DESIGN AND CONSTRUCTION OF A STEREOPHONIC HEARING AID.

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DEDICATION

This final year project is dedicated to God almighty, the fountain and arbiter of my life.

DECLARATION

I Lucky Oria Idemudia, declare that this work was done by me and has never been presented elsewhere for the award of a degree. I also hereby relinquish the copyright to the Federal University of Technology, Minna.

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ABSTRACT

This project titled ``design and construction of a stereophonic hearing aid`` with a volume control is poised towards helping people with impaired hearing. In the world we live today, there is a great increase in the number of hearing impaired persons due to noise pollution in urban areas, over exposure to industrial noise, infections, deformities at childbirth and ageing. As the need to correct this problem increased, it has led to the development of electronics hearing aids which are devices that amplify sound signals, and boost sound level to compensate for defects in the eardrum structure of the impaired ear. The various units consist of the power unit which is made up of 9volt battery. The preamplifier stage which is made up of a micro power operational amplifier LM358, which amplifies the picked up voice signal from the input transducer (microphone), converts the signal (sound) into electric signals. The electric signals is fed into the power amplifier stage (LM386), which amplifies the signals and the amplified signal is converted back to sound at the output stage (earphone) i.e. the output transducer.

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CHAPTER ONE

1.1 INTRODUCTION

Due to the alarming rate of hearing impairment in our society, as a result of detrimental effects caused by loud noise produced by heavy machines in operation., The use of explosive by civil engineers to break down rocks, hills or mountains when constructing roads, bridges across them, exposure to loud music played in concerts, and loud music blasting from loud speakers, loud noises from the horns of motor vehicles, noises from railway stations and airports, ageing, birth defects, the actions of viruses, bacteria and other disease causing organisms to our ears. It became imperative to device a solution which will aim at ameliorating the problem of hearing impairment. In quest for this solution gave rise the conception of the design and construction of a stereophonic hearing aid.

A stereophonic hearing aid is an electronic device which converts sound signals to electrical signals and back to sound signal. However, it is an electroacoustic body worm device or apparatus which is typically fits in or behind the wearers' ear and is design to amplify and modulate sounds. The device work on the principle of amplification of sound.

1.2 AIMS AND OBJECTIVES

The aims and objectives of this project are as follows;

1. To design and construct a device which would help a great deal to alleviate the suffering of people with partial hearing disabilities in our society.

2. To ensure that this device is made from electronics components that is easily sourced and available locally, and at an affordable price for the poor masses suffering from hearing disabilities.

3. To enable the children as well as adults, with hearing disabilities to learn with ease alongside their mates with proper hearing abilities in convectional schools, thereby reducing the burden of expensive school fees paid by parent of children with hearing disabilities in special school made for them.

4. Another objective of this project is to ensure social inclusion for those with hearing problems, and also reduce the stigmatization always inflicts in them.

5. Finally, to ensure the comfort and convenience individual suffering from one form of hearing disabilities to another, and also to ensure the efficiency of such individual if employed.

1.3 METHODOLOGY

In order to meet up with the aims and objectives of this project "design and construction of a stereophonic hearing aid", an electrets microphone with a high sensitivity was selected among many. Similarly, in the amplifier stage, an LM358, a dual operational amplifier which has the capacity of amplifying low level millivolt-range microphone signals was also selected. Besides, LM386; a power amplifier which is capable of amplifying the signal from the pre-amplifier stage was chosen because it meets the system requirement such as a power gain of 46 dB, high power output, high sensitivity, and high efficiency. A volume control was integrated to help in a great deal to regulate the variation of signal before it is fed into the power amplifier. A 9 volt battery was used because it has the best trade off in terms of size and lifespan.

In the construction of the device, a high sensitive electrets microphone was connected to the small signal amplifier (LM358), configured in the inverting mode to reduce noise pickups. The input signal was fed to the comparator where comparism is made before it is transferred to the overload detector (CD4066, a quad bilateral switch) otherwise, the signal is fed directly to the power amplifier, and finally to the headphones.

1.4 SCOPE OF THE WORK

The project is focus to provide solution of suitability, audibility and comfortability to people with hearing disabilities. The device is designed to operate at a range of frequency between 20HZ to 20,000HZ which is the range of frequency human being can hear. The device provides a gain of 46 decibel and the distance of operation is from 0 to 80 metres.It will interest to note here that at a distance above 50 metres the sensitivity of the microphones tend to reduce. This is due to the fact that at low frequency it is the microphone that causes limitation to the hearing aid. While at high frequency it is the earphones that is responsible for the limitation. For future improvement, a more powerful amplifier should be employed so as to cover a much greater distance, and increase in the gain of the device. Besides, miniaturization of the device should be put into consideration so that it can come out in a more portable form.

1.5 SOURCES OF MATERIAL USED

Most of the materials used (in form of electronics components) in carrying out this work were sourced locally. This was done intentionally so as to make the final product easily affordable. Similarly, the sources of information used in the production of this device includes, the internet, libraries(both school and national).past work of students, consultations with friends, technicians and other experts in the field.

1.6 PROJECT OUTLINE

Chapter one involves the introduction to the subject matter, the aims and objectives, the methodology, the scope of work and the sources of material used. Chapter two contains literature review and the historical background of the device. Chapter three contains the theoretical background, and the design implementations. Chapter four contains testing of the work, results and discussion of results obtained, troubleshooting tips etc.Finally, chapter five contains summary of the project work and recommendations.

CHAPTER TWO

LITERATURE REVIEWS/HISTORICAL BACKGROUND

2.1 HISTORICAL BACKGROUND

Since the advent of medicine, especially in the study of otology, early scientists and otologist had strived to obtain a solution for the problems suffered by individuals with hearing impairment. The strive to obtain solution to this problems led to the invention of the earliest hearing aid which were in form of ear trumpets invented sometimes in the 17th century[1,9].He trumpets were made of long horns with a large opening at one end, and smaller at the other end, which was placed in the ear. The trumpet work under the principle that; sound pressure waves entering the large end recondensed into smaller volumes, hereby increasing the audible sound pressure. With the invention of the telephone by Alexander Graham Bell in 1876, came the next phase of development of the hearing aid. This type of hearing aids were referred to as carbon aids, and worked on the principle that sound waves were converted to electrical waves, and the back to sound waves.However,they had some major drawbacks in the sense that they used very large batteries and could only help people with moderate hearing loss[3].

By the 1920s a more sophisticated telephone type of hearing aids were developed which looked modern days hearing aids; with a microphone, electrical circuit ,diaphragm and battery. But this type of hearing aids used vacuum tube technology. It was popularly used through the 1930s[3,13]. With the advent of transistor in 1948, hearing aids of greatly reduced sizes and weights were invented. In the year 1969 the first direction microphones was developed and was incorporated into hearing aids, thereby leading to a more natural sounding hearing aids[3].

By the mid 70s, integrated circuits were applied to the hearing aids to help users distinguish between speech and background noises, also the use of the latest tiny batteries allowed for the In-the-canal (ITC) hearing aids to be developed. Facilitated by the introduction of lithium batteries and the advent of digital signal processing (DSP) in the 1980s, hearing aids were beginning to return into mini computers. By this period also ideal of surgical implants to the cochlea was widely anticipated as cures for deafness were the use of hearing aids could not do much.

