UNIVERSITY OF NAIROBI

FINAL YEAR PROJECT

TITLE: DESIGN OF A 20 WATT PUSH PULL AMPLIFIER DRIVEN BY A 6 BAND GRAPHIC EQUALIZER

PROJECT NO. 80

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Department of Electrical and Information Engineering
DEDICATION

To my parents

For always encouraging me to work hard any time I felt discouraged in life.
Acknowledgement

As a sincere and a committed Christian, I always try my level best to remember to acknowledge my God; whom I believe assists me in the accomplishment of all honorable and wholesome goals. My heartfelt thanks go out to my supervisor Mr. Ogaba, for his guidance throughout the development of this project.

I am also greatly indebted to all my lecturers in the department of Electrical and Information Engineering, for their intellectual nourishment and guidance up to my final year of study.

Lastly but not least, my gratitude goes to my mom and dad for their moral and financial support, without which my efforts would be futile.

GOD BLESSS YOU ALL

THANK YOU
Abstract

All audible information recorded for playback began as very low level electrical signals. For example, the signal levels at the outputs of microphones and audio tapes typically average a few millivolts. Such signal level voltages must be increased (amplified) to become useful. Amplifiers are required to accomplish this task.

Equalization is the process through which segments of audio frequency spectrum are manipulated. This may be done to compensate for the poor acoustic characteristics of a room or creative recording purposes. An equalizer is an electrical device that is used to adjust the frequency response of an audio signal. Equalizers are widely used in live music events where speakers and microphones operate in simultaneity.

Chapter one is the general introduction theory to amplifier and equalizers as well as their various types, characteristics and their applications.

Chapter two is built up with in-depth analysis of amplifier topologies, filter fundamentals and their application in graphic equalizer design.

Chapter three encompasses the singled out active filter method of design and various filter approximations method as well as their respective frequency responses. Equalizers are built around band pass filters, and thus great attention has been given to these filters.

Chapter four then dealt with design for both the amplifier circuits and the equalizer. It also tackles general circuit stability precautions and discussion about heat sinks.

Chapter five covers the construction, results, recommendations and the reference materials used in the development of the project. Lastly, the report ended by the references of materials used in build up and appendix part that consists of various standard values of components and datasheet.
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CHAPTER 1

1.0 INTRODUCTION

An amplifier is an active electronic circuit that is used to raise the level of an alternating signal from a transducer to a value that can be used in some useful applications. For example, recorded music is stored in terms of magnetic information on magnetic tapes or as bit strings of ones and zeros in a compact disc. In the computer world, the information is also stored on mass storage systems, which at present level of technology are mainly magnetic. In radio or TV transmission, the received signals are rarely above $5 \mu V m^{-1}$. All these signals must therefore be amplified to useful levels, thus requiring the services of amplifiers. The output signals from a transducer e.g. a cassette head, a condenser microphone, an LP pick up can either be a voltage or a current. Depending on the amplifier design, the output from the amplifier could also either be a voltage or a current.

Amplifiers used in professional recording applications and commercial public address systems may incorporate pan controls, delay lines, and harmonic modification capabilities whereas some input devises such as tape decks, CD players and FM receiver microphone may be switched within the amplifier circuitry.

1.1.1 Amplifier properties

All amplifiers have three fundamental properties; gain, input impedance and output impedance. The quality of an amplifier is the ability to provide an accurate reproduction of its input signal over the full range of the signal frequencies. The input impedance of an amplifier must be high to ensure that the signal source is not loaded. The output impedance of an amplifier must be low enough to ensure that the output signal is not loaded by the impedance of the load.

Amplifier gain, $A$

The gain of an amplifier is the multiplication factor that relates the magnitude of the output signal to the input signal. In most cases, the gain should be linear otherwise the output signal will be distorted. There are three types of gains: voltage gain current gain...
and power gain. Gain is usually represented using the letter $A$, as shown in the table below.

Table 1.1 Amplifier gains

<table>
<thead>
<tr>
<th>TYPE OF GAIN</th>
<th>Symbol</th>
</tr>
</thead>
<tbody>
<tr>
<td>Voltage</td>
<td>$A_v$</td>
</tr>
<tr>
<td>Current</td>
<td>$A_i$</td>
</tr>
<tr>
<td>Power</td>
<td>$A_p$</td>
</tr>
</tbody>
</table>

**Amplifier Input Impedance ($Z_{\text{in}}$)**

When an amplifier is connected to a signal source, the source sees the amplifier as a load. The input impedance of the amplifier is the value of this load. If we make the assumption that the input impedance of the amplifier is purely resistive, the signal voltage at the amplifier is then found as

$$v_{\text{in}} = v \left( \frac{Z_{\text{in}}}{R_s + Z_{\text{in}}} \right)$$

Since both $R$ and $Z$ form a potential divider, the input voltage to the amplifier must be lower than the rated value of the source.

**Amplifier Output Impedance ($Z_{\text{out}}$)**

When a load is connected to an amplifier, it acts as a source to that load. Just like any other source, there is some measurable value of source impedance. In this case the output impedance of the amplifier. Assuming that the output impedance of the amplifier is purely resistive, the value of the load voltage can be found using the voltage divider equation

$$V_L = V_{\text{out}} \left( \frac{R_l}{Z_{\text{out}} + R_l} \right)$$

1.3 **ANALOG AUDIO SIGNALS**

Analog audio signals are supplied from two major sources. The majority of these sources are outputs from powered electronic devices such as radio tuners, cassette
These signals typically have a level of the order of 1 volt for maximum output and have a 'flat' frequency response. These are frequently referred to as line level sources.

The other major source of analog audio signals are the transducers for example in audio cassettes which are created using transducers to turn the electrical signal from a microphone pick up—which in turn goes through a transducer to convert the sound waves in electrical signal-in to magnetic fluctuations on the tape head. These magnetic fluctuations are then read and converted by another transducer (in this case a stereo system) to be turned back into electrical signal, which is then fed to the speakers. They act as yet another transducer to turn the electrical signals into audio waves.

In vinyl LP disks, the signal source consists of a voltage generated by a mechanical transducer either responding to displacement or the velocity. Since the available dynamic range of the vinyl LP disc was significantly lower than that of the music to be recorded, the technique of pre-emphasis during recording process was employed to enhance this dynamic range. As a result, the frequency spectrum of the signal source is not ‘flat’. The recorded amplitude varies as a function of frequency. One of the prime functions of a pre-amplifier with an input for analog LP disc replay is to provide both the gain to match the other line level sources and the equalization to convert the signal back to the original music spectrum for the ultimate performance. The analog LP amplifier may be designed as a stand alone unit which then feeds a preamplifier through a line input, this unit is often referred to as a phonon preamplifier.

### 1.2 AUDIO AMPLIFIERS

#### 1.2.1 The need for audio amplifiers

An audio amplifier is an electronic amplifier that amplifies low power audio signals (signals composed primarily of frequencies between 20Hz and 20 kHz, the human range of hearing) to a level suitable of driving loudspeakers. It’s usually the final stage in a typical audio playback chain. The preceding stages in such a chain are low power audio amplifiers which perform tasks like pre-amplification, equalization, tone control,
mixing/effects, or audio sources like record players, CD players and cassette players. Most audio amplifiers require these low level inputs to adhere to line levels.

Audio amplifiers are essential elements in the audio chain for sound systems and home theatres. A Preamplifier usually precedes another amplifier to prepare the electronic signal for further amplification or processing. In an audio system, the second amplifier is actually a power amplifier. Important applications of audio amplifiers include public address systems, theatrical and concert sound reinforcements, and domestic sound systems. Note that, the sound card in a personal computer contains several audio amplifiers (depending on the number of channels) as does every stereo or home theatre system.

The function of any given preamplifier is to amplify a low level signal to a line level when the source level is too low and has to be pre amplified for further processing, control or any other use. Line level is a term used to denote the strength of an audio signal (in decibels) used to transmit analog sound information between audio components such as CD and DVD players, TVs, audio amplifiers and sometimes MP3 players. Some common low level signal sources include; a microphone turntable, equalization, tone control and other transducers.

In audio systems, the term 'preamplifier' may be used to describe electronic equipment which merely switches between different line level sources and applies a volume control, so that no actual amplification may be involved. The pre amplifier provides a voltage gain (about 10 millivolts to 1 volt) but with no significant current gain. The power amplifier provides the higher current required to drive the speakers.

1.2.2 Classifications of Audio Amplifiers

Audio power amplifier classification is based on relationship between the output voltage swing and the input voltage swing, which is the amount of time the output devices operate during one complete cycle of the signal swing.

Class A operation is whereby both devices conduct continuously for the entire cycle of the signal swing, or the bias current flows in the output devices all the time.
Since both devices are always on, in reality class A amplifiers are not complementary designs but rather single ended designs with only one type polarity output devices. They are inefficient and run very hot due to the amplifier constantly operating at full power. However, they are linear, with the least amount of distortion.

Class B. In this class, there are two output devices, each of which conducts alternately (push pull) for exactly 180° of the input signal. These amplifiers are subject to cross over distortion if the transition from one element to the other is not perfect. Due to this, class B designs show high efficiency but poor linearity around the cross over region. This restricts class B designs to power consumption critical applications e.g. battery operated equipments such as two way radio and other communications audio.

Class AB; in this class, both devices are allowed to conduct at the same time but barely. The output bias is set so that current flows in a specific output device appreciably more than half cycle but less than the entire cycle. Thus the inherent non-linearity of class B design is eliminated, without the gross inefficiencies of the class A design. It’s this combination of good efficiency with excellent linearity that makes them popular in audio amplifier design.

Class C: This class is characterized by turning on of one device at a time for less than one half cycles. In essence, each output device is turned is pulsed on for some percentage of the half cycle, instead of operating continuously for the entire half cycle. This makes for an extremely efficient design capable of enormous output power. Class C amplifiers are restricted to broadcast industry for radio frequency (RF) transmission.

Class D: These use switching to achieve a very high power efficiency (more than 90% in modern designs). By allowing each output device to be either fully on or off, minimizes losses. The analog output is created by pulse width modulation i.e. the active device is switched on for shorter or longer intervals instead of modifying its resistor.

Other Classes: There are other amplifier classes, although they are marked by variations of the other classes. For example, Class G and Class F are marked by variation of the supply rails (in discrete steps or a continuous fashion, respectively) following the input
signal. These amplifiers are mainly used in for specialized applications such as very high power units.

