BAND PASS CIRCUIT

The circuit shown in figure 1.4 is developed based on the negative feed back topology, which uses the bridge – T RC network of figure 1.3 (a).

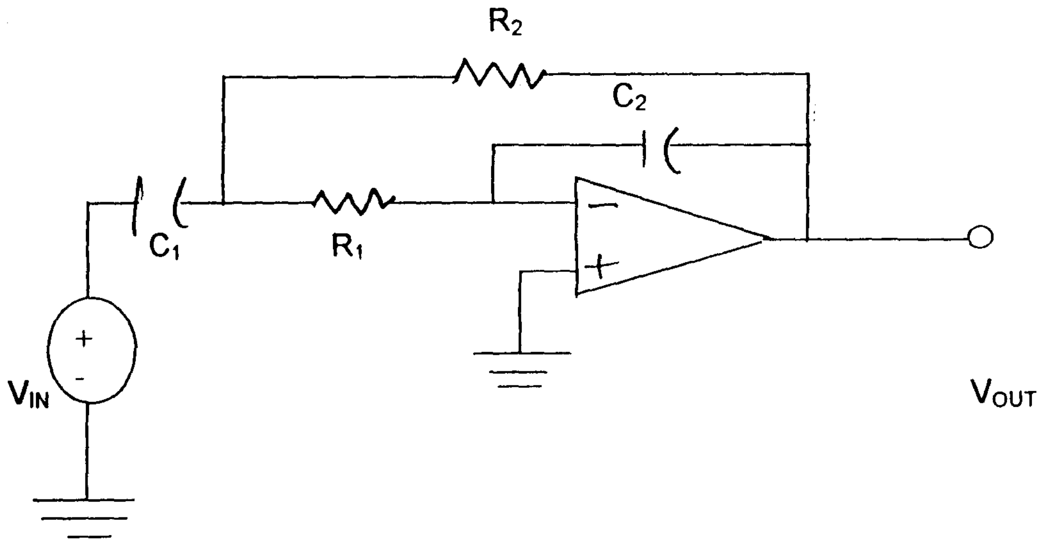


Fig 1.4 Negative feedback band pass circuit.

Assuming an ideal op-amp the transfer function of the active RC circuit is

$$T_{BP} = \frac{\frac{1}{R_1C_2}S}{S^2 + S\left(\frac{1}{R_1C_1} + \frac{1}{R_2C_1}\right) + \frac{1}{R_1R_2C_1C_2}}$$

This has the form of the second order band pass filter function where the pole frequency ω and pole Q are given by

$$\omega_P = \sqrt{\frac{1}{R_1R_2C_1C_2}}$$

$$Q_P = \frac{\sqrt{\frac{1}{R_1R_2C_1C_2}}}{\left(\frac{1}{R_2C_1} + \frac{1}{R_2C_2}\right)}$$

In order to obtain the band pass filter to pass all the frequencies between 60Hz and 16KHz, four of the above biquad circuits are combined to form the 4-band graphic equalizer circuit of Figure 1.5. This approach is known as the cascade topology.

In figure 1.5 the band circuit of figure 1.4 has been reduced to a simple block for illustrative purposes full details of the design are given in chapter two of this report.

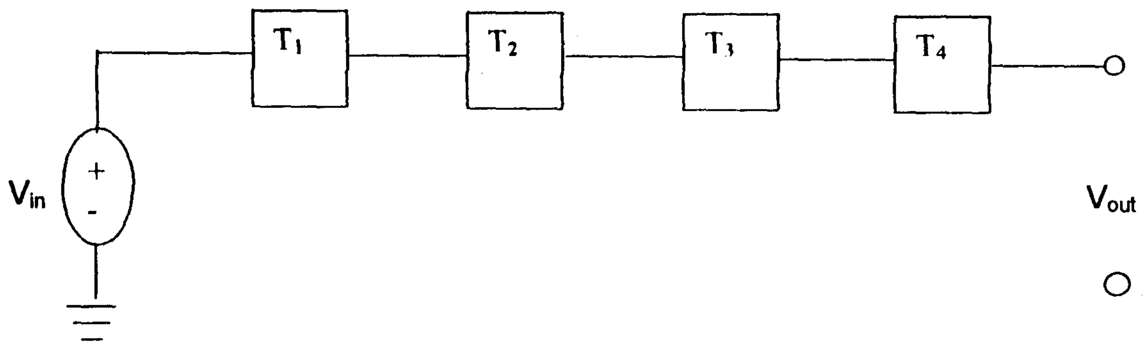


Fig. 1.5 The cascaded topology of four biquads.

The output impedance of each biquad in figure 1.5 is negligibly small compared with the impedance of the next biquad hence the output voltage from one block is not loaded down by the next block. These yields: -

$$V_o = T_1 T_2 T_3 T_4 V_{IN}$$

From which

$$\frac{V_o}{V_{IN}} = T_1 T_2 T_3 T_4$$

Thus the transfer function is the product of the individual transfer functions, provided that the input impedance of each network is very large compared with the output of the proceeding network.