By the 90s,hearing aid that could boast of two channel sounds automatic volume control , or remote control for the smallest of ITC(in-the anal) instruments by now when worn were totally indivisible to all. By this time also, fully digital audio processor were developed, global researchers took it upon themselves to produce a working hearing aid system which they named adaptive speech alignment, which could boast of multi tone banding, and dual processing, one for recognizing vowels and consonants respectively.
Hearing aids of present time have self assessment of listening comfort, and memory cards for the remote controls. Micro-magnets implanted next to the eardrum, which will never need to be replaced is being foreseen as a major development in the hearing aids of the future. The development of the electronic hearing aids was described as the biggest advancement ever made by science and engineering technology in helping deaf people [1, 11].

There are many types of hearing aids, which vary in size, power and circuitry. Among the different sizes and models are:

Body worn aids: This was the first types of hearing aid invented by Harvey Fetcher while working in at Bell laboratories; thanks to development in technology they are now rarely used. This consists of a case containing the components amplification and an ear mold connected to the case by a cord. The case is about the size of a pack of playing cards and is worn in the pocket or on a belt. Because of large size, body worn aids are capable of large amount of amplification and were once used for profound hearing loss.

Behind the ear aids(BTE):These have a small plastic case that can that can fits behind the pinna (ear) and provides sound to the ear via air conduction of sound through a small length of tubing, or electrically with a wire and miniature speaker placed in the ear canal. The delivery of sound to the ear is usually through the ear mold that is custom made, or other pliable fixtures that contours to the individual's ear.BTE can be used for mild to profound hearing losses and are especially useful for children because of their durability and ability to connect to assertive listening devices such as classroom FM systems.

In the ear aids (ITE): This device fit in the outer ear bowl(called the cochlea);they are sometimes visible when standing face to face with someone.ITE hearing aids are custom made to fit each individual's ear. They can be used in mild to some severe hearing losses

Receiver in the ear aids (RITE): These devices are similar to the BTE aid. There is however one crucial difference: The speaker(receiver) of the hearing aid is placed inside the ear canal of the user and thin electrical wires replace the acoustic tube of the BTE aid. There are some advantages with this approach: Firstly, the sound of the hearing aid is arguably smoother than that of a traditional BTE hearing aid. With BTE, the amplified signal is emitted by the speaker which is located within the body of the hearing aid (behind the ear).the amplified signal is directed to the ear canal through an acoustic tube, which create peaky frequency response. The speaker is right in the ear canal and the amplified output of the hearing aid does not need to be pushed through an acoustic tube to get there, and is therefore free of this distortion. Secondly, it can typically be made with a very small part behind-the-ear and the wire connecting the hearing aid and the speaker is extremely inconspicuous[1,3].

In-The-Canal (ITC), mini-in-the-canal (MIC) and completely-In-The-Canal (CIC) aids: ITC aids are smaller, filling only the bottom half of the external ear. You usually cannot see much very of this hearing aid when you are face to face with someone.MIC and CIC aids are often not visible unless you look directly into the wearers ear. These aids are intended for mild to moderately-severe losses.CICs are usually not recommended for people with good low frequency hearing, as the occlusion effect is much more perceivable.

Bone Anchored Hearing Aids (BAHA): This auditory prosthetic which can be surgically implanted, the BAHA uses the skull as a pathway for sound to travel to the inner ear. For people with conductive hearing loss, it bypasses the external auditory canal and middle ear, stimulating the functioning cochlea. For people with unilateral hearing loss, it uses the skull to conduct the sound from the deaf side to the side of the functioning cochlea.

Eye glass hearing aids: During the late 1950s through 1970s, before in-the ear aids became common, people who wore both glasses and the hearing aids frequently choose a type of hearing aid that was built into the temple pieces of the spectacles. However the combination of glasses and the hearing aids was inflexible: The range of frame styles was limited, and the user had to wear both hearing aids and the glasses at once or neither. Today, people who use both glasses and hearing aid can use in-the ear types, or rest a BTE neatly alongside the arm of the glasses. There are still the specialized situations where hearing aids built into the frame of eyeglasses can be useful, such as a person has hearing loss mainly in one ear: Sound from a microphone on the bad side can be sent through the frame to the side with better hearing[1, 3].

CROS (contralateral routing of offside signal) type hearing aids: This is wireless in form and it sends sound from the bad side to the better side of the ear. Recently, a new type of eyeglass aid was introduced. The hearing glasses features directional sensitivity: Four microphones on each side of the frame effectively work as two directional microphones, which are able to discern between sound coming from the front and sound coming from the side or back of the user. This improves the signal-to-noise ratio by allowing sounds coming from the sides or back. Only very recently has the technology required become small enough, in size, to be put into the frame of glasses. As a recent addition to the market, this hearing aid is currently available only in Netherlands and Belgium[3].

CHAPTER THREE

3.1 THEORETICAL BACKGROUNDS

Hearing aids are fundamentally simple acoustic amplifying systems and are made up of four basic parts which includes;

1. A microphone to pick up sounds and convert them into very small electrical signal.

2. An amplifier to increase the size of the electrical signal

3. An earphone to return the electrical signal into an acoustic one which is fed into the ear through an ear mould.

4. A power source which is usually a battery for powering the amplifier. The figure below shows the block diagram of typical hearing aid units

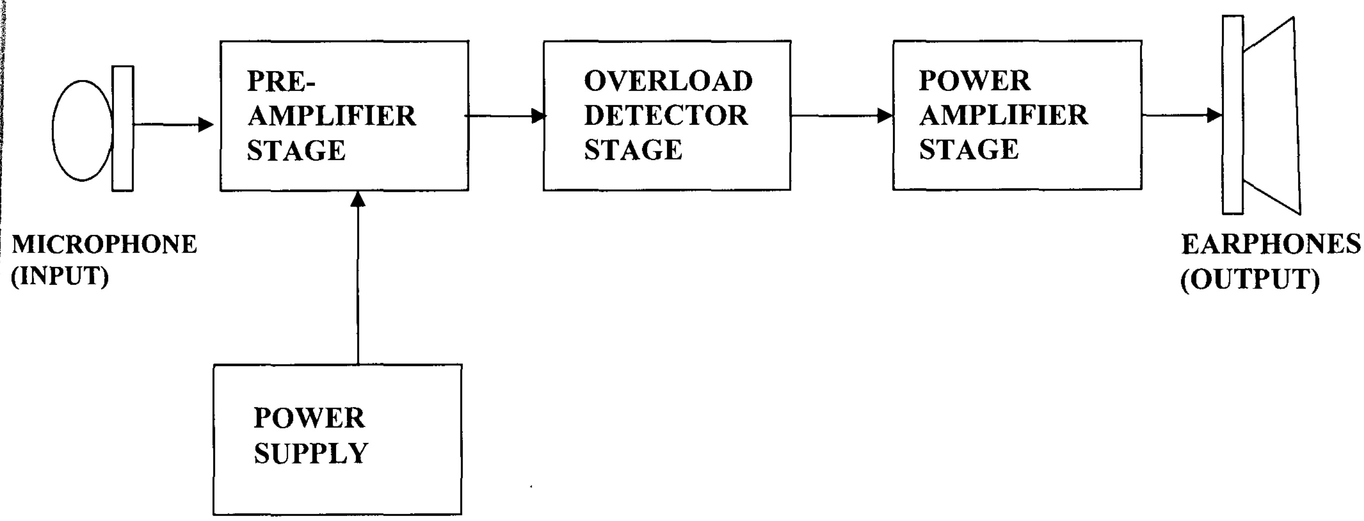

Fig 3.1: Block Diagram of a Typical Hearing Aid

3.1.1 THE INPUT STAGE

A microphone makes up the input stage of a typical hearing aid. Microphones are devices which acts as transducers to convert acoustic signals (sound waves) into electrical signals [8].Microphones come in all shapes and sizes. When choosing a microphone for any particular application, the following features needed to be considered;

The frequency response on axis: The microphone should respond equally to sounds over the whole range of frequency range of interest. Thus in high quality system the graph of signal output voltage plotted against frequency for a constant acoustic level input over the range 20-20000Hz(the normal limit of human hearing) should be a straight line.