1.2.3 Figures of merit

The quality of an amplifier can be characterized by a number of specifications enumerated as; Gain, Bandwidth, Efficiency, Noise, Output dynamic range, Slew rate, Overshoot, Stability factor, Linearity and Settling time.

The gain of an amplifier is the ratio of the output to the input power or amplitude, and is usually measured in decibels (dB). For example, an audio amplifier with a gain of 20dB will have a voltage gain of ten though a power gain of 100 would only occur in the unlikely event that the input and output impedances were identical.

The bandwidth (BW) of the amplifier is the range of frequencies for which the amplifier gives ‘satisfactory’ performance. A half power point is the frequency at which the power goes down by a half its peak value on the power vs frequency curve. Thus, bandwidth can be defined as the difference between the lower and upper half power points.

Efficiency is a measure of how much the input power is usefully applied to the amplifiers output. Class A amplifiers are usually very inefficient in the range of 10% to 25%. Class B amplifiers are very efficient though very impractical due to there high levels of distortion. Amplifiers of class C to F are usually known to be very efficient. Much more efficient amplifiers run much cooler and do not require cooling fans even in multi-kilo watt designs. The reason is that the low efficiency produces heat as a byproduct of energy lost during conversion of power. In more efficient amplifiers, there is less loss of energy so in turn less heat.

Noise is a measure of how much an unwanted signals is introduced in the amplification process. Noise is an undesirable but inevitable product of the electronic devices and components. Noise factor is the ratio of input signal to that of the output signal.

Output dynamic range; It’s the range, usually given in dB, between the smallest and the largest useful output levels. The lowest useful level is limited to noise, while the largest
is usually limited most often by distortion. If $S$ is the maximum allowed signal power, $N$ the noise power and $DR$ is the dynamic range, then the dynamic range, $DR$ is given by

$$DR = \frac{S + N}{N} \quad \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots \ldots ..
1.3.1 Equalization

Equalization is the process of changing the frequency envelope of a sound in audio processing. In passing through any channel, an audio signal will "spread" from its original qualities. The goal of equalization is thus to correct, or make equal the frequency response of the signal. It’s done to compensate for the poor acoustic characteristics of a room or the inaccurate frequency response of a speaker system. In other cases, this is done to change the characteristic sound of a particular signal source such as a voice or a musical instrument. One of the most direct uses of equalizations is at a live event, where microphones and speakers operate simultaneously. An equalizer is used to ensure that there is no frequency bands where there is around trip gain greater than one, as these are heard as audible feedback. Those frequencies are cut at the equalizer to prevent this problem. A typical equalizer for live sound reinforcement might have as many as 24 to 31 bands. A typical 31 band equalizer is also called a 1/3 octave equalizer because the center frequencies of the sliders are spaced one third of an octave apart.

1.3.2 Graphics and Parametric

Equalizers fall into two very large categories: graphics and parametric. Graphic equalizers further divide into two groups dominated by 15 band 2/3-octave equalizers and 30 band 1/3-octave equalizers. Functionally, parametric fall between 15 band and 30 band equalizers. The 15 band and 30 band equalizers offer great economy but very little flexibility or control.

Parametric give great control flexibility at the cost of limited bands. With three controls per band, very few bands are possible per instrument typically, they are only able to correct four, five or utmost eight frequency spots per equalizer. The 30 band equalizer is the preferred choice by sound professional, at a cost equal to, or slightly higher than parametric, but with the ease and convenience of being able to apply correction to 30 places.

Graphic equalizers get their name from the fact that the relative positions of the 15 or 30 sliders supposedly form a 'graphic' picture of the frequency response.
correction being applied. Parametric get their name from the fact that all the three 'parameters' of the filters are fully adjustable, i.e. Centre frequency, amplitude and bandwidth.

In graphic equalizer, the centre frequencies are fixed at standard locations and likewise, the bandwidths are normally set at one, 2/34-octave, or 1/3 – octave widths.

1.3.3 Types of equalizers

There are many types of equalizers; each has got a different pattern of attenuation or boost. A peaking equalizer 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, Hz is called a parametric equalizer. If there is no control for the bandwidth, then it’s known as a quasi-parametric or semi parametric equalizer.

Shelving type equalizers increase or attenuate the level of a wide range of frequencies by affixed amount. A low shelf will affect low frequencies up to a certain point and then above that point it will have a 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 equalizers is the graphic equalizers, which consists of a bank of sliders for boosting and cutting different bands (or frequency ranges) of sound. Normally, these bands are tight enough to give at least 3dB or 6dB maximum effect for neighboring bands, and cover the range from around 20Hz to 20 kHz.

Passive Equalizer is a variable equalizer that requires no power to operate. It consists of only passive components (inductors, capacitors and resistors). They are favored for their low noise performance (no active components to generate noise), high dynamic range (no active power supplies to limit voltage swing), extremely good reliability (passive components rarely break), and lack of RFI interference (no semiconductors to detect radio frequencies).

They are however disliked for their cost (inductors are expensive), size and weight, hum susceptibility (need careful shielding), and signal loss characteristic (passive equalizers
always reduce the signal). Also inductors saturate easily with large low frequency signals, causing distortion. They are primarily used for notching in permanent sound systems.

Active equalizers are variable equalizers that require power to operate. They are favored for their low cost, small size, light weight, loading indifference, good isolation (high input and low output impedances) gain availability (signal boosting is possible), and line driving ability.

They are disliked for there increased noise performance, limited dynamic range, reduced reliability due to use of feedback there is a possibility of instability, and RFI susceptibility.

Parametric equalizer allows the operator to adjust the boost and cut of different bands. It also allows for the adjustment of “Q” (the quality of the curve of the band passed the filter). One drawback of the parametric equalizer is they usually have fewer bands than a graphic equalizer though they are extremely useful in fine tuning a system with only a few flows in its frequency response.
CHAPTER 2

2.1 FILTER FUNDAMENTALS

Equalizer correction is accomplished by use of band pass filters, each designed to function over a different range of frequencies. The filter passes only a specific band of frequencies. A filter is simply a frequency selective electrical circuit, that is, it removes or attenuates signals at some particular frequency or frequencies while passing all the others. Traditionally, they were designed using RLC networks of varying complexity.

Filter order; ideally a filter should completely reject all signal energy in its stop band(s) and pass all signal energy within its pass band(s). This behavior is termed as the brick wall response which is the limiting response as we increase the filter order to infinity. For high and low pass filters, for each increase in the filter order, the asymptotic roll off rate of the filter increases by 20dB/decade.

There are two possible ways of implementing first order low pass filter; either using $RC$ or

![Low pass filter structures](image)

Figure 2.1 Low pass filter structures  (a) First-order low pass network.  
(b) A second order RC filter with buffering between sections. (c) Second order low pass LC filter
Second order filters can however be implemented in many different ways e.g. By cascading two first order low pass sections. However, a buffer has to be between the two sections to prevent loading of the first section an RL

2.2 THEORY OF ACTIVE FILTERS AND THEIR APPLICATIONS IN EQUALIZER DESIGN

Active filters are distinguished from passive filters by the use of one or more active components i.e. voltage amplifiers or buffer amplifiers. The frequency response curve of for each type of active filter is shown in figure (3.1). The low pass filter passes all frequencies from 0Hz up to a cut off frequency, f_c2. The high pass filter passes all the frequencies above a lower cut off frequency f_c1. The curve implies that a high pass filter does not have an upper cut off frequency. The op-amp in a high pass filter does have a unity gain frequency, so the circuit must have a value of f_c2. However, this value is normally well beyond the range of frequencies that is applied to the circuit. Therefore, it is normally of no consequence. The band pass filter passes all frequencies that fall between its values of f_c1 and f_c2. The band stop filter blocks all frequencies between its values of f_c1 and f_c2. In other words, it passes all frequencies below f_c1 and f_c2. Note that, band pass and notch filters have opposite frequency characteristics.

\[
\begin{align*}
\text{(a) Low pass filter} & \quad \text{(b) High pass filter}
\end{align*}
\]
The concept of, Q, centre frequency, and bandwidth are related primarily to the band pass and notch filters. When dealing with band pass and notch filters, we are generally concerned with the band widths of such amplifiers, along with their respective values of Q and $f_0$. However, when we are dealing with low pass and high pass filters, we are interested only with the value of $f_{c2}$ or $f_{c1}$ respectively.

**General terminology**

The following terms are commonly used to describe active filters. The first of these is the term pole. The term pole is used in active filters to refer to an RC circuit. Thus a one pole filter contains one RC circuit. The higher the number of poles in an active filter, the higher the gain roll off rate when the circuit is operated outside its pass band. The table 2.1.2 below illustrates the relationship among order; poles and gain roll off for Butterworth filters;

**Table 2.1.2 Number of poles versus the total gains roll off**

<table>
<thead>
<tr>
<th>Filter type</th>
<th>Number of poles</th>
<th>Total gain roll off</th>
</tr>
</thead>
<tbody>
<tr>
<td>First order</td>
<td>1</td>
<td>20 dB/decade</td>
</tr>
<tr>
<td>Second order</td>
<td>2</td>
<td>40 dB/decade</td>
</tr>
<tr>
<td>Third order</td>
<td>3</td>
<td>60 dB/decade</td>
</tr>
</tbody>
</table>
2.2.1 BUTTERWORTH, CHEBYSHEV AND BESSEL FILTERS

The Butterworth filter has a relatively constant gain across its pass-band. The term flat response is commonly used to describe this constant gain characteristic. Due to there flat responses characteristics, Butterworth filters are sometimes referred to as maximally flat or flat-flat filters.

Chebyshev filters have got a higher roll off rate (per pole) as compared to Butterworth filters. However, there are two inherent problems with chebyshev filters; one, the gain of a chebyshev filter is not constant across its pass band. Two, the chebyshev filter has got a high roll off rate only for frequencies just outside the pass band. As the operating frequency moves further outside the pass band, the chebyshev and Butterworth filters have equal roll off rates. The ripple width of the filter is the maximum variation in the filter gain, measured in dB. It should be noted that the width of a chebyshev filter can be reduced design.

\[ A_v (\text{dB}) \]

\[ 3 \text{ dB} \]

\[ f_{c2} \]

Flat responses

Figure 2.2 Butterworth filter flat response

This is however done at the expense of its high initial roll off rate. In other words, if a chebyshev filter is designed for a lower ripple width, its roll of characteristic come closer to those of a butter worth filter.