One important advantage of the cascade approach is that the individual biquads are totally isolated so that any change in one biquad does not affect any other biquad.

1.3 LITERATURE REVIEW

Sound is recorded and produced for a wide variety of purposes. Music provides Entertainment, the spoken voice is recorded for business purposes (dictation, lectures and language training). Controls, in the form of filters (Audio graphic equalizers) are necessary to adjust the frequency level of the sound systems to suit the purpose of the recording and the acoustic environment where it is being reproduced.

Electrical elements have being used to make frequency selective filters since the early part of the 20th century. Modern filter design really began with the arrival of highly integrated circuit operational amplifiers in the early 1960's. Modern filters utilize op-amps in combination with RC feed back networks provide countless filter circuits with a wide range of frequency selective properties such as band pass, high pass, low pass and delay equalizer filters.

These days, the band pass characteristics of active filter is utilized in audio graphic equalizers to enhance the recording and reproduction of sound.

CHAPTER TWO

SYSTEM DESIGN

The system is divided into two parts: -

(1) The audio graphic equalizer circuit and

(2) The power supply circuit.

These will be discussed one after the other.

2.1 AUDIO GRAPHIC EQUALIZER CIRCUIT

As mentioned in section 1.4 of chapter one, the equalizer is made up of a cascade of 4-biquad circuits each with it's output connected to a tuning circuit shown in figure 2.1. The tuning circuit permits the listener to adjust the output level of each filter to the desired amplitude. Figure 2.2 shows the detailed design of the equalizer circuit.

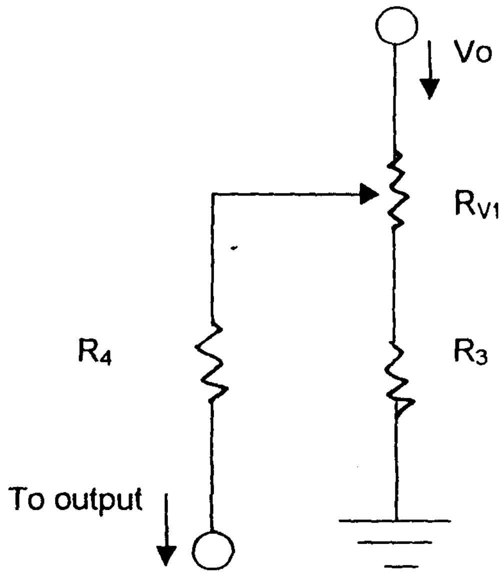


Fig. 2.1 Equalizer tuning circuit.