Directivity: Microphone directivity is its ability either to respond equally to sound arriving from all directions or to discriminate against sounds from particular direction.

Frequency response off-axis: Ideally any high quality microphone whatever its directivity pattern, should maintain the same frequency response at all angles; there is a need for polar pattern uniformity.

Sensitivity: The conversion efficiency of a microphone i.e. the output voltage produced by a given incident sound pressure level, should be as high as possible.

Self noise: The inherent noise level of a microphone and this include any built-in-amplifier should be as low as possible.

Distortion: For the waveform of the electrical signal to be a faithful representation of the original sound wave, on-linear distortion must be as low as possible. Such distortions are mainly associated with high signal levels and the onset of overload or saturation effects in the transducer mechanism [2].

The following are the types of microphones;

1. The moving-coil (dynamic) microphone

2. The ribbon microphone

3. The condenser microphone (capacitor or electrostatic)

4. The electrets microphone

5. The piezoelectric (crystal) microphone

6. The carbon (loose contact) microphone.

Microphones use diaphragms which respond to sound waves in the following ways: The magnitude of the change in air pressure on the diaphragm as in the case of carbon-granules and crystal microphones) has a direct effect on the electrical output. The difference between the pressure at closely situated points, i.e. pressure gradient (as in the case of a moving-coil, and ribbon microphones) may results in an electrical output. The microphone used in the construction of this hearing aid is the condenser microphone. It was chosen, because of its high sensitivity, portability, and due to the fact that it could be made very small and rugged.

3.1.2 AMPLIFICATION STAGE

The main component of the amplification stage in a typical hearing aid is the audio amplifier. Audio amplifiers are electronic amplifiers that increase low audio signals (composed of frequencies between 20Hz to 20 kHz, which makes up the human hearing range) to a level high enough for driving a load speaker[12], or in this case an earphone, which normally makes up the final stage of audio playback system. Certain stages precede the amplification stage. When designing an audio amplifier for use in device such as hearing aids for example, the following parameters are to be considered [12].

Frequency response of the amplifier: This is the measure of the system response to a signal of varying frequencies at the output with respect to constant amplitude. It is usually measured in decibels.

Gain of amplifier: This is defined as the mean ratio of the signal output of a system to the signal input of the same system. When measured in decibels (dB), power gain is given as;

$$Gain = 10\log\left[\frac{p_{out}}{p_{in}}\right] \text{dB} \tag{3.1}$$

Where p_{in} and p_{out} are input and output power respectively.

If voltage is used to calculate the gain of an amplifier, we obtain from the expression,

$$P = I^2\mathrm{R} \tag{3.2}$$

$$I = \frac{V}{R} \tag{3.3}$$

Substituting equation (3.3) into equation (3.2), we obtain;

$$P = \left[\frac{V}{R^2}^2\right]R \tag{3.4}$$

$$P = \frac{V^2}{R} \tag{3.5}$$

Substituting equation (3.4) into equation (3.1) we obtain;

$$Gain = 10\log\left\{\frac{V_{out}^2}{R_{out}}\middle|\frac{V_{in}^2}{R_{in}}\right\} \tag{3.6}$$

Since input and output impedances are usually equal in most cases, equation (3.6) could be simplified as;

$$Gain = 10\log[\frac{V_{out}}{V_{in}}]^2 \text{ dB} \tag{3.7}$$

Therefore,

$$Gain = 20\log\left[\frac{V_{out}}{V_{in}}\right] \text{dB} \tag{3.8}$$

Equation (3.8) is used to calculate gain in decibels only when both input and output impedances are equal.

Distortion of the amplifier: This is the alteration of the original shape to unwanted waveform by the amplifier. Distortion of audio systems could be corrected by using special filters known as equalizers.

3.1.3 OUTPUT STAGE

The output stage of a typical hearing aid system is made up of tiny loudspeakers known as earphone or headphones in some cases. The headphones act as transducers which convert the amplifier electrical signals from the audio amplifier, back to sound waves. The headphones have an advantage of not exciting room resonances and thus giving the listener a more accurate sense of the recorded acoustics [2].There are different types of earphones and are classified based on their principle of operation. These include;

Moving iron earphones: This type of earphones lie on the use of a fine wire wound onto magnetic yoke held lose to stiff disc made of soft magnetic alloy such as stalloy. Permanent magnet is used to pull the thin disc towards the yoke with a constant force, and audio signal fed into the coil causes this force to vary with respect to the input. Moving iron earphones are very sensitive, needing hardly any power to drive them. They have very poor sound quality and are used in telephone receivers.

Moving coil earphones: Here is a coil of wire suspend in a radial magnetic field in an annular magnetic gap is connected to a small radiating cone. When alternating audio signals are applied to the coils, the coil vibrates axially with respect to the signals, thereby recreating an analogue form of the original wave shape. The cone then converts this into the corresponding fluctuations in air pressure with the listener then perceives as sound.

Electrodynamic/othodynamic earphones: These types of earphones are similar to the moving coil type, but in this case the coil is unwound and fixed to a thin light plastic diaphragm. The annular magnetic gap has been replaced by opposing bar magnets which cause the magnetic field to be squashed more or less parallel to the diaphragm. The coil in this case is now a thin conductor zigzagging or spirally its way across the surface of the diaphragm, oriented at right angles to the magnetic field, so that sending a constant direct current through the conductor results in a more or less equal unidirectional force which displaces the diaphragm from its rest position. As a result of all this, an alternating sound signal therefore causes the diaphragm to vibrate with respect to it. Thereby creating an analogue form of the sound waves[2].

Electrostatic earphones: Just like the electrodynamics earphones, the electrostatic earphones use thin plastic diaphragms, but in this case, instead of a copper track, the diaphragm is treated to make it very slightly conductive and that the surface can hold an electrostatic charge. This makes it usually light. The diaphragm is stretched under low mechanical tension between two perforated conductive plates to which the audio signal are fed through a step up transformer. There is a central diaphragm which is kept charged to a very high voltage with respect to the outer plates using special type of power supply, capable of non-dangerous, low current, high voltage from the house mains or alternatively by an energizer which uses some of the audio signal to charge the diaphragm to a similarly high but safe voltage as a result of this, the diaphragm experiences electrostatic attractions toward the outer plates. Care is taken to ensure that the film does not collapse on any of the plates, but stays stable between the outer plates and is attracted to each one equally during non-signal condition. When an audio signal of a few milivolts is applied to the primary terminal of the step-up transformer, it is stepped up at the secondary to around a thousand volts. This unbalances the force on the diaphragm with respect to the audio signal, causing it to be attracted alternately to each plate, thereby reproducing an analogue form of the original sound.