The chebyshev has a higher initial roll off rate than the butter worth filter. However, as the circuit operating frequency moves further outside the pass band,
the two circuits eventually reach the same roll off rate. As a result, the Butterworth is by far more commonly used of the two.

![Figure 2.2.2 the effect of increasing roll off rates](image)

Even so the Butterworth filter has its own drawback, the time delay (from input and output) produced by the Butterworth filter is not constant across its pass band. This means that two (or more) frequencies applied to the Butterworth filter do not experience the same phase shift from input to the output. The gain provided by the filter is constant, but the phase shift is not. This can produce severe signal distortion from the input to the output.

![Figure 2.2.3 Chebyshev filter](image)
The Bessel filter is designed to provide a constant phase across its pass-band. The constant phase shift of the Bessel filter results in greater fidelity (ability to reproduce a waveform accurately) than either the Butterworth or Chebyshev filters.

Of the three filter types introduced in this section, the Butterworth is the most commonly used, we will concentrate on this type of circuit in the upcoming design section of the project.

Figure 2.2.4 Chebyshev, Butterworth, and Bessel response

2.3 LOW PASS AND HIGH PASS FILTERS

2.3.1 Single pole low pass filter

The single pole low pass Butterworth filter is designed as either a high gain circuit or a voltage follower. A voltage follower has a voltage gain of 0dB, or unity. The circuit below is an example of a single pole low pass filter.

Non-inverting amplifier with an RC circuit added to the input. Since the reactance of the capacitor decreases as the frequency increases, the RC circuit limits the high frequency response.

The value of $f_c$ for this type of circuit is found as
\[ f_{c2} = \frac{1}{2\pi R C} \] (2.3)

Figure 2.3.1 Single pole low pass filter

2.3.2 Two pole low pass filter

It has a roll off rate of 20dB. The two commonly used two pole low pass filter configurations are as shown in figure 2.3. It can be seen that each has got two RC circuits, R1C1 and R2C2. As the operating frequency increases beyond fc2, each RC circuit reduces AC at the rate of 20dB/decade, giving a roll off rate of 40dB/decade. The cut off frequency for each circuit is given by

\[ f_{c2} = \frac{1}{2\pi R_1 R_2 C_1 C_2} \] (2.4)

There is however a restriction on the closed loop gain of a two pole low pass filter.
Figure 2.3 Two common two pole low pass filter configurations a

a) Unity gain filter  b) Variable gain filter

For the filter to have a Butterworth response curve $A_{cl}$ should not be greater than 1.586 (4dB). This means that you can not have a high gain two pole low pass filter with a flat response curve. The derivation of the value 1.586 involves calculus and is not covered here. In this case we simply accept the value as valid.

The variable gain filter can be designed for any value of $A_{cl}$ between 1 and an upper limit of 1.586. Note that this circuit is usually designed according to the following guidelines:

1. $R1 = R2$
2. $C2 = 2C1$
3. $R7 \geq 0.586 \times R$

High pass filters

The high pass filter differs from the low pass filter in two aspects; First, is the fact that the resistors and capacitors are swapped positions. Second, the multi-pole circuits are designed to fulfill the following conditions: $C5=C6$ and $R1=2R2$. Since the capacitors are in series with the amplifier input, they limit the low frequency operation of the circuit. Note that the value of $f_{c1}$ for each circuit is found using the same equation that we used to find $f_{c2}$ for the equivalent low pass filter. As the frequency decreases, the reactance of a given series capacitor increases. This causes a larger portion of the input signal to be dropped across the capacitor. When the operating frequency reaches the value $f_{c1}$, the series capacitors reduce the gain of the amplifier by 3dB. These circuits abide by all the rules established up to this point.
The only difference is that they pass frequencies above their cut off frequencies.

Figure 2.4. Typical high pass filter one, two and three pole respectively.

Filter gain requirements

Having been provided with the gain requirements for several low pass and high pass active filters, fulfilling these requirements provides a Butterworth response curve. That is each filter has relatively constant gain until its cut off frequency is reached. Then the gain rolls off at an approximate rate of 20dB/decade/pole. Thus almost every type of low pass or high pass active filter has gain requirements that must be fulfilled if the circuit is to have a Butterworth response curve. Any of the multiple filter can be constructed using the approximate number of two pole
and one pole cascaded stages. For example, a five pole circuit can be constructed by cascading two-two pole circuits and a one pole circuit.

2.4 Operational Amplifier Frequency Response and Compensation

Signals applied to operational amplifiers experience phase shifts as they pass from the input to output. These phase shifts are greatest at high frequencies and at some particular frequency, the total loop phase shift (from the inverting input to the output and back to the input via the feedback network) can add up to 360°. When this occurs, the amplifier circuit can easily go into a state of unwanted oscillation / be unstable. Two conditions have to be fulfilled for a circuit to oscillate: one, the loop gain must be equal to or greater than one, and two, the loop phase shift should equal 360°. The various measures taken to combat instability include the use of capacitors and resistors to reduce the total phase shift. The loop gain is the voltage gain around the loop from the inverting input terminal to the amplifier output, and back to the input via the feedback network. The loop phase shift is the total phase shift from the around the loop from the inverting input terminal to the amplifier output, and back to the input via the feedback network.

Most operational amplifiers have compensating components included in the circuitry to ensure stability. In some cases, compensating components have to be connected externally to stabilize the circuit.

Due to the presence of a feedback network, high frequency oscillations can occur in many operational amplifier circuits, and when this happens, the circuit is termed as unstable.

2.4.1 Stray capacitance effects

Stray capacitance, C, at the input terminals of an operational amplifier effectively introduces an additional phase lag network in the feedback loop, thus making the op-amp circuit unstable. Stray capacitance problems can be avoided by good circuit design techniques that keep the stray to a minimum. The effects of stray capacitance also depends on the resistor values used in the feedback network. High resistance values
make it easier for small stray capacitances to produce phase lag. With low resistances, small stray capacitances normally have little effect on the circuit stability.

### 2.4.2 Biasing operational amplifiers

Biasing bipolar op-amps

Operational amplifiers must be correctly biased if they are to function properly. As already discussed, the inputs of an operational amplifier are the base terminals of the transistors in a differential amplifier.

Base currents must flow into these terminals for the transistors to operate. Consequently, the input terminals must be directly connected to suitable dc bias voltage sources. The most appropriate dc bias voltage level for op-amp inputs is approximately halfway between the +/- supply voltages. One of the two input terminals is usually connected in some way to the op-amp output to facilitate negative feedback. Where a +/- supply is used, the input must be biased directly to the ground via signal source. See the fig. Below

![Op-amp circuit with one input](image)

**Figure 2.5 Op-amp circuit with one input**

grounded via resistor $R_1$, and the other input connected to the output via $R_2$

Base current $I_{B1}$ flows into the op-amp non-inverting input via the signal source, while $I_{B2}$ flows from the output into the inverting input, as illustrated.

Figure 2.5 shows a situation whereby one input is connected via $R_1$ to the ground, and the other is connected via $R_2$ to the op-amp output. Once again base
current for the input stage transistors flows into the both input terminals. \( R_1 \) and \( R_2 \) should have equal resistance values, so that voltage drops \( I_{B1}R_1 \) and \( I_{B2}R_2 \) are approximately equal. Any difference in these two voltage drops appears as op-amp dc input voltage which may be amplified to produce a dc offset at the output.

If very small resistance values are selected for \( R_1 \) and \( R_2 \) in the circuit, then a very small input resistance \((R_1)\) would be offered to a capacitor coupled signal source. On the other hand, if \( R_1 \) and \( R_2 \) will be very large, and the voltage drops \( I_{B1}R_1 \) and \( I_{B2}R_2 \) might be ridiculously large. An acceptable maximum voltage drop across these resistors must be very much smaller than the typical \( V_{BE} \) level for a forward biased base-emitter junction.

\( R_1 \) should be selected such that;

\[
V_{R1} \approx V_{BE}/10 \qquad \text{.......................................................... (2.4)}
\]

\[
R_{(\text{max})} = V_{R1}/I_{B(\text{max})} \qquad \text{.......................................................... (2.5)}
\]

\[
R_{R1 (\text{max})} = V_{BE}/10I_{B (\text{max})} \qquad \text{.......................................................... (2.6)}
\]

Where a voltage divider \((R_1 \) and \( R_2\)) are used to provide an input bias level from the supply voltages, the voltage divider current \( I_2 \) should be selected to be very much larger than the input bias current.

That is; \( I_{1(\text{min})} = 100I_B(\text{max}) \)

And \( R_3 = R_1 || R_2 \)

This is done so as to minimize any difference between \( I_{B1}(R_1||R_2) \) and \( I_{B1}R_3 \) that could behave as a dc input voltage.

**2.4.3 Thermal Stability of Bias Circuits**

\( V_{BE} \) and \( I_{CBO} \) Variations

Since many circuits are required to operate over a wide range of temperature, one aspect of bias circuit stability is thermal stability or how stable \( I_C \) and \( V_{CE} \) remain
when the circuit temperature changes. The base emitter voltage, $V_{BE}$ and the collector base reverse saturation current, $I_{CBO}$ are the two temperature sensitive quantities that largely determine the thermal stability of a transistor circuit.

For a silicon transistor, $V_{BE}$ changes by approximately $-1.8\text{mV/}^\circ\text{C}$, and $I_{CBO}$ approximately doubles for every $10^\circ\text{C}$ rise in temperature. An increase in $I_{CBO}$ causes $I_C$ to be larger, and $I_C$ increase raises the collector base junction temperature. This in turn results in a further increase in $I_{CBO}$. Thus, this effect is cumulative so that the end result may be a substantial collector current increase. This could produce a significant shift in the circuit Q point, or in the worst case $I_C$ might keep on raising until the transistor collector base junction overheats and burns out. This effect is known as thermal runaway. Measures taken to avoid thermal runaway are similar to those required for good bias stability against hfe spread.

2.5 VOLTAGE FOLLOWER CIRCUITS

2.5.1 Direct-coupled voltage follower

The very basic application of the IC op-amp is the direct-coupled voltage follower. The output terminal is connected directly to the inverting input terminal, the signal is then applied to the non-inverting input, and the load is directly-coupled to the output.