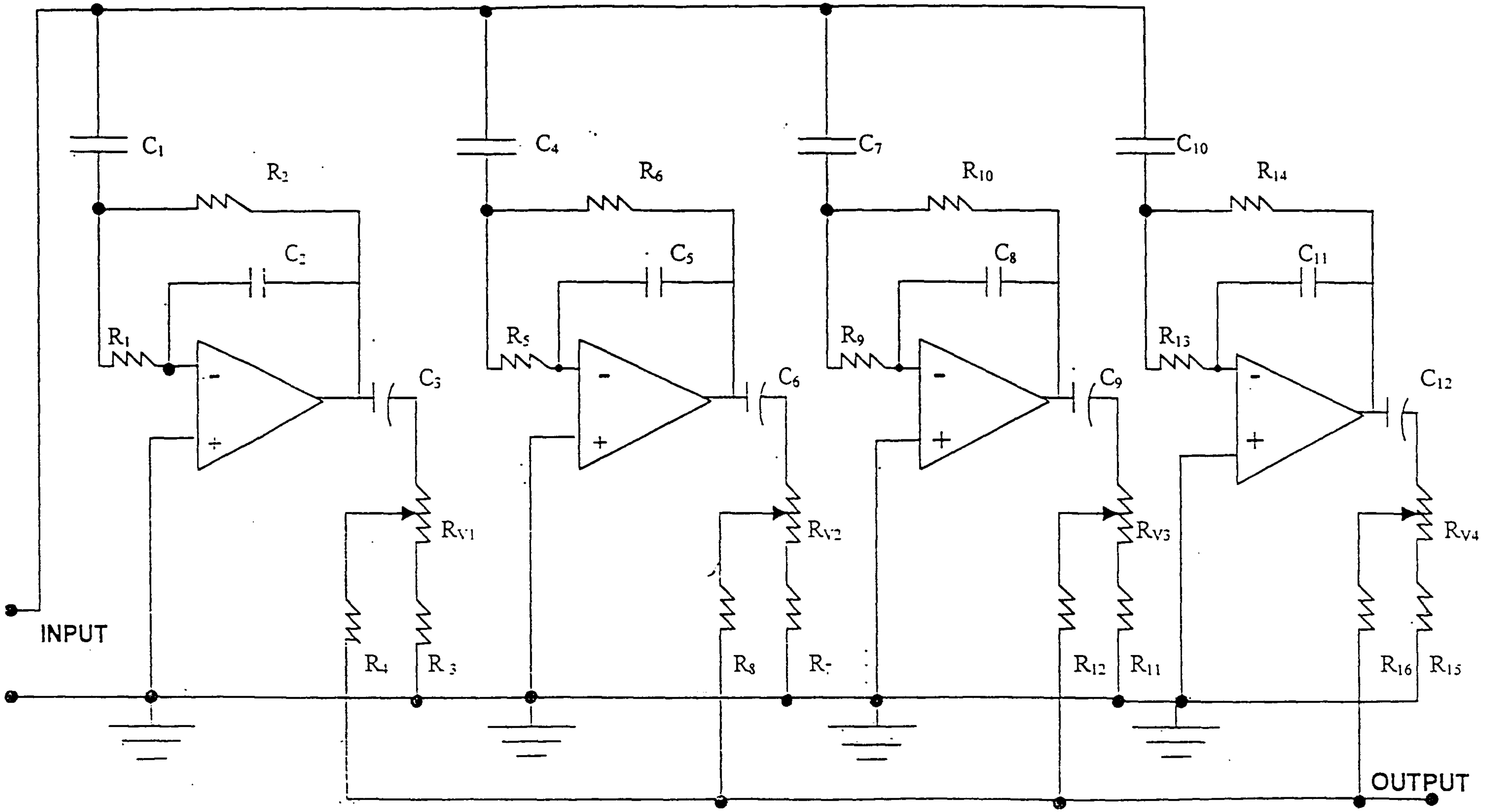


Figure 2-2 Circuit diagram of the audio graphic equalizer

2.2 DESIGN SPECIFICATIONS

Figure 2.3 shows the band pass response of a filter. f_L is the lower frequency and f_h is the upper frequency, the center is denoted as f_o. The bandwidth is defined as

$$B = f_h - f_L \qquad (2.1)$$

and

$$f_o = f_L f_h \qquad (2.2)$$

The quality factor Q is related to the center frequency and bandwidth by the relationship

$$Q = \frac{f_o}{B} \qquad (2.3)$$

The lower and upper frequencies can be expressed in terms of RC components as

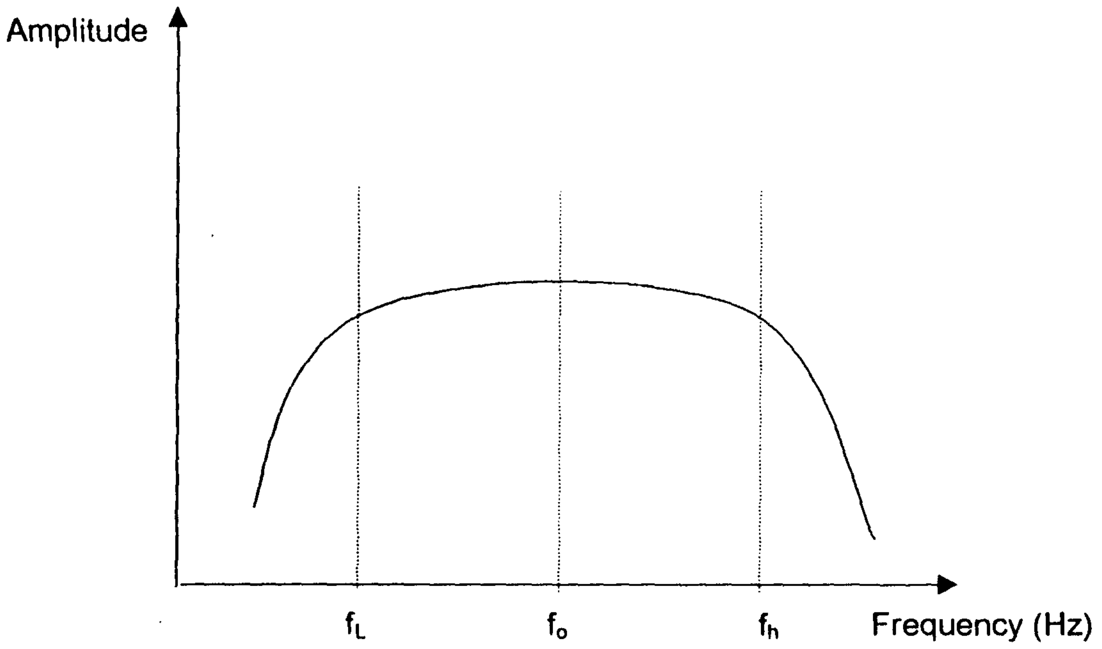
$$f_L = \frac{1}{2\Pi R_i c_i} \qquad (2.4)$$

and

$$f_h = \frac{1}{2\Pi R_f C_f} \qquad (2.5)$$

where R_i and C_i are the Resistor and capacitor values of the low pass filter while R_f and C_f are capacitor values for the high pass filter.