Electrets earphones: These types of earphone are similar to the electrostatic types but use the electrostatic equivalent of a permanent magnet (i.e. a material which permanently retains electrostatic charges) unlike the electrostatic types, the electrets earphone does not require the use of an additional power supply. Earphone could be further classified based on the mode of wearing them such as [13].

1. Circum aural
2. Supra aural
3. Ear buds
4. Canal earphones.

The moving coil earphone was made use of in the construction of this hearing aid. The type used is the ear bud type and this was chosen because of its high sensitivity of the earphones and its matching impedance with the hearing aid unit and also its availability.

3.1.4 POWER SOURCE

This section of the hearing aid deals with the type of power source required to power the audio amplifier and the pre-amplifier stages of the hearing aid. There are various sources of power which could be used to power the hearing aid amongst which includes,

1. Solar power
2. Batteries.

The power source used in this hearing aid is a battery since it is cheaper and easily available.

3.1.5 AUDIBILITY OF THE HEARING AID

The purpose of the hearing aid is to amplify the sound sufficiently so that they can be heard by listener. Audibility is the key concept underlying speech perception. Generally speaking, the more speech sound we hear, and the better we understand. In describing audibility, the following characteristics are considered.

Audible range: The limit of frequencies which are audible. The normal ear of young adult detects sound hearing frequencies in the range 16Hz to 16 kHz; although it is possible for some people to detect frequencies outside these limits.

Decibel [dB]: The level of noise is measured objectively using a sound level meter. This instrument has been specifically developed to mimic the operation of the human ear. The human ear responds to moderate pressure variation in air. These pressure variations can be likened to the ripples on the surface of water. The pressure variation in air causes the ear drum to vibrate and this is heard as sound in the brain. The stronger the variation, the louder the sound. The pressure of sound waves is normally quoted in Pascal (Pa); but because of the range of pressure involved, logarithmic unit, the decibel was introduced. It is important to state that the range of decibel of the hearing aid is from 0 to 46dB. The table below shows the pressure of sound in Pascal and in Decibel.

Table 3.2: Pressure of Sound in Pascal and in Decibel.

Decibel(dB)	Pascal(Pa)	Comments
-6	10μ	Inaudible
0	20μ	Threshold of hearing
40	2000μ	Very quite speech
80	0.2	Loud speech
100	2	Damaging noise
120	20	Becoming painful

3.2 DESIGN AND IMPLEMENTATION

The step taken in the design of this stereophonic hearing aid system is a modular approach. Here each module would be carefully explained, and their circuit diagrams drawn. The stereophonic hearing aid system comprises of the underlisted subsystems;

1. The power supply unit
2. The microphone transducers
3. The microphone small signal pre-amplifiers
4. The overload detector / attenuator
5. The power amplifier.

A block diagram of the various subsystems which makes up the design of this stereophonic hearing aid system is as shown in the figure below.

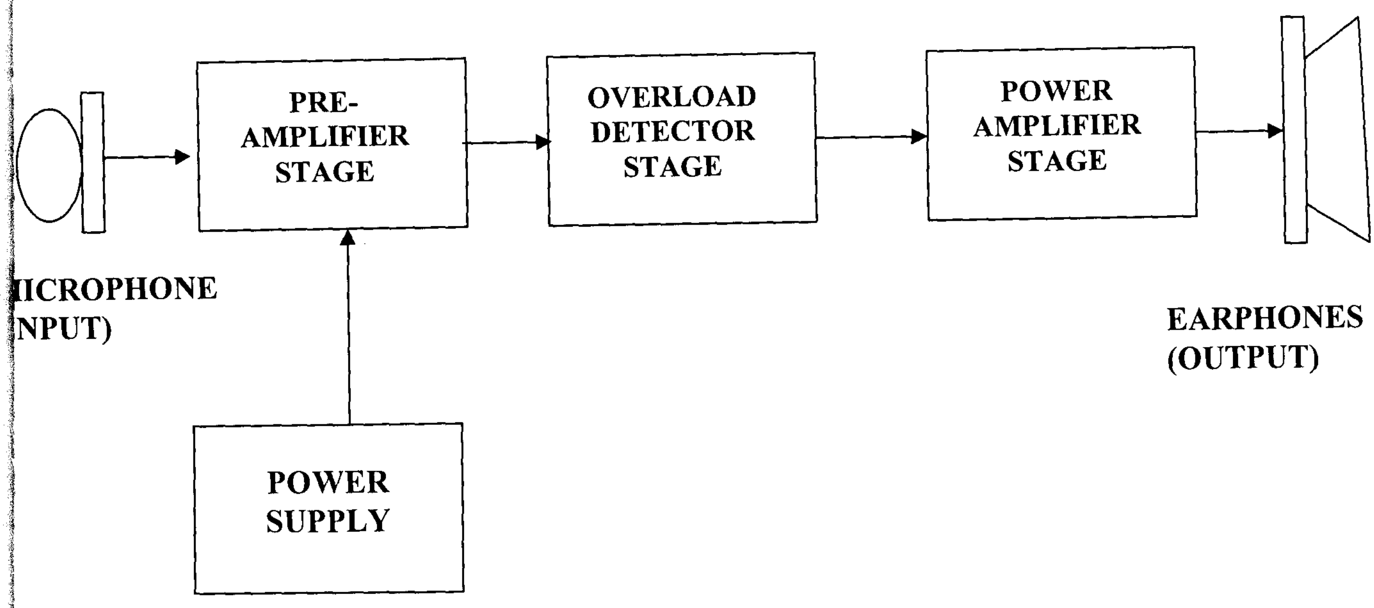

Fig 3.3: A block Diagram of a Stereophonic Hearing Aid System.

3.2.1 SYSTEM REQUIREMENT

In other to produce a stereophonic hearing aid of a very high quality, the following systems requirements were put into consideration;

1. High power output (at least 1 watt),
2. High sensitivity,
3. Low noise distortion,
4. High efficiency,
5. High sonic clarity.

Based on these requirements mentioned above, the system components were selected from among many. A power amplifier capable of providing the required power output level, coupled with a less voltage operation as low as 4volts was required, and the LM386 integrated circuit meets this requirement. Besides, LM386 was chosen because of the following characteristics.

1. Few external components,

2. Low voltage consumption (this makes it ideal for battery operation),

3. Wide supply voltage range 4-12v or 5-18v,

4. Low quiescent current drain of 4mA,

5. Voltage gains from 20 to 200 (i.e. 21 to 46 dB)

6. Low distortion of 0.2%.

NB: For more information please see appendix 1(Datasheet of LM386).

Similarly, in the preamplifier stage, the LM385 micro power 3-terminal voltage reference operational amplifier was chosen. This is as a result of its unique characteristics such as;

1. Extremely low power drain,

2. Low temperature coefficient,

3. Low noise,

4. Low dynamic impedance,

5. Wide dynamic operating range varying supplies with excellent regulation,

6. Adjustable voltage range from 1.24 to 5.3v and good temperature stability.

In other to obtain high sensitivity, a condenser microphone capable of picking sound signals over a wide range was used. Similarly, a moving coil ear bud type of earphone was used to obtain a high sonic clarity from the hearing aid.