2.5.2 Capacitor coupled voltage follower

When a voltage follower is to have a capacitor coupled input and output terminals, the non-inverting input must be grounded via a resistor. The resistor is required for passing bias current to the non-inverting input terminal. It also offers an input resistance to the signal source, rather than a short circuit. However, the dc offset voltage at the op-amp output terminal can limit the amplitude of the output voltage. So a resistor ($R_2$) equal to $R_1$ should be included in series with the inverting input terminal to equalize the $I_B R_B$ voltage drops and thus minimize output offset voltage.
Fig. 2.6 a capacitor coupled voltage follower

Design of the capacitor-coupled circuit above involves calculation of R1, C1, and C2. The circuit input impedance is $R_1 || Z_i$, however since $Z_i$ is always greater than $R_1$, the circuit input impedance is simply taken as $Z_{in} = R_1$.

Normally the load resistance $R_L$ is much smaller than $R_1$, and consequently, the smallest capacitor values are calculated when $C_2$ is selected to set $f_1$. Therefore,

$$X_{C2} = (Z_0 + R_L) \text{ at } f_1$$

Capacitor $C_1$ should then be calculated from,

$$X_{C1} = (Z_{in} + r_s)/10 \text{ at } f_1$$

so that it has got no significant effect on the circuit lower cut off frequency.

$$X_{C1} = (R_1 + r_s)/10 \text{ at } f_1$$

2.6 Non-inverting Amplifiers

The non-inverting amplifier circuit behaves similarly to a voltage follower circuit with one major difference. Instead of all of the output voltage being fed directly back to the inverting input terminal, (as in the voltage follower circuit), only a portion of $V_o$ is fed back. The output voltage is divided by resistors $R_3$ and $R_2$, and the voltage across $R_3$ is applied to the inverting input terminal. As in the case of a voltage follower, the output voltage changes as necessary to keep the inverting input terminal voltage equal to that at the non-inverting input. Thus the voltage $V_{R3}$ always equals $V_i$, and the output voltage is then determined by the resistances of $R_2$ and $R_3$. 

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Since the signal voltage is applied to the op amp non-inverting input terminal, the output always has the same polarity as the input. A positive going input produces a positive going output, and vice versa. Thus, the input is not inverted (at the output), and the circuit is thus identified as a non-inverting amplifier.

![Capacitor coupled non inverting amplifier](image)

Figure 2.7 Capacitor coupled non inverting amplifier

The voltage divider current $I_2$ through $R_2$, is always selected to be very much larger than the operational amplifier input bias current, and

$$V_{R3} = V_i = I_2 R_3 \quad \ldots \quad (2.7)$$

$$I_2 = \frac{V_o}{R_2} + R_3 \quad \ldots \quad (2.8)$$

The circuit voltage gain is given by

$$A_{CL} = \frac{V_o}{V_i} = I_2 \frac{R_2}{R_3} + \frac{R_2 R_3}{I_2} \quad \ldots \quad (2.9)$$

$R_1$ is selected to be approximately equal to resistance seen looking out of the inverting input terminal.

Thus, $R_1 = R_3 \parallel R_2 \quad \ldots \quad (2.10)$

Design of a non-inverting amplifier mostly involves determining suitable voltage divider resistors ($R_2$ and $R_3$). Hence design commences with selection of the voltage divider current to be much larger than the op amp input bias current.
2.5.1 Capacitor Coupled Non inverting Amplifier.

When a non-inverting amplifier is to have a signal capacitor coupled to its input, the op amp non-inverting input terminal must be grounded via a resistor to produce a path for the input bias current. This is illustrated in the figure below where $R_1$ allows for the passage of the $I_{B1}$. The input resistance is essentially equal to $R_1$ for the capacitor coupled non-inverting amplifier.

![Figure 2.8 Capacitor coupled non inverting amplifier.](image)

Resistors $R_1$, $R_2$ and $R_3$ in the capacitor coupled circuit are determined exactly as for a direct coupled non-inverting amplifier. The capacitor values are calculated in the same way as for a capacitor coupled voltage follower.

2.7 Inverting Amplifiers

Direct Coupled Inverting Amplifiers

The circuit below is an example of an inverting amplifier because, with $V_i$ applied via $R_1$ to the inverting input terminal, the output goes negative when the input goes positive and vice versa. The non-inverting input terminal is grounded via resistor $R_3$. With the non-inverting terminal grounded, the voltage at the op amp inverting input terminal remains close to the ground. Any increase or decrease in the (very small) difference voltage between the two input terminals is amplified by the op amp open loop gain and fed back via $R_2$ and $R_1$ to correct the change. Because the inverting input terminal is not grounded but remains close to
the ground, the inverting input terminal in this applications termed as a *virtual
ground or virtual earth*.

The circuit input current can be calculated as,  \[ I_1 = \frac{V_{R1}}{R_1} \]

The input voltage is applied to one end of \( R_1 \), and the other end of \( R_1 \) is at
ground level

Consequently,  \( V_{R1} = V_i \) and  \( I_1 = V_i R_1 \)

\( I_1 \) is always selected to be very much larger than the op amp input bias current
\( (I_B) \). Consequently, all of \( I_1 \) flows through resistor \( R_2 \). The voltage drop across \( R_2 \) is
given by \( V_{R2} = I_1 R_2 \)

The left side of \( R_2 \) is connected to the op amp inverting input terminal, which
means that the right side of \( R_2 \) (the output terminal) is \( V_{R2} \) below ground.

\[ V_O = -I_1 R_2 \] \……………………………………………………………………………………..(2.10)

Hence, the voltage is given by

\[ A_{CL} = \frac{V_O}{V_i} = -\frac{R_2}{R_1} \] \……………………………………………………………………(2.11)

If the input of the inverting amplifier is grounded, the circuit is seen to be
exactly the same as non-inverting amplifier with \( R_3 \) as the input resistor and zero
input voltage. Thus negative feedback occurs to maintain the op-amp input
terminal at the same voltage level as the non-inverting input terminal.

The input impedance of an inverting amplifier is easily determined by recalling that
the right side of \( R_1 \) is always at ground level, and that the signal is applied to the
left side of \( R_1 \). Therefore \( Z_i = R_1 \).

In the design of an inverting amplifier, the voltage divider current \( I_1 \) is selected
very much larger than the op-amp input bias current.
2.8 SIGNAL DISORTION IN AMPLIFIERS

The output signal of an amplifier may not be the exact replica of the input. The phase shift that the signal suffers on passing through the amplifier is a nonlinear function of frequency so that the phase relationship in the frequency components in the input signal is not preserved. Further, the gain of the amplifier is a function of frequency and the different frequency components undergo different amplification. Thus, the spectral shape of the signal is not preserved. This effect is referred to as signal distortion.

There are a number of mechanisms through which a given amplifier introduces distortion and often, these mechanisms will be active simultaneously. How much the amplifier preserves the signal integrity is known as the amplifier fidelity, and an amplifier with a high fidelity is described as a HI-FI (hi-fi) amplifier in the audio industry. The various distortion mechanisms include;

Non-linear or amplitude distortion: It arises due to the nonlinearity of the amplifier transfer characteristic, whereby new frequencies are generated in the amplifier that were not present in the input signal which may result to amplitude distortion.

Frequency distortion. This arises from the frequency dependence of the amplifier transfer function, so that different frequency components in the signal are amplified with different gains. The shape of the signal spectrum is thereby changed by the amplifier.

Phase distortion. Since the phase shift through the amplifier is a non-linear function of frequency, the phase relationship of the different components suffer different time delays.

In the ideal case, the phase shift should be proportional to frequency so that each component suffers the same time delay.
CHAPTER 3

3.1 BAND PASS FILTERS

A band pass filter is an electrical circuit that passes frequencies within a certain range and rejects (attenuates) frequencies outside that range. An example of an analogue electronic band pass filter is an RLC circuit (a resistor-inductor-capacitor circuit). A band pass filter is characterized by three major parameters namely; center frequency, amplitude response (gain), and bandwidth. Center frequency is the frequency at which the amplitude is maximum. Gain is the maximum amplitude response occurring at the center frequency, and bandwidth (or pass band) is the frequency range between the -3dB points located at either side of the center frequency. Band width is usually expressed in octaves. One octave is doubling of frequency whereas a one third octave is a 26% increase in frequency.

The Quality Factor of a filter, 'Q' is given by the centre frequency divided by the bandwidth. High values of Q translates to narrow bandwidths and vice versa. When designing an audio filter, normally the required BW in octaves is known and the associated Q needs to be calculated; once the filter has being designed, then Q is easily calculated by measuring the -3dBN frequency points, taking the difference and dividing it into the center frequency; and lastly the BW in octaves is then calculated.

![Amplitude (dB)](image)

Figure 3.1 Band pass filter parameters
Band pass filters are designed to pass all frequencies within their band widths, while notch (band stop filters) are designed to block all the frequencies within their bandwidths. In this section, we shall look at several band pass and notch filters. It is common for a band pass filter to be constructed by cascading a high pass filter and a low pass filter. Such a circuit is shown in figure 3.2. The first stage of the amplifier passes all the frequencies that are below its value of \( f_{c2} \). All the frequencies passed by the first stage are coupled to the next stage, which passes all frequencies above its value of \( f_{c2} \). The circuit action is as shown in Figure 3.2. Note that the only frequencies through the amplifier are those that fall between the pass band of both amplifiers. The values of \( f_{c2} \) are found as shown in Section 3.2. Once the values of frequency analysis of a circuit like the one in Figure 17.21a is relatively simple \( f_{c1} \) are known, the circuit values of bandwidth, geometric center frequency and \( Q \) are found as follows

\[
20) \quad BW = f_{c1} - f_{c2} \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad (3.1)
\]

\[
21) \quad Q = \frac{O_{f}}{BW} \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad (3.2)
\]

\[
22) \quad Q = \frac{f_{o}}{BW} \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad \quad (3.3)
\]

The two stage band pass filter is the easiest of the band pass filters to analyze. However, it has the disadvantage of requiring two op-amps and a relatively large number of resistors. Thus the physical construction of the multiple feedback band pass filter is much simpler than that of the two stage filter. However, the frequency analysis of the multiple feedback band pass filter is a bit more difficult.

### 3.1.2 Multiple feedback band pass filters

The multiple feedback filter derives its name from the fact that it has two feedback networks, one capacitive and another one resistive. Note the presence of the input series capacitor (C1) and the input shunt capacitor. The series capacitor affects the low frequency response of the filter, while the shunt capacitor (C2) affects the high frequency response.
To simplify our discussion, we shall assume that:

1. \( C_2 < C_1 \)

2. A given capacitor acts as an open circuit until a short circuit frequency is reached. At that point, the capacitor is represented as a short circuit.