In order to adhere to the design specification of 60Hz – 16KHz frequency range, the RC components are selected so that the lower frequency of the first biquad is 60Hz and the upper frequency of the fourth biquad is 16 KHz. Figure 2.4 illustrates how the upper and lower frequencies of the intermediate biquads are selected from the audio frequency curve. As seen from figure 2.3, the amplitude of a band pass filter does not return to zero instantly at the frequency limits but rolls down gradually. As such the frequency limits of the biquads within the band should be selected carefully to take account of the amplitude roll off otherwise there would be elimination of some frequencies in the Band.



Fig, 2.3. Illustration of Band pass filter response

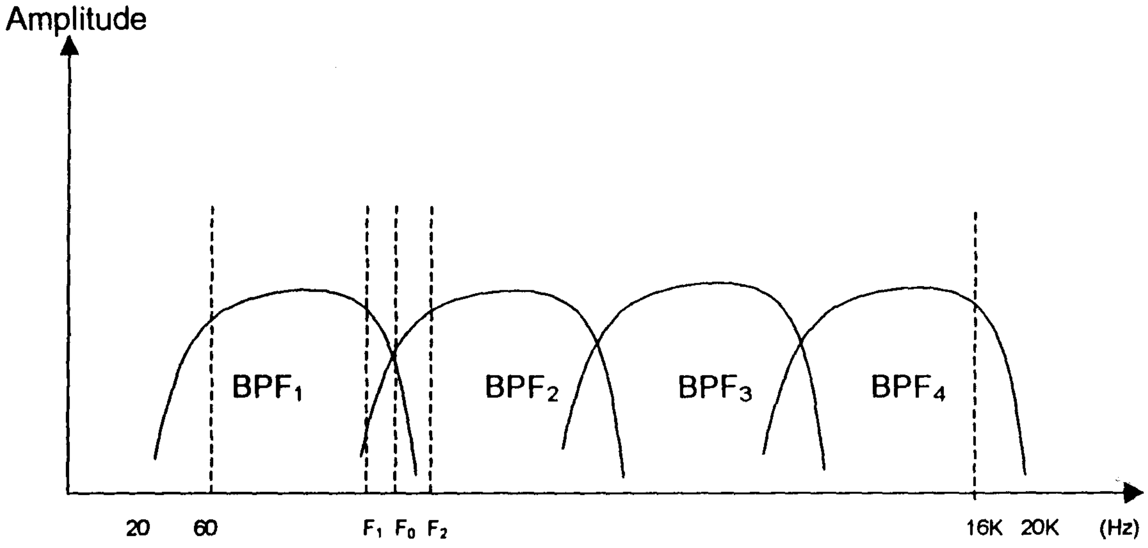


Figure 2.4 frequency response of the 4 biquad circuits.

Considering filter responses of filters one and two in figure 2.4, as an illustration the upper frequency of BPF$_1$, should be greater than f_o and the lower frequency of BPF$_2$ must be less than f_o for their frequency response to overlap properly.

2.2.1 COMPONENTS VALUES CALCULTIONS

By carefully selecting the lower and upper frequencies for each band, the values of the resistors used can be computed from equations 2.4 and 2.5 since specific values of capacitors are harder to find than resistors, the capacitors used here were selected based on their availability and their suitability for the circuit. The computations for each biquad are as follows: -

<u>First Biquad</u>

$f_l = 60Hz, \quad f_h = 4095Hz.$

From equations 2.4 and 2.5,

$$R_i = \frac{1}{2\Pi f_l C_i}$$

And

$$R_f = \frac{1}{2\Pi f_h C_f}$$

$$R_l = \frac{1}{2\Pi x60x330x10^9}$$

$$= 8038\Omega$$

$$R_2 = \frac{1}{2\Pi x4095x100x10^9}$$

$$= 388\Omega$$

<u>Second Biquad</u>

$f_L = 3995Hz, \quad f_h = 8080Hz$

$$R_5 = \frac{1}{2\Pi x3995x100x10^{-9}}$$

$$= 398\Omega$$

$$R_6 = \frac{1}{2\Pi x8080x27x10^{-9}}$$

$$= 730\Omega$$

<u>Third Biquad</u>

$F_L = 7,980Hz, \quad f_h = 12065Hz$

$$R_9 = \frac{1}{2\Pi x 7980 x 10 x 10^{-9}}$$

$$= 1994\Omega$$

$$R_{10} = \frac{1}{2\Pi x 12065 x 4.7 x 10^{-9}}$$

$$= 280\Omega$$

Fourth biquad

f_L = 11,965Hz, f_h = 16000Hz

$$R_{13} = \frac{1}{2\Pi x 11965 x 2.2 x 10^{-9}}$$

$$= 6046$$

$$R_{14} = \frac{1}{2\Pi x 16000 x 560 x 10^{-9}}$$

$$= 17763.$$

The bandwidth B, geometric center frequency f_o and quality factor Q are from the chosen frequencies using equations 2.1, 2.2 and 2.3 respectively.