3.2.2 POWER SUPPLY

Due to the necessity of making the unit body-wearable by the user, Dc power source was utilized. A 9 volts battery was selected as it gives the best trade of in terms of size against battery lifespan.

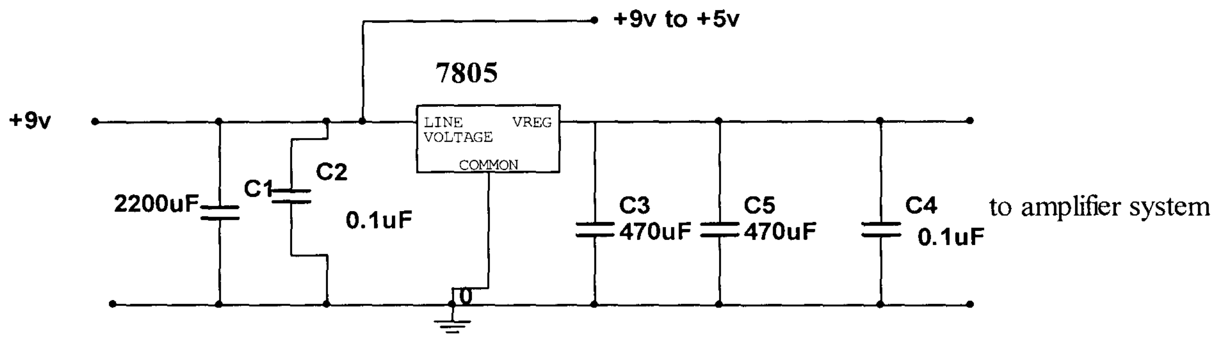

Fig. 3.4: System Power Supply.

A 2200uF capacitance connected across the 9V power source provided buffering of the input voltage. The parallel connected 0.1uF provided filtering. The 9V was regulate down to +5V required for small signal as depicted in fig 3.4

3.2.3 MICROPHONE PICKUPS

Two electrets microphones were utilized as the input trasducers.The microphones were biased ON from the 5V regulated supply as shown below.

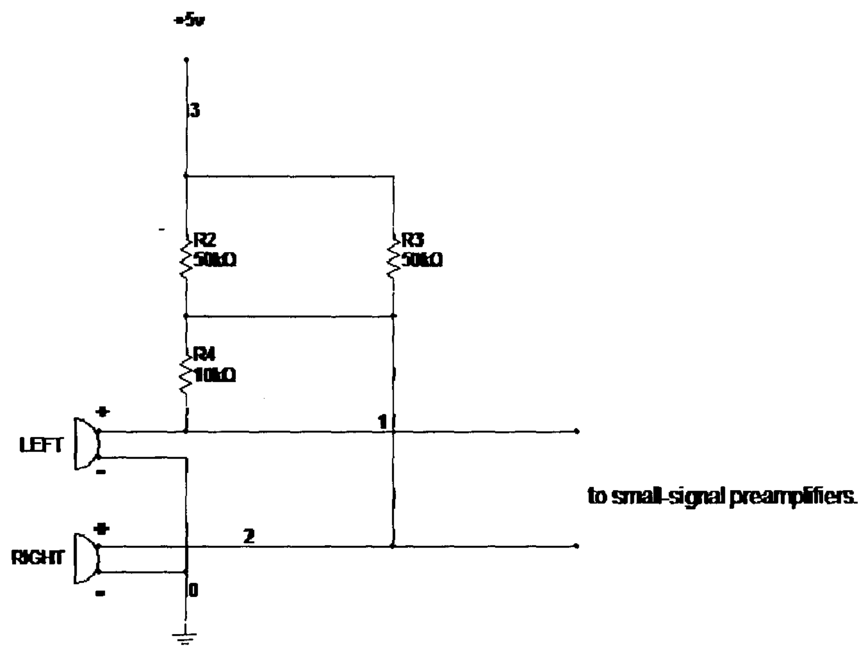

Fig.3.5: Biasing of Electrets Microphones.

To cater for any dc bias requirements, a variable-voltage bias source was implemented on board. Two separate biased systems were used to minimize crosstalk and make for the highest possible separation of the stereo image in the sound field. Two 50kΩ preset resistors were connected across the 5 volt supply as in fig 3.4 The DC biased levels were adjusted until a high-enough fidelity in the amplified version of a test sound source was obtained. The bias voltages were measured at 1.2 and 1.35 volts respectively on the left and right microphones.

The resistances were chosen high enough to reduce the drain on the power supply.

The current drain by each resistor was easily calculated using

$$I = \frac{V}{R} = \frac{5}{50{,}000} = 100\mu A \qquad (3.9)$$

3.2.4 SMALL SIGNAL AMPLIFIER

An LM358 dual operational amplifier was used for amplifying the low level milivolt-range microphones signal. The operational amplifiers were configured in the inverting mode to reduce noise pick up.

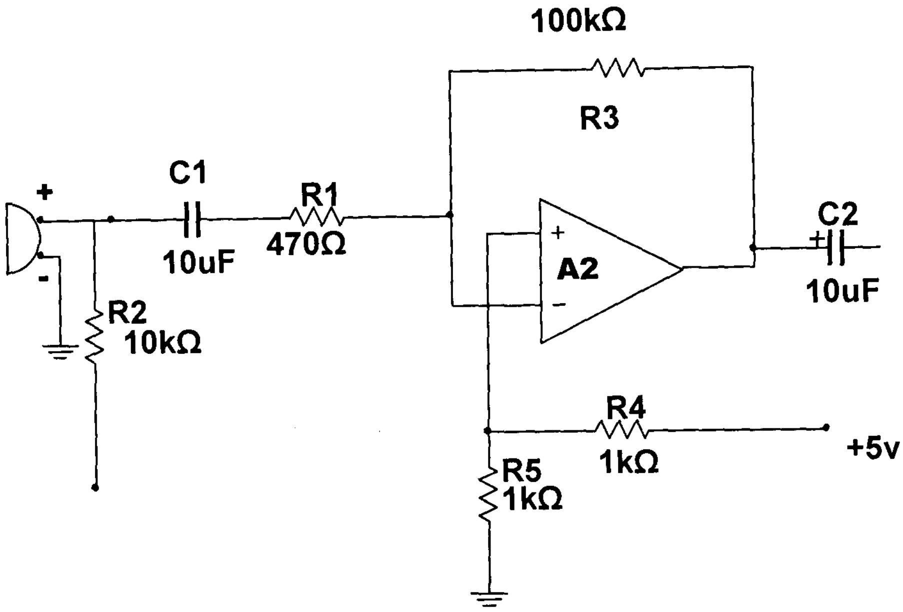

Fig.3.6: Microphone Amplifiers

The gain factors of the amplifiers were deduced from the expression:

$$\frac{V_{out}}{V_{in}} = -\frac{R_f}{R_i+R_s} \qquad (3.10)$$

Where R_f =100kΩ (The feedback resistance)

R_i =470Ω (The input resistance)

R_s =2.2kΩ (The Microphone impedance)

Thus, the amplifiers voltage gain:

$$Vout/Vin = -100{,}000/(470 + 2200) \cong 45 \qquad (3.11)$$

An input high pass filter was formed by the 10uF coupling capacitor and the reset 470Ω resistance. The cut-off frequency was evaluated using:

$$F = 1/2\pi RC = 1/2\pi\,(470x10 \times 10^{-6})\ \text{Hz} = 33.85\text{Hz} \qquad (3.12)$$

The amplified voltage was fed into an overload detector/attenuator as shown in figure 3.7 below.