(Granted, the operation of the capacitors is more complicated than this, but this is just a start for getting the overall picture of how the circuit works). When the input frequency is below the short circuit frequency for \( C_1 \), both capacitors act as open circuits. Since \( C_1 > C_2 \), we know that the short circuit frequency for \( C_1 \) is lower than that of \( C_2 \).

![Figure 3.2 Two stage Band pass filters](image)

Therefore, as long as \( C_1 \) is open, \( C_2 \) is also open. Having both capacitors acting as opens gives us the equivalent circuit with the input signal completely isolated from the op-amp. Therefore, the circuit has got no input.

Assuming that the lower cut off frequency \( f_{c1} \) equals the short circuit frequency for \( C_1 \), then, when \( f_{c1} \) is reached, \( C_1 \) becomes a short circuit, while \( C_2 \) remains open. This gives the equivalent circuit with \( V_{in} \) which has got no problem making it to input of the op-amp, and the filter acts as an inverting amplifier. Thus at frequencies above \( f_{c1} \), the filter has an output. Assume now that \( f_{c2} \) equals the short circuit frequency of \( C_2 \). When \( f_{c2} \) is reached, we have the equivalent circuit now with both capacitors acting as short circuits. The input signal has no
problem making it the op-amp, but now the output is shorted back to the input. Since the gain of an inverting amplifier is found as;

\[ A_{CL} = \frac{R_3}{R_1} \]  

(3.4)

Figure 3.3 A multiple feedback band pass filter

With \( R_F \) effectively shorted, the closed loop voltage gain of the circuit is zero at frequencies above \( f_{c2} \). If we put these three equivalent circuits together, we have a circuit with zero voltage gain when operated at frequencies below \( f_{c1} \), relatively high voltage gain when operated at frequencies between \( f_{c1} \) and \( f_{c2} \), and zero voltage gain when operated at frequencies above \( f_{c2} \). This by definition is a band pass filter. The cut off frequency for a given low pass or high pass active filter is given by

\[ f_c = \frac{1}{2\pi V_1 R_2 C_1 C_2} \]  

(3.5)

This equation is modified to provide us with the following equation for the geometric center frequency of a multiple feedback band pass filter.

\[ f_c = \frac{1}{2\pi V_1 R_2 C_1 C_2} \]  

(3.5)

Once the value of the center frequency, of a multiple feedback filter is known, we can calculate the value of \( C \) for the circuit using the following equation;

\[ C = \sqrt{C_1 C_2} \]  

(3.7)
Once the values of the center frequency, Q and band width are known, the values of $f_{c1}$ and $f_{c2}$ for the multiple feedback filter can be determined. The equations used to determine the cut off frequencies of the filter will vary with the Q of the circuit. That is, we shall use one set of equations to find $f_{c1}$ and $f_{c2}$ when $Q>2$ and another set to find $f_{c1}$ and $f_{c2}$ when $Q<2$.

Since $f_{ave}$ is halfway between the cut off frequencies, we can use the following equations to approximate the values of $f_{c1}$ and $f_{c2}$ when $Q>2$

$$f_{c1} \approx f_0 - \frac{BW}{2} \quad (3.8)$$

And

$$f_{c2} \approx f_0 - \frac{BW}{2} \quad (3.9)$$

(When $Q>2$)

The following equation is used to find the closed loop voltage gain of a multiple feedback filter

$$A_{CL} = \frac{R_f}{2R_i} \quad (3.10)$$

Where $R_i$ is the circuit series input resistor

### 3.2 Gyrator Design of Bandpass Filters

A gyrator is an electrical circuit that converts an impedance into its inverse. This lets one to replace an inductor with a capacitor, a couple of op-amps and some resistors. It is primarily used in active filter design and miniaturization. The primary use of a gyrator is to simulate an inductive element in a small electronic circuit. Before the invention of the transistor, coils of wire with large inductance were be used in electronic filters.

A real inductor can be replaced by a much smaller assembly containing a capacitor, operational amplifiers or transistors, and resistors. This is especially useful
in integrated circuit technology. Additionally, 'real capacitors' are often much closer to ideal capacitors than real 'ideal inductors'. Because of this, a synthetic inductor realized with a gyrator may improve the quality of filter networks that would otherwise be built using inductors. Also, the Q factor of a synthesized inductor can be selected with ease. Since gyrators use active components, they only function as a gyrator within the power supply range of the active element. Hence gyrators are usually not very useful for situations requiring simulation of the fly back property of inductors, where a large voltage spike is caused when current is interrupted.

3.2.1 Operation of the gyrator circuit

The circuit works by inverting the effect of the capacitor. The desired effect is an impedance of the form of an ideal inductor $L$ with a series resistance $R_L$:

$$Z = R_L + j\omega L \quad \text{.................................................................(3.11)}$$

From the Fig 3, the input impedance of the op-amp circuit is:

$$Z_{in} = \frac{(R_L = j\omega RC)||[(R + 1/j\omega C)]}{...} \quad \text{................................................. (3.12)}$$

With $R_LRC = L$, it can be seen that the impedance of the simulated inductor is the desired impedance in parallel with the impedance of $C$ and $R$. If $R$ is chosen to be adequately large than $R_L$, then

$$Z_{in} = R_L + j\omega R_L RC \quad \text{................................................................. (3.13)}$$

This is the same as resistance $R_L$ being in series with an inductor, $L = CR R_2$

![Figure 3.4; Gyrator simulating Inductance.](image)
Note that the two Zin have impedance similar values

3.2.2 Applications of gyrators

The primary application of a gyrator is to reduce the size and cost of a system by removing the need for bulky, heavy and expensive inductors. For example, since RLC band pass filter characteristics can be realized using capacitors, resistors and op-amps without using inductors, thus Hi-Fi graphic equalizers can be realized using without using inductors because of the 'invention' of the gyrator. They are also used in parametric equalizers, band stop and band pass filters and in FM pilot filters.

3.3 POWER AMPLIFIER WITH AN OP-AMP DRIVER

3.3.1 Power Amplification

A power amplifier, or a large signal amplifier, develops relatively large output voltages across low impedance loads. Audio amplifiers are large signal amplifiers that supply ac output power to speakers. In general, a power amplifier is designated as the last amplifier in a transmission chain (the output stage) and is the amplifier stage that requires most attention to power efficiency.

The amplifier load may be transformer coupled, capacitor coupled or direct coupled. Direct coupling usually gives the best performance, but plus and minus supply voltages are required. The output stage of the amplifier may use power BJT's or power MOSFET's. IC operational amplifiers may also be used in power amplifiers.

Power MOSFET's have got several advantages over power BJT's for large signal applications. One of the most important is that MOSFET transfer characteristics (\(I_D/V_{GS}\)) are more linear than \(I_C/V_{BE}\) characteristics for BJT's. This helps to minimize distortion in the output waveform. Thermal runaway does not occur with power MOSFET's, so the emitter resistors in the BJT output stage are not needed in a MOSFET amplifier, the wasted power dissipation in the emitter resistors is eliminated.
Power MOSFETs can be operated in parallel to reduce the total channel resistance and increase the output current level. Unlike BJTs operated in parallel, there is no need for resistors to equalize current distribution between parallel connected MOSFETs.

**Circuit Operation**

The amplifier shown in figure 3.5 uses an operational amplifier (A1) for the input stage. Resistors R4 and R5 together with the two diodes provide bias for the complementary emitter follower BJT output stage. There is a 100% dc negative feedback via R3 to keep the dc output at the same level as the op-amp non-inverting input, which is grounded via R1. Overall ac negative feedback via R2 and R3 controls the amplifier ac voltage gain. No amplification is produced by the intermediate (output biasing) stage. Instead, resistors R4 and R5 provide active pull up for transistors Q1 and Q2.

![Circuit diagram](image)

*Figure 3.5 A power amplifier circuit with an op amp driver*
When the op-amp output at the junction of D1 and D2 is increased in a positive direction, A1 supplies current through D2 and R5. So, the voltage drop across R4 is reduced, allowing it to pull the Q1 base up to the required level while supplying increased bias current to Q1. This is illustrated by the voltage levels shown in Fig.3.6.

It should be noted that Q2 is biased off when the output voltage is at its peak.

*Fig.3.6* Analysis of the power amplifier with an op amp driver shown in figure 3.5

*NB.* To simplify our discussion, we have ignored the input stage of the operational amplifier.
Figure 3.7a; analysis of push pull amplifier with bootstrapping capacitor (a)

A1 pulls current through R4 and D1, leaving R5 to the base of Q2 down to the required voltage level while supplying the increased base current. At this time, Q1 is biased off as indicated by the voltage levels. The diode voltage drops do bias Q1 and Q2 at least in to a low current on state. Although this may not completely eliminate cross over distortion, the negative feedback reduces this distortion by a factor \((1 + A_VB)\), where \(A_V\) is the circuit open loop gain and \(B\) is the feedback factor. Thus, a high open loop gain of an op-amp will severely attenuate the cross over distortion that would have been present without the negative feedback.
3.3.3 The need for bootstrapping capacitors

The resistance of (equal resistors) R7 and R6 as shown in the fig 3.16 above is limited by the need to supply base current to the output transistors. Also, there is a need for a minimum voltage drop across R7 and R6 to produce the base current. This minimum resistor voltage requirement keeps the amplifier maximum peak output voltage well below the supply voltage level, hence limiting the amplifier efficiency.

Bootstrapping capacitors (C3 and C4) can therefore be introduced in the circuit to improve this situation. The resistors R4 and R5 are divided by in two equal value resistors (R8, R9 to and R9, R10) as shown in figure 3.17. These capacitors couple the output the output voltage back to the junctions of these components. The bootstrap capacitors are usually equal and designed using the equation

$$X_{C1} = 0.1R_1 \text{ at } f_1$$

(3.14)

Where $f_1$ is the designed lower cut off frequency.

![Fig.3.7b; analysis of push pull amplifier with bootstrapping capacitor (b)](image-url)
The operation of the circuit is as follows:

Under quiescent conditions, the output is at zero volts, and the positive supply is divided by R8 and R9. The base of the upper transistor, Q1 will be at about +0.7V, just sufficient to bias the transistor. As the output swings positive or negative, the voltage swing is coupled via capacitor C3, so that the voltage across R9 remains constant. The current through resistor R9 is therefore constant, since the resistor maintains a constant voltage across it.