First Biquad

$B = f_h - f_L$ = 4096 – 60 =4035Hz

$f_o = \sqrt{f_L f_h}$ = $\sqrt{4095x60}$

= 459.68Hz

$Q = \frac{f_o}{B}$ = $\frac{495.68}{4035}$ = 0.12

Similarly for second biquad,

B = 4033Hz, f_o = 9812.17Hz, Q = 2.40

And for the fourth biquad,

B = 4035Hz, f_o = 13,836.18Hz, Q = 3.43

The calculation of Q above show that it increases as the geometric frequency f_o increases. The transfer functions for each biquad can be obtained by substituting the RC values of each biquad into the transfer function expression for the biquad topology.

$$T(S) = \frac{V_o}{V_{IN}} = \frac{\frac{-S}{R_1 C_2}}{S^2 + S(1+1) + 1}$$

Thus for the first biquad we get

$$T_1(S) = \frac{-1250S}{S^2 + 8149S + 9.7125X10^6}$$

Comparing this result with the general expression for Transfer function of a Band pass filter;

$$T(S) = K\frac{S}{S^2 + aS + bS.}$$

Where K is the gain, $a = \frac{\omega}{Q_P}$

and

$b = w_p^2$

The pole frequency w_p = 9.7125 x 106

= 3116 radians

and the pole $Q_p = \frac{3116}{8149}$

= 0.38.

Similarly, for the second biquad,

$$T_2(S) = \frac{-92592.6S}{S^2 + 38698.6S + 1.26864 \text{ x } 10^9}$$

Thus, gain /K/ = 92592.6, ω_p = 5614 radians and Q_p = 0.92 for the third biquad.

$$T_3(S) = \frac{-106383S}{S^2 + 85714.3S + 3.7994x10^9}$$

Hence,

Gain, /K/ = 106383, ω_p = 61639.2 radians and pole Q_p = 1.15 for the fourth biquad.

$$T_4(S) = \frac{-297619S}{S^2 + 85714..3S + 3.7994x10^9}$$

Gain, /K/ = 297619

Pole frequency ω_p = 8717.8 radians and pole Q_p = 0.8606.

The results obtained from these calculations show very high gain values for the biquads. This conforms to the assumption that the gain of an ideal op-amp is infinite.

2.3 THE OPERATIONAL AMPLIFIER (MC14 58 CP1)

This is a dual low noise JFET OP AMP. Dual in the sense that it consists of two Op Amps embedded in one 8 pinned IC. For a four band equalizer two of these IC's have been used. Figure 2.5 shows the block diagram of the MC1458CPI IC.

2.4 COUPLING CAPACITORS

These are used to couple the output of the Op-amp to the tuning circuit. The value of the capacitor is selected such that the higher frequency bands use a smaller value of coupling of capacitors than the low frequency bands. This is because at low frequencies a larger value of the op-amp output appears across the capacitor. Hence the larger the capacitor the lower the voltage dropped across it as seen in the equation below.

$$Vc = \frac{1}{C}\frac{dI}{dt}$$

Coupling capacitors are primarily used to block any d.c signal which can introduce distortion in the performance.

2.5 THE TUNING CIRCUIT.

This, as shown in figure 2.1 is a potentiometer with a maximum resistance of 10KΩ for each of the biquads. When the movable arm is raised, the resistance is decreased and more of the output voltage from the op-amp is loaded to the output via the 1KΩ resistor. Lowering the arm increases the resistance and hence decreases the output voltage. A 120Ω resistor is connected in series with the 10KΩ variable resistor.