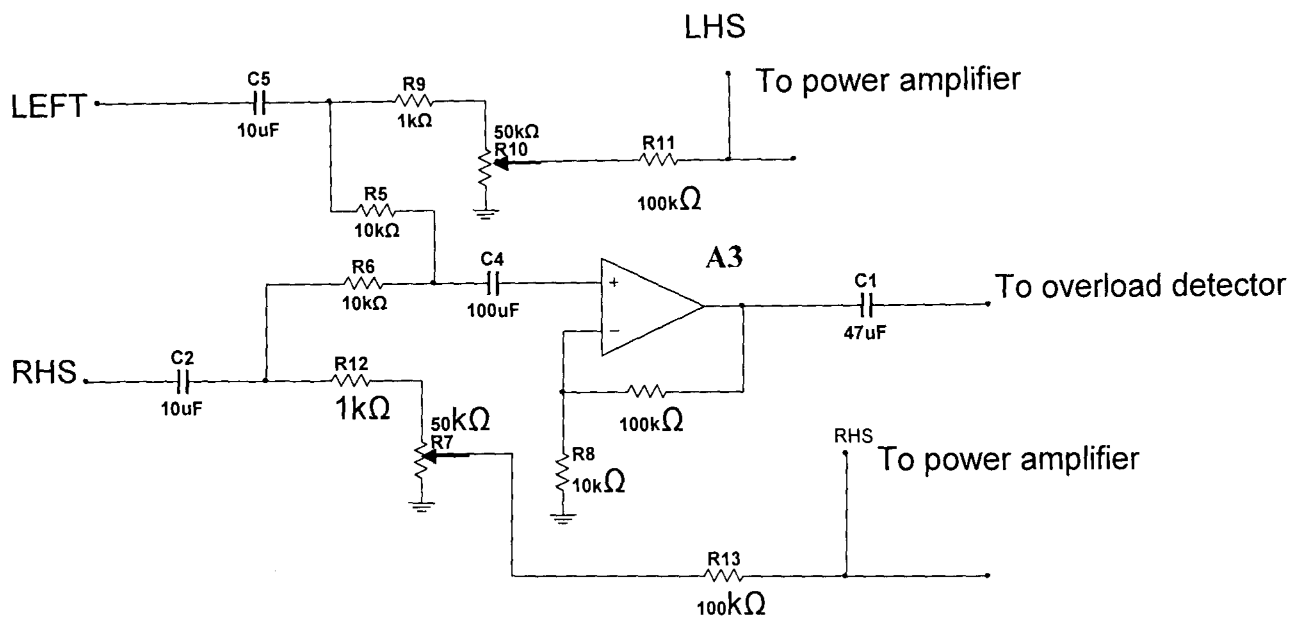

Fig. 3.7: Volume Control

And the gain of the amplifier is calculated thus:

$$\frac{V_{out}}{V_{in}} = -\frac{R_f}{R_6} = -\frac{100K\Omega}{10k\Omega} = 10 \qquad (3.13)$$

The stereo signals were summed together at the amplifier A3 and amplified by factor of 10.The amplified signal from A3 was converted to DC by an AC-to-DC converter as shown below.

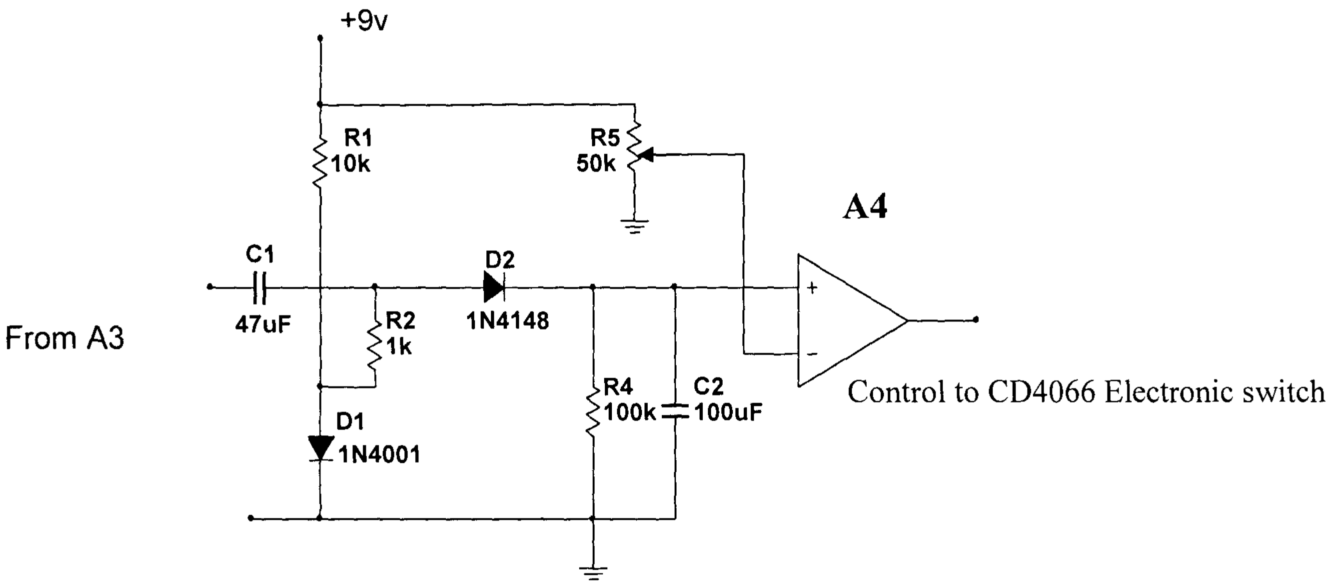

Fig.3.8: AC-DC Converter/Alternator Control Logic.

An improved AC-DC converter was used as in fig 3.8 to eliminate the voltage drop associated with a forward-biased PN junction diode, an auxiliary bias voltage was provided by R1, R2 and D2. D1 clamps the voltage at the anode of D2 at V_f. This eliminates signal loss that would occur if D2 is now biased. Audio signal arriving from A3 is fed into the AC-DC converter via 47uF capacitor. The signal voltage is converted to unipolar waveform by D2, smoothens by R4 and C2, and then compared with the reference voltage set by R5.If the voltage level in the non-inverting input (i.e. the amplifier microphone signal) exceeds the reference voltage, A4 switches its output high, otherwise it is low.R5 is adjusted to provide overload protection at the required sound level.R4 and C2 form an RC charge-discharge circuit with time constant of

$$T = R \times C \tag{3.14}$$

$$T = (100uF \times 100k\Omega) = (100 \times 10^{-6}) \times (100 \times 10^{3}) = 10 \text{ seconds} \tag{3.15}$$

Whenever V_{in} (+) exceeds Vref (reference voltage), A4 switches high for 10seconds and then low. This delay ensures that the external sound level has appreciably reduced before amplification by the hearing aid unit

3.2.5 OVERLOAD DETECTOR / ATTENUATOR

A CD 4066 electronic switch was used to effect a potential divider with the valve control as shown in fig 3.9 below.