However, the above applies only to AC voltages, as the capacitor will charge if there is a DC variation.

3.4 NEGATIVE FEEDBACK

Negative feedback is produced by feeding a portion of the amplifier output back to the input where it behaves as an additional signal. The feedback quantity is applied in opposition to the signal, so that the effective input is reduced. This results in stabilized amplifier gain, extended bandwidth, reduced distortion, and modified input and output impedances.

Negative feedback can be implemented using two schemes, namely

1) Series Voltage Negative Feedback

In this scheme, the feedback voltage is applied in series with the signal voltage. The instantaneous polarity of the feedback voltage is normally opposite to the signal voltage polarity, (they are in series opposition). So, the feedback voltage is negative with respect to the signal voltage. It increases the input impedance of an amplifier by a factor of \((1 + AVB)\) where \(B\) is the feedback factor and \(AV\) is the open loop gain. It also reduces the output impedance of an amplifier by a factor of \((1 +AVB)\).
2) Parallel Current Negative Feedback

In parallel current feedback, a portion of the output current is fed back in parallel with the signal source. It stabilizes the current gain. It reduces the circuit input impedance by a factor of \((1 + A \lambda B)\)

However, the most important advantage of negative feedback is that it produces amplifiers with stable, predictable, voltage gains.

Additional effects of negative feedback include

A) Attenuation constant. It occurs when different frequencies are amplified by different amounts. It’s the result of the amplifier open loop gain being frequency dependent. When negative feedback is used, the closed loop gain is a constant quantity largely independent of signal frequency within the circuit band. It’s reduced by a factor of \((1 + A \lambda B)\)

B) Phase shift. Audio (and other) signals at an amplifier input are usually made up of a combination of component waveforms with different amplitudes and different frequencies. So, any variation in the amplifier introduced phase shift will create distortion in the output waveform that is not present in the input. To investigate the effect of negative feedback on amplifier phase shift, assume that the open loop gain has a phase shift angle of \(\phi_o\) and that the closed loop phase shift is \(\phi_{cl}\). Then, though not shown here, negative feedback reduces amplifier phase shift by an angle of

\[
\tan^{-1} \frac{A \lambda B}{1 + A \lambda B} \phi_o
\]

C) Noise. Circuit noise generated within the feedback loop of an amplifier is reduced by a factor of \((1 + A \lambda B)\). However, it should be noted that only the noise generated within the feedback loop is reduced by feedback. Noise produced by bias resistors outside the feedback loop will not be affected by the feedback.
D) Circuit stability. For a circuit to oscillate, the loop phase shift must be 360° when the loop gain is 1. The loop phase shift of an amplifier can approach 360° as the gain falls off at the high end of the bandwidth. In this case, the amplifier may oscillate (be unstable).

**Volume Controls**

The volume control of an amplifier can be a manually operated potentiometer. The volume control is a resistive voltage divider. The resistance taper of a volume control must be logarithmic because the human ear has a logarithmic response to sound waves.

An attenuator is a volume control that is switched in equal decibel steps. It’s not continuously variable. The volume control or attenuator should not affect the operating bias of the amplifier.

### 3.5 TRANSISTOR OPERATION

A bipolar junction transistor (BJT) has three layers of semiconductor material. These are arranged in npn (n-type p-type n-type) sequence, or in pnp sequence, and each of the three layers has a terminal. A small current at the central region terminal controls the much larger total current through the device.

#### 3.5.1 The npn transistor Operation

The centre layer of a transistor is very much narrower than the two outer layers. The outer layers are much more heavily doped than the centre layer, causing the depletion layers to penetrate deep into the base. Due to this penetration, the distance between the two depletion regions is very short (within the base).

For normal operation, the base emitter, BE junction is forward biased and the collector base, CB junction is reverse biased. The forward bias at the BE junction reduces the barrier voltage and causes electrons to flow from the n-type emitter.
to the p-type base. Holes also flow from the p-type base to the n-type type emitter, but since the base is much more lightly doped than the collector, almost all of the current flow across the BE junction consists of electrons of electrons entering the base from the emitter. Thus electrons are the majority charge carriers in the npn devise.

Because the BE junction is forward biased, it has the characteristics of a forward biased diode. Substantial current will not flow until the forward bias is about 0.7V for a silicon devise and 0.3V for a germanium. Reducing the level of the BE emitter bias voltage reduces the pn junction forward bias, and thus reduces the current that flows through from the emitter through the base to the collector. Increasing the BE bias voltage increases this current. Decreasing the BE bias voltage to zero, or reversing it turns the current off completely. Thus, variation of the small forward bias voltage on the BE junction controls the emitter and collector currents, and the BE controlling voltage source has to supply only the small base current.

3.5. The pnp transistor operation

In an unbiased pnp transistor, the barrier voltages are positive on the on the base and negative on the emitter and collector. Just as in the case of an npn transistor, the emitter and the collector are heavily doped, so that the BE and the CB depletion regions are penetrate deep into the lightly doped base.

A pnp transistor behaves exactly the same way as the npn devise, with an exception that the majority charge carriers are holes. The BE junction is forward biased by an external voltage source, and the CB junction is reverse biased. Holes are emitted from the p-type emitter across the forward biased BE junction into the base. In the lightly doped n-type base, the holes find few electrons to absorb. Some of the holes flow out through the via the base terminal, but most are drawn across to the collector by the positive negative electric field at the reverse biased CB junction. Variation of the forward bias voltage at the BE junction controls the small base current and the much larger collector and emitter currents.
NB: Although one type of charge carrier is in the majority in an pnp or npn transistor, two types of charge carriers (holes and electrons) are involved in the current flow. Consequently, these devices are termed as bipolar junction transistors (BJT). This is what distinguishes them from Field effect transistors, FET which are termed as unipolar devices because only one type of charge carrier is involved.

3.5.3 Transistor Voltages and Currents

Voltage Source Connections

For an npn transistor, the base is biased positive with respect to the emitter. The collector is then biased to a higher positive voltage than the base. The base bias voltage \( V_B \) is usually connected via a resistor \( R_B \) and the collector supply \( V_{CC} \) is connected via a resistor \( R_C \). The negative terminals of the two voltage sources are connected at the transistor emitter terminal. \( V_{CC} \) is always much larger than \( V_B \), and this ensures that the CB junction remains reverse biased; positive on the collector (n-side); and negative on the base (p-side).

Typical collector voltages might be anything from 3V to 20V for most types of transistors though in many the collector voltage may be greater than 20V.

For a pnp device, the base is biased negative with respect to the emitter. The voltage sources are connected via resistors, and the source positive terminals connected at the emitter. With the \( V_{CC} \) larger than \( V_B \), then the (p-type) collector is more negative than the (n-type) base, hence keeping the CB junction reverse biased.

NB. All transistors (npn and pnp) are normally operated with the CB junction reverse biased and the BE junction forward biased.

Transistor Currents

The various currents that flow in the transistor are: emitter current \( I_E \), base current \( I_B \) and the collector current \( I_C \). For an pnp device, \( I_E \) can be thought of as a flow of
holes from the emitter to the base. \( I_E \) and \( I_B \) both flow out of the transistor while \( I_E \) flows into the transistor.

Therefore \( I_E = I_C + I_B \) \ldots (3.15)

Typically, 96% to 98.5% of \( I_E \) flows across the collector base junction to become the collector current.

\[ I_C = \alpha_{dc} I_E \]

where \( \alpha_{dc} \) is the emitter to collector current gain. Numerically, \( \alpha_{dc} \) is typically 0.96 to 0.995, so collector current is almost equal to the emitter current.

Since the CB junction is reverse biased, a very small reverse saturation current \( I_{CBO} \) flows across the junction, and its so small that it can be neglected.

Substituting \( I_E \) from eqn 1, we have

\[ I_C = \alpha_{dc}(I_C + I_B) \] \ldots (3.16)

Which gives \( I_C = \alpha_{dc}I_B/(1-\alpha_{dc}) \) \ldots (3.17)

The above equation can be rewritten as \( I_C = \beta_{dc} I_B \) \ldots (3.18)

Whereby \( \beta_{dc} = \alpha_{dc}/(1-\alpha_{dc}) \) \ldots (3.18)

\( \beta_{dc} \) is the base to collector current gain. Typically, it ranges from 25 to 300. Instead of \( \beta_{dc} \), it’s also represented as \( h_{FE} \). This originates from the \( h \) parameter circuit and its the symbol used on many transistor data sheets.

Typical collector and emitter currents for low power transistors range from 1 mA to 25 mA, and base currents are usually less than 100 \( \mu \)A. High power amplifiers handle currents of several amperes.

3 Transistor Testing

In Circuit Testing

This is a quick test that can be carried out to test whether a transistor is operational. It can be performed while the device is still connected to the circuit. A
voltmeter is used to measure the collector voltage ($V_C$), its then noted. Then the emitter and the base terminals are temporary short-circuited. This should turn the transistor off, and the $V_C$ should return to its previous level. If the change in the $V_C$ level does not occur, then the transistor is not operational. This may be due to fault in the transistor or some other problem in the circuit.

CHAPTER 4.0

4.1 EQUALIZER DESIGN

4.2.1 Introduction

Graphic equalizers are extensively used in the music industry for recording performance and high end audio applications. A high performance graphic equalizer which covers the audio range in about fifteen bands or more is a highly desirable piece of equipment though it is quite complex and expensive to build.

However, a relatively simple unit consisting of six bands is cheap to build, but still gives far more control over sound than basic bass and treble tone controls. The unit covered here is a six band graphic equalizer which has approximate centre frequency of 30Hz, 100Hz, 300Hz, 1KHz, 3KHz and 10KHz. This corresponds to bass, lower middle frequency, middle frequency, upper middle frequency, lower and upper treble ranges. The maximum cut and boost available from each control is of range +/-10dB. The dB points on each band are at half and twice the centre frequencies. With all controls at maximum the unit has a total gain of 18dB. The unit will accept an input of up to about 1Vrms (3V peak to peak) before distortion occurs with all controls at maximum.

4.2.2 Design Procedure

The complete circuit diagram is shown in figure 4.2. The input is buffered by the first section of IC1 (AR7), which has a unity gain and consistent output impedance.
The value of $C_1$ is chosen according to the strength of the input signal which could have been higher for large signals, for this case the appropriate value is 470nF. Resistors $R_1$ and $R_2$ provide biasing to IC1 (AR7) and set input impedance to 23.5. The resistors $R_3$ and $R_4$ sets the gain at unity and $R_5$ provides the consistent output impedance.