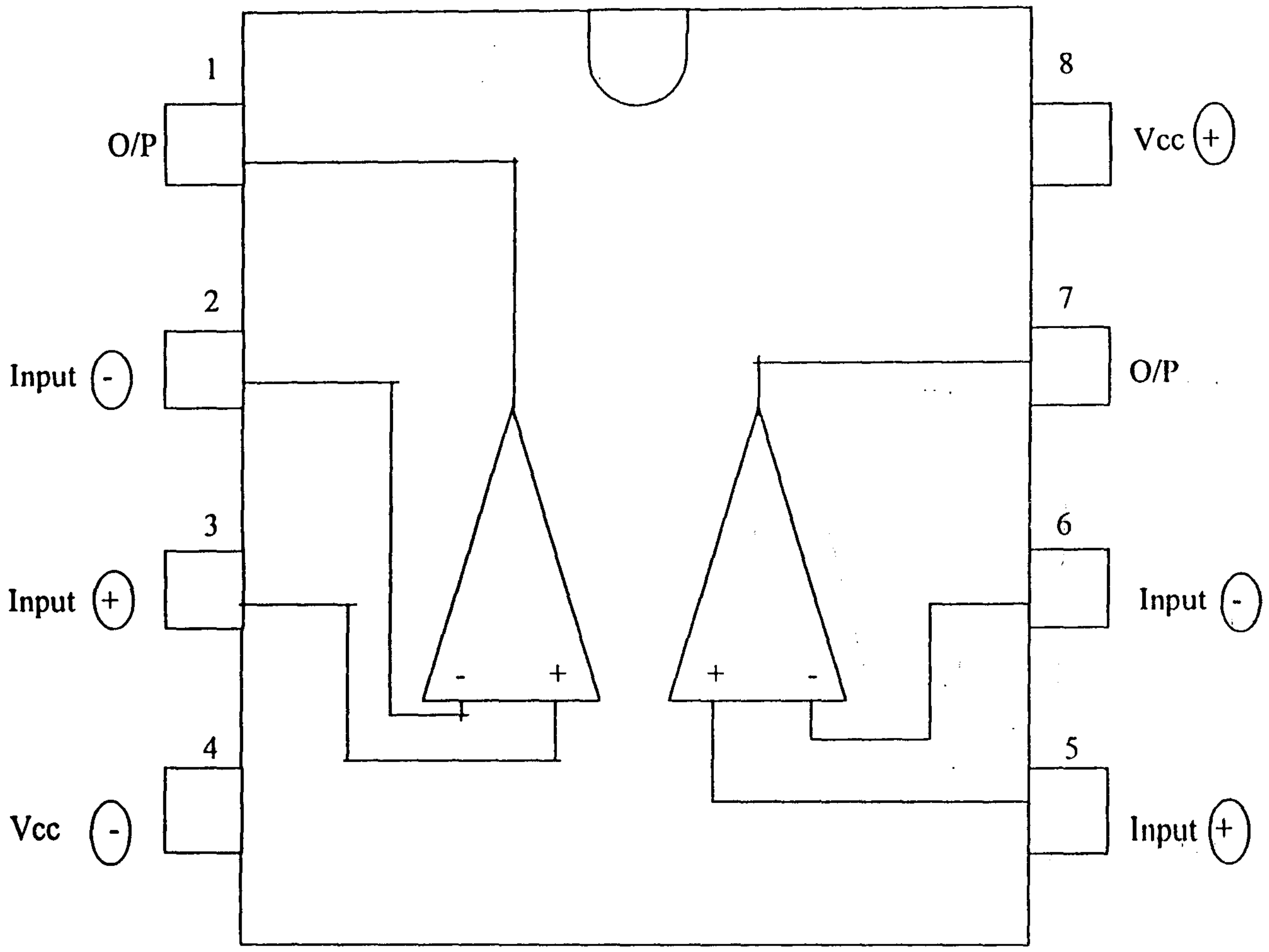


Figure 2.5 Block diagram of the MC1458CPI dual OP amp IC

2.6 THE POWER UNIT.

The circuit used for this design is called a full wave voltage doubler rectifier circuit. It consists of a 220 to 12 volts centretapped step down transformer whose secondary is connected to a full bridge rectifier network as shown in figure 2.6.

The output of the rectifiers has 1000μF/35V capacitors connected across them so as to reduce the ripples. The 39Ω resistors and the zener diodes provide voltage regulation for the circuits.