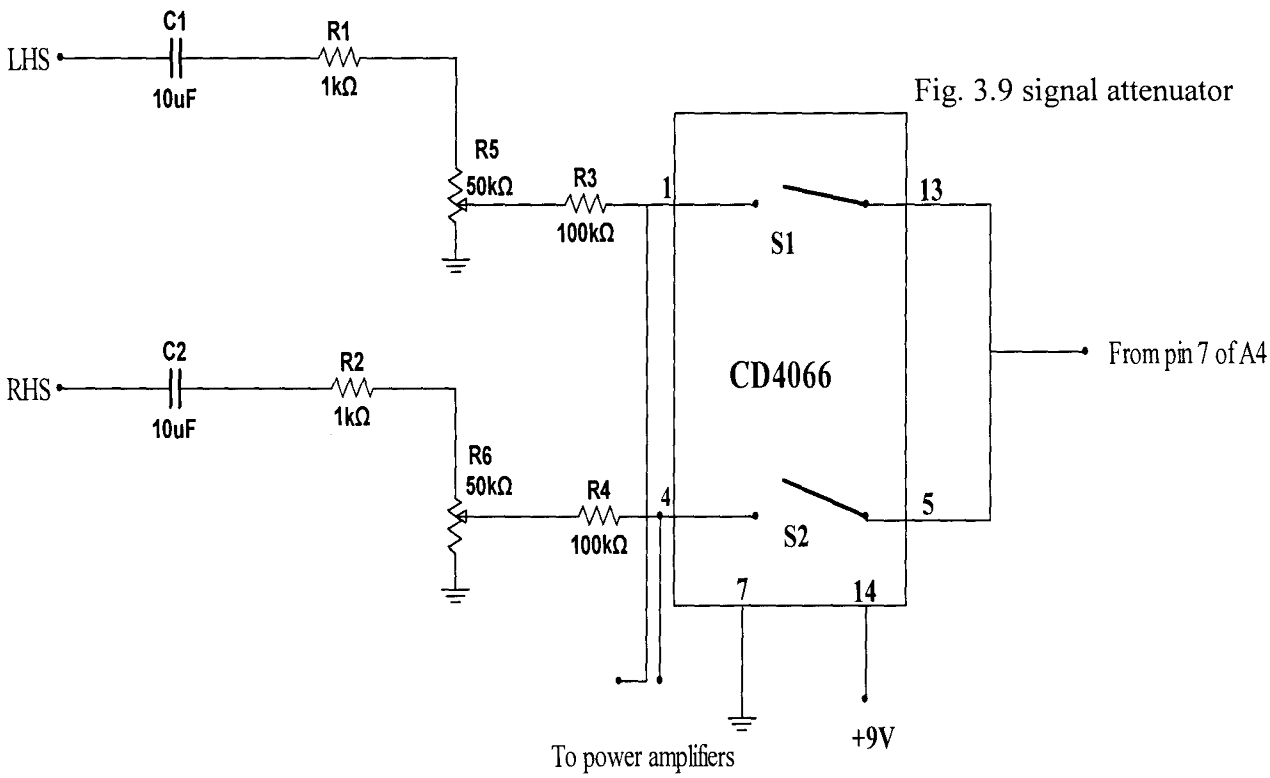

Fig 3.9: Signal Attenuator

R5 and R6 are logarithmic 50kΩ variable resistors used as the volume controls. The potential divider effect is realized by the R3 and R4 resistances, and the on resistances of S1 and S2 in the CD4066.The on resistances of S1 and S2 were measured (with their control pin active) as 13.6kΩ.The R3, R4, R5 and on RS1 and RS2 on form an equivalent potential divider show below.

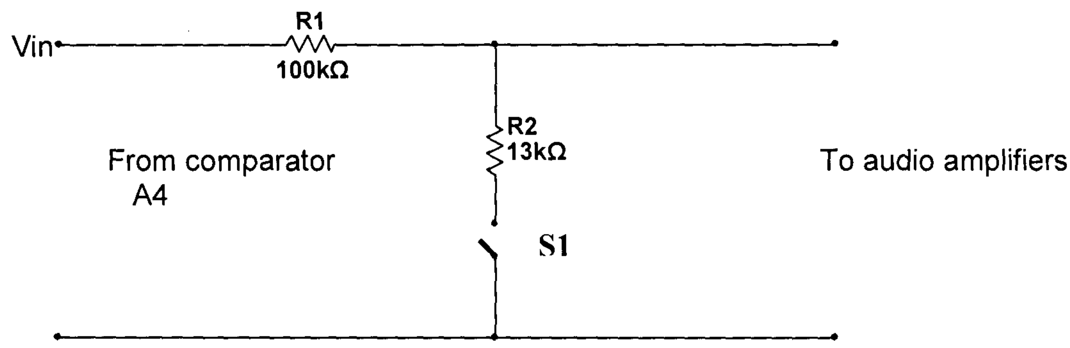

Fig 3.10: Potential Divider.

When the overload limit is not exceeded, S1 is open; the network then is a 100kΩ resistance in series with the signal path. However when the overload detector commands S1 and S2 closed, a two-resistor attenuator is affected. The attenuation factor is calculated thus:

$$V_{out} = {}^{13}/(100 + 13) \times V_{in} \tag{3.16}$$

$$^{Vo}/_{Vi} = {}^{13}/_{113} = 0.115 \tag{3.17}$$

$$Vout = 0.115\ V_{in} \tag{3.18}$$

Thus, the signal voltages presented to both power amplifiers reduces the gain factor by 0.115.

The corresponding change in power output is evaluated using the relation,

$$P_{o1} = V^2/_{R_L} \tag{3.19}$$

Where P_o is the peak power.

Therefore, the peak power with overload is calculated thus

$$P_{o1} = {}^{(0.115)^2}/_{R_L} \tag{3.20}$$

R_L is the resistance of the CD4066 when active = 13.6KΩ

$P_{o2} = (0.115V_i)^2 / 13600 = 9.72 \times 10\text{-}7V_i^2$ (3.21)

Similarly, the peak power without overload

$P_{o2} = (0.115\ V_{in})^2 / 100{,}000 = 1.32 \times 10\text{-}7\ Vi^2$ (3.22)

There, the ratio of the two power outputs i.e. without overload and at overload is given as:

$P_{o2} / P_{o1} = 9.72 \times 10^{-7}V^2 / 1.32 \times 10^{-7}$ (3.23)

$P_{o2} / P_{01} = 9.72 / 1.32 = 7.348$ (3.24)

The power ratio in Decibel is thus:

$P_{o2} / P_{01} = 10\log(7.348) = 8.7dB.$ (3.25)

3.2.6 POWER AMPLIFIER

Two monolithic audio power amplifiers were employed in the final stage of the final stage of audio reproduction. The devices were configured for a gain of about 200 by connecting a 10uF capacitor between pin 1 and 8 as in the figure 3.11. 2200uF capacitors were used to couple the amplifier to the loudspeaker. The power supply to the LM386 was bypass by the 0.1 capacitance to prevent high frequency noise appearing on the reproduced audio information.