To discuss the second stage, we assume that all the six frequency selective sections have disappeared, as well as five of the control pots. The wiper of the remaining pot is connected to ground via an imaginary resistor in place of the tuned band pass filters, lets say 1K0 resistor as shown in the figure.

If the pot is in the upper position or fully clockwise, the 1K0 resistor appears between the inverting input of the op-amp and ground, giving the stage a gain of ten. If the pot track resistance is 10K, equivalent of six 50K pots in parallel, then the signal at the non-inverting input is halved, giving a total gain of five.

With the pot in the lower position or anticlockwise, the input to the non-inverting input is reduced to a tenth, and the gain of the op-amp circuit is two, giving a total of five. With the pot in the centre, the gain of the stage is unity, since the attenuation of the input signal is cancelled by the gain of the op-amp. If our imaginary 1k0 resistor is replaced with a tuned circuit, the effects described above will only occur around its centre frequency. In this circuit, we have six tuned circuits giving the six bands.

Traditionally, the tuned circuits would consist of a capacitor and an inductor in series. Due to the lack of availability of suitable inductors, in modern designs, gyrator circuits are used to simulate the inductors. This uses an op-amp to reverse the phase relationship of a capacitor to make it appear like an inductor as shown in the figure. This is the simulated inductance of the bass frequency stage. Having the centre frequency as 100Hz, we can get an arbitrary value of capacitor using appendix rules, that is 360nF.
The value of the simulated inductance was then calculated using equation 4.3 i.e. \( \hat{L} = 7.036 \). But from equation (B1), the simulated inductance is given as \( \hat{L} = CRR \). From the equation we find that we have three unknowns, so the value of \( C2 \) (real capacitor) was taken arbitrarily as 360nF to equal that of \( C3 \), this was done to simplify the design. The remaining values of two resistors were taken such that they multiply to the value of \( R8*R7 = \hat{L}/C3 \) derived from equation (3.17) but ensuring the value of \( R8 \) was much higher than \( R7 \). This was because \( R6 \) controls the reactance of the simulated inductor, and therefore the "Q" of the tuned circuit as seen from equation 4.1 where the value of \( Rf = R6 \).

Taking the first stage in figure (4.2), \( C2 \) is the real capacitor and the op-amp and the remaining components form the gyrator. In this case, we do not want a particularly sharp response, so the Q is fairly low and constant for all the six stages. This is given by the equation below.

\[
Q = \prod f_0 Rf C………………………………………………………………………………… (4.1)
\]

The above design criteria for selecting the real capacitor and gyrator component values, is repeated using similar values for all six band stages but only changing values of the capacitors. The capacitor values changes because it the component which varies with frequency. Thus, the values of the capacitor in the next band stages are calculated by this equation

\[
C_n = C_p/\sqrt{10}………………………………………………………………………………… (4.2)
\]

Where \( C_p \) is the previous capacitor value from the bass side and \( C_n \) is the next capacitor value towards the treble side. The \( \sqrt{10} \) is the ratio of the next centre frequency over the previous one. \( R3, R2 \) and \( C2 \) all affect the "inductance"; the final output of the circuit is buffered by a unity gain op-amp stage. SWI selects whether the equalizer is in the audio path. The circuit requires a supply of +/-12V to +/-15V, at less than 10mA. This does not need to be regulated but must be smooth and have minima ripple.
Figure (4.2) six band graphic equalizer design

Figure (4.3) below is an equalizer equivalent with only one potentiometer, grounded via an imaginary resistor.
4.3 Amplifier design

To improve the full power efficiency of class A type amplifier, it's possible to design the amplifier circuit with two transistors in its output stage producing a "push pull" type amplifier configuration. Push pull operation uses two 'complementary' transistors; an npn type and the other an pnp type with both power transistors receiving the same input signal together that is equal in magnitude, but in opposite phase to each other. This results in one transistor only amplifying one half (or 180°) of the input waveform while the other transistor amplifies the remaining 180°. The resulting 'two halves' are put back at the output terminal.

While designing, it's very important to use an op amp with a high slew rate. This reduces crossover distortion. TIP121 and TIP126 complementary pair are 80V, 60W, both with a minimum beta of 1000 at 5A were identified.
The driver transistors, TIP41 and TIP42 are 65W, 100V, with a minimum beta of 15 at 5A. They are faster than the output complementary pair, with $f_T$ of 3MHz compared to 1MHz for the complementary output pair.

The transistors are biased using four diodes connected in series, such that their turn on/turn off points actually overlap, so that both transistors are in a state of conduction for a brief moment during the cross over period, resulting in a class AB. However, this results in increased power consumption of the circuit, because during the moments of the time where both transistors are conducting, there is current conducted through the transistors that is merely being shorted from one power supply rail to the other. This not only is a waste of energy but more heat is dissipated in the transistors. Page 643 (design). The peak output voltage ($V_P$) and peak output current ($I_P$) are calculated from the specified output power and load resistance ($R_L$)

$$V_P = \sqrt{2R_LP_O}$$  \hspace{1cm} (4.3)

The supply voltage ($V_{CC}$) calculated using the equation

$$V_{CC} = +/-[V_P + (I_PR_E) + V_{CE(sat)}]$$  \hspace{1cm} (4.4)

The output transistors and there respective drivers should have a breakdown voltage greater than the determined $V_{cc}$. The power transistors are able to dissipate approximately 40% of the maximum output power while the driver can dissipate this amount divided by the current gain of the output.

NB. The peak output voltage can be limited by the output voltage range of the op-amp. For most op-amps the output voltage range is 1V to 1.5V less than the positive and negative voltages. This can only be overcome by use of rail to rail op-amps. The emitter resistors for the output stage are typically selected to be any value between 5% to 10% of the peak output voltage.

$$R_{E2} = R_{E3} \approx 0.05R_L \text{ to } 0.1R_L$$  \hspace{1cm} (4.5)

When the peak load current flows ($I_P$), the peak base current to $Q_1$ and $Q_2$, is given by
peak base current flows bias network current \( I_B \) should be larger than the peak base current for both \( Q_1 \) and \( Q_2 \). \( R_1 \) and \( R_2 \) are equal value resistors that bias the op-amp input terminals. \( R_2 \) is calculated from \( R_3 \) to give the required voltage gain. Only a couple of capacitors are added to this circuit to bring it to the final form. A 100 \( \mu \)F capacitor connected between the base and emitter of the NPN transistor helps reduce the cross over distortion at low volume settings.

The combined impedance of resistor \( R_7 \) and \( R_6 \) was set to be slightly greater than 0.01V/\( I_{CEO} \) of the output transistor, \( Q_2 \). Their sum impedance was 3k\( \Omega \). The base current to resistor \( R_4 \) was calculated from equation (4.6). Resistors \( R_4 \) and \( R_5 \) are equal to avoid imbalance of the biasing currents to the driver transistors. Using equation 3.14, the value of the bootstrapping capacitors \( C_7 \) and \( C_8 \) was calculated at a cut off frequency of 100Hz. These capacitors are necessary to make resistors \( R_7 \) and \( R_8 \) behave like constant current sources. These capacitors bootstraps the collector of transistor \( Q_2 \).

Resistors \( R_{10} \) and \( R_{11} \) were used to set a gain of 20. \( R_7 \) and \( R_6 \) are equal to avoid voltage offset. Capacitors \( C_4 \) and \( C_6 \) set the lowest cut off frequency. In the final circuit, resistors of 0.01\( \mu \)F were added to bypass the input terminals of the op-amp to ground. This is necessary to reduce instability.

4.2.1 Circuit description

The 20Watt push pull amplifier is shown in figure 4.3. Operitional amplifier U1 provides the voltage gain given by \( A_V = (1 + R_{11}/R_{10}) \). The input signal is connected to the non-inverting input of the operational amplifier via a decoupling capacitor \( C_4 \) which sets the lowest cut off frequency at 100 Hz. Capacitors \( C_5 \) and \( C_6 \) are power supply bypass capacitors. Diodes D1 and D2 provide forward bias voltage to just turn the on the driver transistors \( Q_3 \) and \( Q_4 \) and prevent crossover distortion. Resistor \( R_4 \) and \( R_5 \) provide a constant base bias for transistors \( Q_3 \) and \( Q_4 \). Resistors \( R_6 \), \( R_7 \) and \( C_7 \) provide a constant current source to the transistor \( Q_3 \). Resistors \( R_1 \) and \( R_2 \) are the emitter resistors of the output transistors \( Q_2 \) and \( Q_1 \) respectively. Negative feedback is provided by resistors.
R11 and R10. Capacitors C8 and C7 are the bootstrapping capacitors to the output transistors.
Figure 4.3 Push pull amplifier design

Figure 4.5 The simulated results of the amplifier (input channel A and channel B)
4.4 Heat Sinks

Audio circuits require a voltage drop across the output transistors in order to reproduce the audio signal, hence they are not 100% efficient. Power is thus dissipated in the form of heat in the output devices. Heat sinks are mounted tightly to these transistors to provide a thermal connection between the two. Heat sinks ensure that the temperature of these output transistors does not rise to unsafe levels for the normal operation of the output devices.

Most heat sinks have got fins and or ridges to increase the surface area which allows the heat to be dissipated more quickly to the surrounding environment.
4.5 Resistor Choice

Most resistors used in this project are typically in the kΩ range. Resistors less than 1kΩ cause excessive current flow and this could possibly cause damage to the devices used. Those greater than 1MΩ cause excessive thermal noise and may make the circuit operation susceptible to significant errors due to otherwise negligibly small bias or leakage currents, hence they have not been used.

Circuit stability precautions

Power supply decoupling

Feedback along supply lines is a major source of op-amp instability. This can be minimized by connecting 0.01µF high frequency capacitors from each supply terminal to ground.

The capacitors must be connected as close as possible to the IC terminals. Sometimes, large value capacitors may be required.

4.5 General Stability Precautions

The following precautions should be observed for circuit stability:

1. Use 0.01µF capacitors or 0.1µF capacitors if necessary to bypass the supply terminals of op-amp(s) to ground. Connect these capacitors close to the ICs.

2. Do not connect oscilloscopes or other instruments at the op-amp input terminals. Instrument input capacitances can cause instability.

3. Keep all component leads as short as possible, and take care of the component placement. A resistor connected to an op-amp input terminal should have the resistor body placed close to the input terminal.