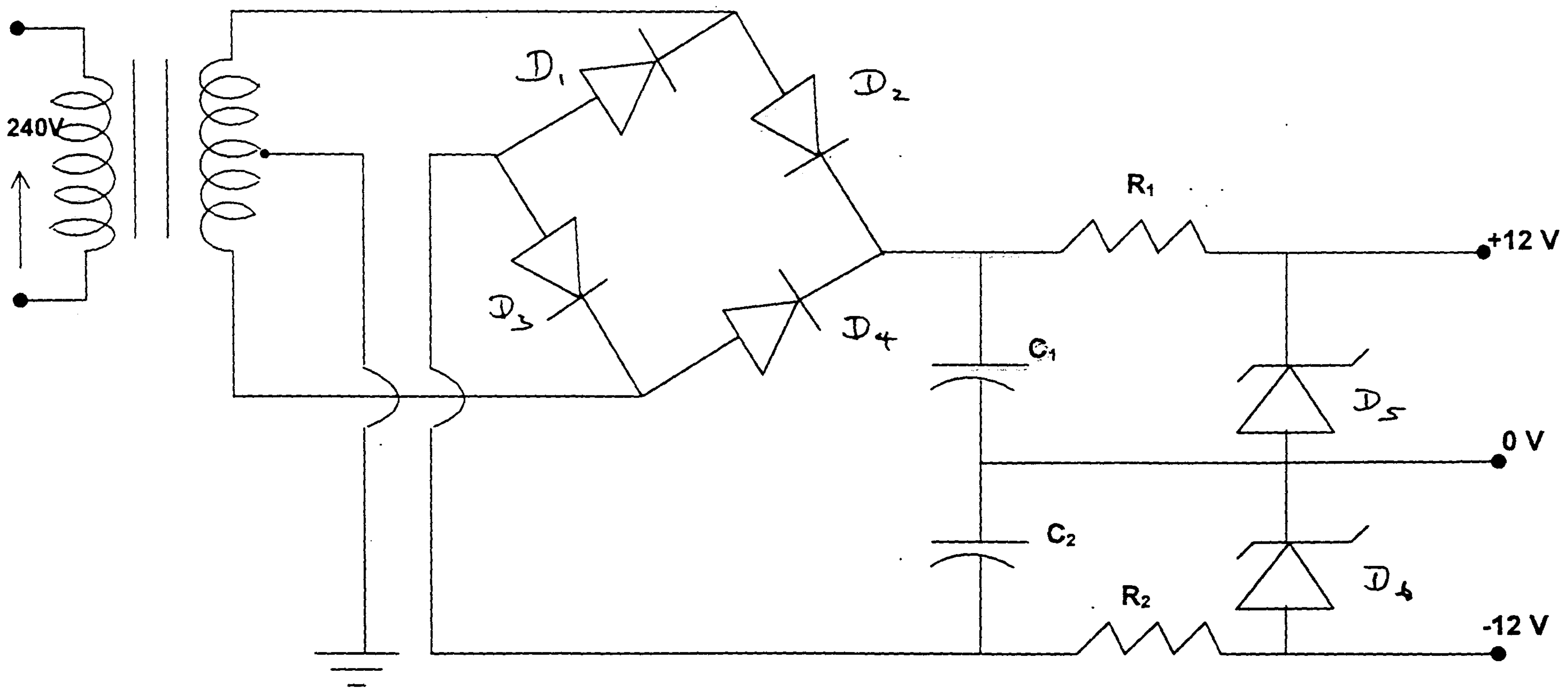


Fig 2.6 Power supply Circuit.

CHAPTER THREE

PROJECT CONSTRUCTION

Below is a list of all the components used during the construction. Not all the resistors have the same value as those obtained from design calculations, however, their tolerances are such that any errors which might arise from the differences would be very minimum.

Resistors

R_1 = 8k ohms

R_2 = 390 ohms

R_3 = 120 ohms

R_4 = 1k ohms

R_5 = 400 ohms

R_6 = 730 ohms

R_7 = 120 ohms

R_8 = 1k ohms

R_9 = 2k ohms

R_{10} = 2.8k ohms

R_{11} = 120k ohms

R_{12} = 1k ohms

R_{13} = 6k ohms

R_{14} = 17.8k ohms

R_{15} = 120 ohms

R_{16} = 1k ohms

$Rv_1 = Rv_1 = Rv_1 = Rv_4 = 10k$ ohms.

Capacitors

C_1 = 330 nF

C_2 = 100 nF

C_3 = 10 nF

C_4 = 100 nF

C_5 = 27 nF

C_6 = 2.2 nF

C_7 = 10 nF

C_8 = 4.7 nF

C_9 = 1 nF

C_{10} = 2.1 nF

C_{11} = 560 nF

C_{12} = 0.47 nF

Two MCI 1458CPI and dual OP-amps IC

Vero board

Plastic board, input and output sockets.

Instruments Used

- Soldering iron
- Soldering pump
- Digital multimeter
- Soldering lead
- Drilling machine

All the above components were soldered into the Vero board using the listed instruments. Plate 1 shows the layout of the circuit and plate 2 shows the finished construction of the equalizer.

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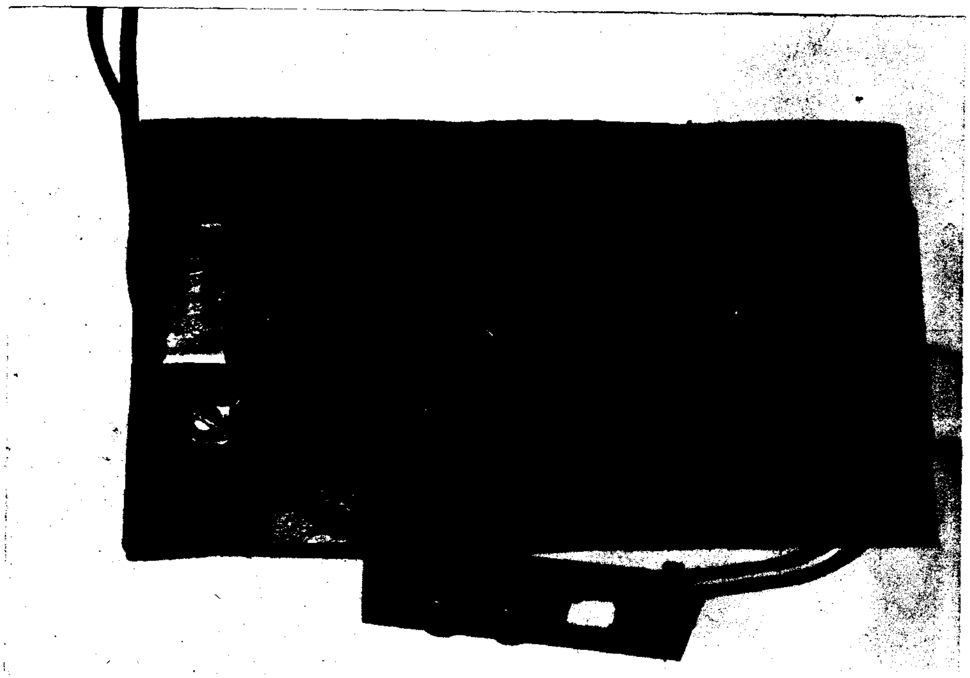