4. Where low frequency performance is required, use an internally compensated op-amp. Alternatively, use Miller effect compensation to give the lowest acceptable cut off frequency.
5. With an op-amp that must be compensated, use the methods and components recommended by the IC manufacture.

6. Always have a signal source connected to a circuit being tested. Alternatively, ground the circuit input. With an open circuited input, very small stray capacitances can cause instability.

7. Use small value resistors in the feedback network, if possible, instead of using the largest possible values.

8. If the circuit is unstable after all the above precautions have been observed, reduce the value of all circuit resistors (except compensating resistors). Also, reduce the signal source resistance if possible.
CHAPTER 5: REALIZATION, TESTING AND CONCLUSION

5.1 Construction

The two circuits were built on a Vero board. The equalizer and the amplifier modules were built on different boards but the output of the graphic equalizer was used to drive the amplifier.

5.2 Results

The two separate circuits were simulated on Protease version 7.2. For the amplifier, the input was set at 0.8V at a frequency of 6kHz in channel B. The output was 15.2V in channel A. The gain was thus 19. The oscilloscope was as shown in figure 4.5

For the graphic equalizer, an input of 0.2V at a frequency of 4 KHz in channel A, and with all the six controls set at 50%, the output was 0.85V volume at channel B. The oscilloscope display was as shown in figure 4.6.

5.3 Recommendations

The op-amp TL082CP that was used in the amplifier was not the ideal one because it has a high noise level. Thus I recommend the use of op-amp STK-405-050 which is a very low noise op-amp, and ideal for audio applications.

Last and not least recommend the use of a high performance graphic equalizer that covers the audio range with a higher number of bands in order to achieve the required frequency response
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**APPENDIX A PART LIST**

Graphic Equalizer design components list

**Resistors (0.5watt 5%)**

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<thead>
<tr>
<th>Resistor(s)</th>
<th>Value</th>
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<td>10K</td>
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**Potentiometer**

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**Capacitors**

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<tr>
<td>C1, 2</td>
<td>1µF</td>
</tr>
<tr>
<td>C3, 4</td>
<td>330nF</td>
</tr>
<tr>
<td>C5, 6</td>
<td>100nF</td>
</tr>
<tr>
<td>C7, 8</td>
<td>33nF</td>
</tr>
<tr>
<td>C9, 10</td>
<td>10nF</td>
</tr>
<tr>
<td>C11, 12</td>
<td>3.3nF</td>
</tr>
</tbody>
</table>

**Semiconductors**

<table>
<thead>
<tr>
<th>Semiconductor</th>
<th>Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>IC 1, 2,3,4,5</td>
<td>TL072</td>
</tr>
</tbody>
</table>
Miscellaneous

Vero board, Case, Knobs, Wire Two phono Sockets.

**Amplifier components parts list**

**Diodes**

D1, 2, 3, 4  1N4007

**Resistors (2 watt 5%)**

| R1, 3  | 47K    |
| R2     | 910K   |
| R6, 7  | 10K    |
| R8, 9, 10, 11 | 4K7   |
| R15    | 47R    |

**Capacitors**

| C1     | 333nF  |
| C2     | 33nF   |
| C3, 4  | 330µF  |

**Semiconductors**

| IC1    | TL082CP |
| Q1     | TIP42C  |
| Q2     | TIP121  |
| Q3     | TIP41C  |
| Q4     | TIP126  |
APPENDIX B Datasheet

TL072 Datasheets

Description

The TL072 is a high speed JFET input dual operational amplifier with a well matched high voltage JFET and bipolar transistors in a monolithic integrated circuit. It has a high slew rate, low input bias and offset current, and low offset voltage temperature coefficient.

Absolute maximum ratings and operating conditions

Table B1.1 Absolute maximum ratings

<table>
<thead>
<tr>
<th>Symbol</th>
<th>Parameter</th>
<th>Unit</th>
</tr>
</thead>
<tbody>
<tr>
<td>$V_{CC}$</td>
<td>Supply voltage (1)</td>
<td>+/- 18 V</td>
</tr>
<tr>
<td>$V_i$</td>
<td>Input voltage(2)</td>
<td>+/- 15 V</td>
</tr>
<tr>
<td>$V_{id}$</td>
<td>Differential input voltage</td>
<td>+/- 30 V</td>
</tr>
<tr>
<td>$R_{thja}$</td>
<td>Thermal resistance junction to ambient (4)</td>
<td></td>
</tr>
<tr>
<td></td>
<td>SO-8</td>
<td>40 $^0$C/W</td>
</tr>
<tr>
<td></td>
<td>DIP8</td>
<td>41 $^0$C/W</td>
</tr>
<tr>
<td>$R_{thjc}$</td>
<td>Thermal resistance junction to case (4)</td>
<td></td>
</tr>
<tr>
<td>$T_{stg}$</td>
<td>Storage temperature range</td>
<td>-65 to +150 $^0$C</td>
</tr>
<tr>
<td>ESD</td>
<td>HBM: human body models</td>
<td>1 $Kv$</td>
</tr>
<tr>
<td></td>
<td>MM: machine model (7)</td>
<td>200 V</td>
</tr>
<tr>
<td></td>
<td>CDM: charged device model(6)</td>
<td>1.5 kV</td>
</tr>
</tbody>
</table>
Table B 1.2 Operating conditions

<table>
<thead>
<tr>
<th>Symbol</th>
<th>Parameter</th>
<th>TL072</th>
<th>Unit</th>
</tr>
</thead>
<tbody>
<tr>
<td>V_{CC}</td>
<td>Supply voltage</td>
<td>6 to 36</td>
<td>V</td>
</tr>
<tr>
<td>T_{oper}</td>
<td>Operating free air temperature range</td>
<td>0 to +70</td>
<td>°C</td>
</tr>
</tbody>
</table>

TL 082 Datasheet

Description

This device is a low cost, high speed, dual JFET input operational amplifier with a very high slew rate. It requires low supply current yet it maintains a large bandwidth. The well matched high voltage JFET input devices provides a very low input bias and offset currents.

Table B1.3 Absolute maximum ratings and operating conditions

<table>
<thead>
<tr>
<th>Symbol</th>
<th>Parameter</th>
<th>Conditions</th>
<th>Typical</th>
<th>Maximum</th>
<th>Units</th>
</tr>
</thead>
<tbody>
<tr>
<td>V_{OS}</td>
<td>Input offset voltage</td>
<td>R_S =10kΩ, T_A =25°C</td>
<td>5</td>
<td>15</td>
<td>mV</td>
</tr>
<tr>
<td>I_{OS}</td>
<td>Input offset current</td>
<td>T_J =25°C</td>
<td>25</td>
<td>200</td>
<td>pA</td>
</tr>
<tr>
<td>I_B</td>
<td>Input bias current</td>
<td>T_J=25°C</td>
<td>50</td>
<td>400</td>
<td>nA</td>
</tr>
<tr>
<td>I_S</td>
<td>Supply current</td>
<td>3.6</td>
<td>5.6</td>
<td>nA</td>
<td></td>
</tr>
<tr>
<td>R_{IN}</td>
<td>Input resistance</td>
<td>10^{12} Ω</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>PSSR</td>
<td>Supply voltage rejection ratio</td>
<td>70</td>
<td>100</td>
<td>dB</td>
<td></td>
</tr>
<tr>
<td>CMRR</td>
<td>Common mode rejection ratio</td>
<td>R_S≤ 10 kΩ</td>
<td>70</td>
<td>100</td>
<td>dB</td>
</tr>
</tbody>
</table>
Table B 1.3 Operating conditions

<table>
<thead>
<tr>
<th>Symbol</th>
<th>Parameter</th>
<th>TL082</th>
<th>Unit</th>
</tr>
</thead>
<tbody>
<tr>
<td>V_{CC}</td>
<td>Supply voltage</td>
<td>6 to 36</td>
<td>V</td>
</tr>
<tr>
<td>T_{oper}</td>
<td>Operating free air temperature range</td>
<td>-0 to 70</td>
<td>°C</td>
</tr>
</tbody>
</table>

COMPLEMENTARY SILICON POWER TRANSISTORS

TIP41C/TIP42C

Description

The TIP41A, TIP41B ,TIP41C are Silicon Epitaxial Base NPN power transistor mounted in a Jede TO-220 plastic package. They are intended for use in medium power linear and switching applications.

The TIP41A and TIP41C complementary PNP types are TIP42A and TIP42C respectively.
Table B1.4 Absolute maximum ratings

<table>
<thead>
<tr>
<th>Symbol</th>
<th>Parameter</th>
<th>Value</th>
<th>Unit</th>
</tr>
</thead>
<tbody>
<tr>
<td>V_CBO</td>
<td>Collector base voltage (I_E=0)</td>
<td>100</td>
<td>V</td>
</tr>
<tr>
<td>V_CEO</td>
<td>Collector emitter voltage (I_B = 0)</td>
<td>100</td>
<td>V</td>
</tr>
<tr>
<td>V_EBO</td>
<td>Emitter base voltage (I_C = 0)</td>
<td>5</td>
<td>V</td>
</tr>
<tr>
<td>I_C</td>
<td>Collector current</td>
<td>6</td>
<td>A</td>
</tr>
<tr>
<td>I_CM</td>
<td>Collector peak current</td>
<td>10</td>
<td>A</td>
</tr>
<tr>
<td>I_B</td>
<td>Base current</td>
<td>3</td>
<td>A</td>
</tr>
<tr>
<td>P_tot</td>
<td>Total dissipation at, T_case ≤ 25°C</td>
<td>65</td>
<td>W</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>w</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>T_stg</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>Storage temperature</td>
<td>-65 to 150</td>
<td>°C</td>
</tr>
<tr>
<td></td>
<td>T_J</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>Maximum operating junction temperature</td>
<td>150</td>
<td>°C</td>
</tr>
</tbody>
</table>

Table B1.5 Thermal characteristics

<table>
<thead>
<tr>
<th>Characteristics</th>
<th>Symbol</th>
<th>max</th>
<th>Unit</th>
</tr>
</thead>
<tbody>
<tr>
<td>Thermal resistance junction to ambient</td>
<td>R_ΘJA</td>
<td>62.5</td>
<td>°C/W</td>
</tr>
<tr>
<td>Thermal resistance junction to case</td>
<td>R_ΘJC</td>
<td>1.92</td>
<td>°C/W</td>
</tr>
</tbody>
</table>