Plate 1. Layout of the equalizer and power supply unit on Vero board.

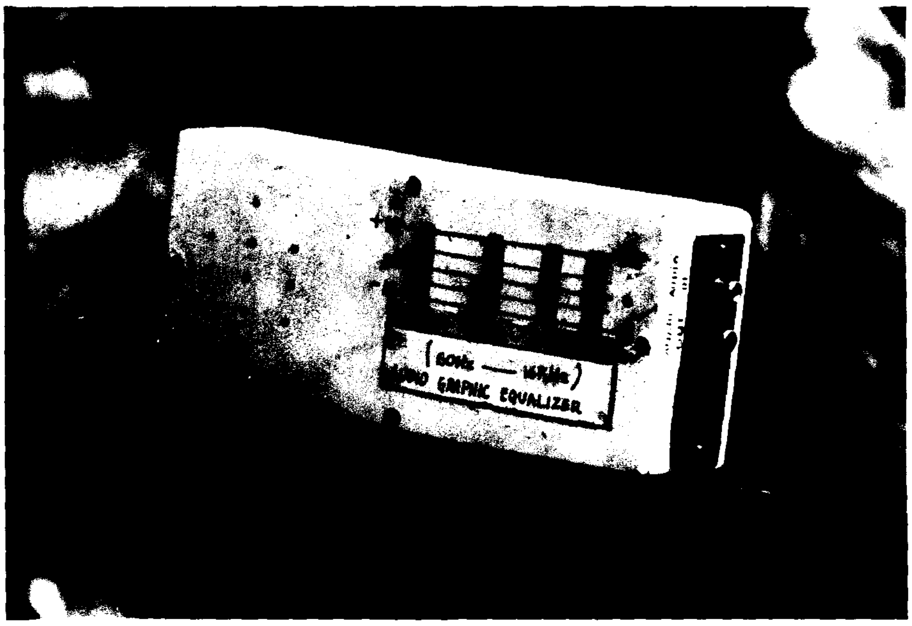

Plate 2. Complete construction of the equalizer.

3.1 TESTING AND RESULTS

The output of the power supply unit is confirmed by measuring the voltage across the zener diodes. This gives a result of +12 volts and –12 volts. The performance of the graphic equalizer is ascertained by the fidelity Test. The input to the equalizer is taken from the auxiliary output of a deck and the output from it is fed into an amplifier which is connected to speakers. The equalizer is then plugged on and the deck is put on play mode. By varying the resistance of the variable resistors of each band the frequencies in the output signal can be emphasized or de-emphasized according to the desired level.

If this output signal is fed into an oscilloscope the various fundamental frequencies of the signal can be seen on the screen. When the output level of each band is varied the response can be seen on the screen as the fundamental frequency concerned increases or decreases in amplitude.

CHAPTER FOUR

4.1 CONCLUSION

Results of the test show that an operational amplifier coupled to an appropriate RC network can be used to realize a band pass filter function. However, due to the negative feed back topology, the input to the equalizer is greatly attenuated by the system. Hence an amplifier is needed at the output to boost the signal to a level that can actuate a speaker. Test results show conformity with theoretical values as depicted by the high gain values of the design which bear close resemblance to the infinite values assumed in theoretical cases. The performance of the circuit without introducing noise to the signal is a confirmation of good design and proper functioning of the equalizer.

4.2 RECOMMENDATION

The performance of the band pass filter components in this project has been very satisfactory. However, there is always room for improvement. In the course of testing the equalizer, it was observed that even when the equalizer's power cord is not plugged on, the circuit still functions. This electrical anomaly is hard to explain due to limited knowledge of electronic circuit intricacies. The development however has some hidden benefits as it means an equalizer circuit can be designed such that it would not require an independent source of power as it would be powered by the signal it is meant to treat. This, if done, would enhance circuit simplicity and performance as well as reduce the cost of the circuit.

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