The LM386 has the manufactures specification and is stated in figure 3.11

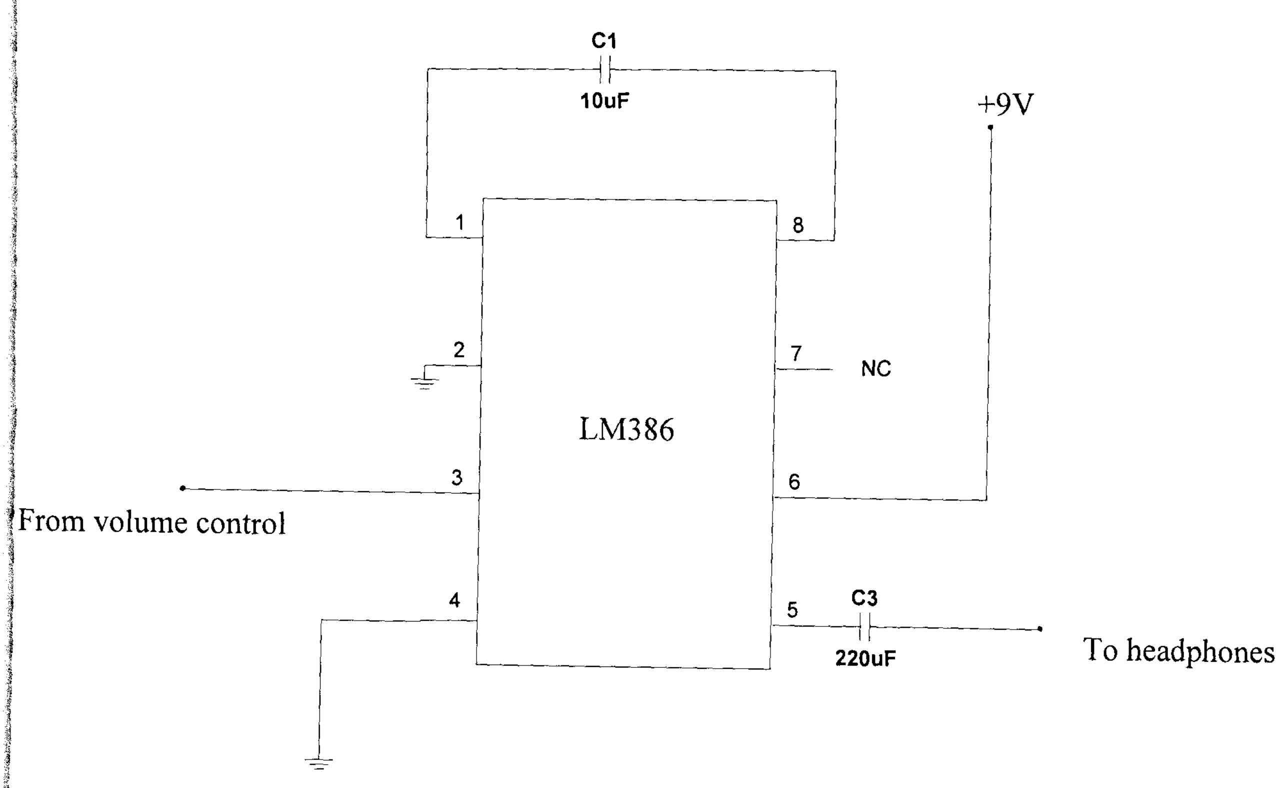

Fig 3.11: Power Amplifier

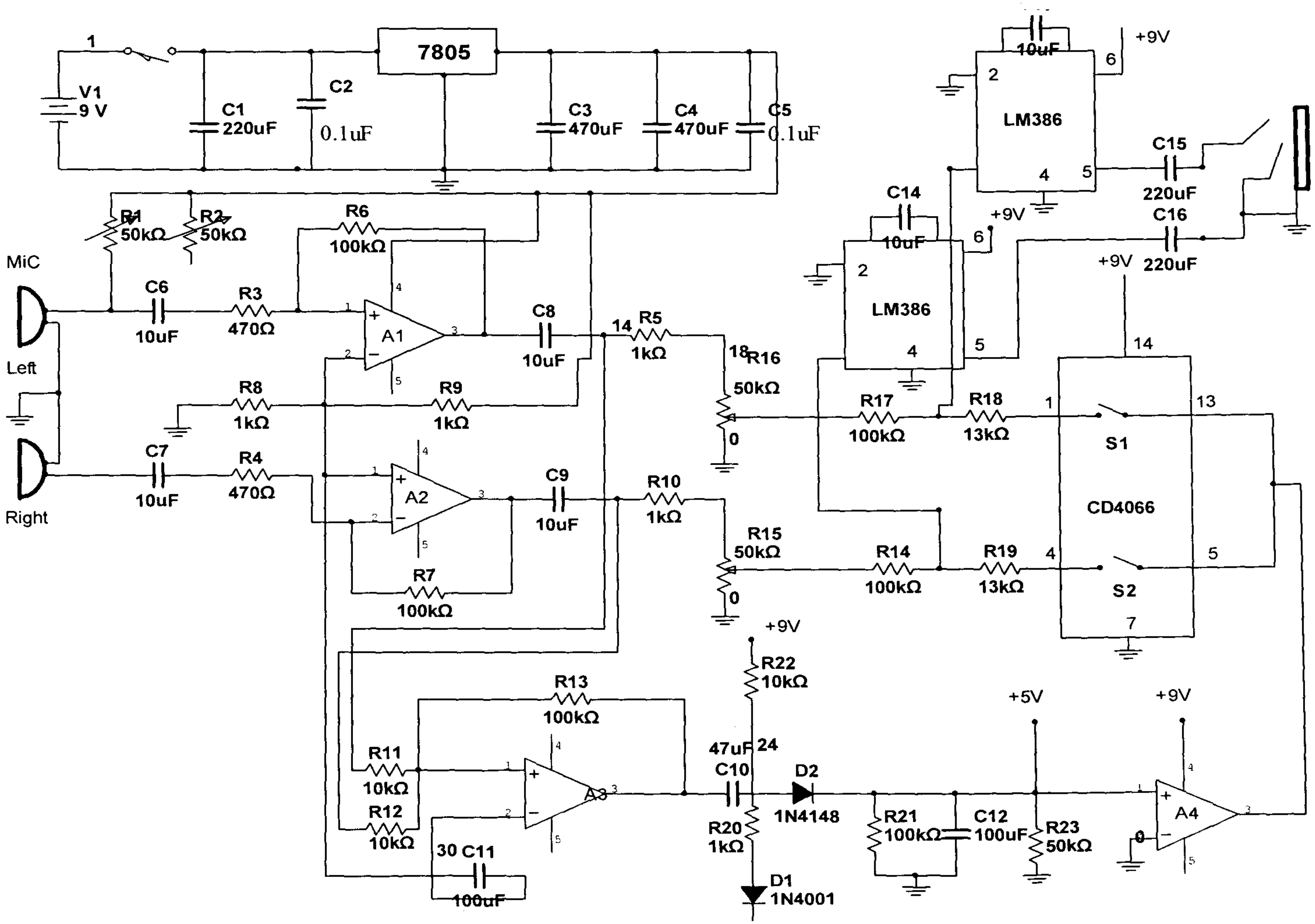

Fig 3.12: Circuit Diagram of a Stereophonic Hearing Aid

CHAPTER FOUR

TESTS, RESULTS AND DISCUSSIONS.

4.1 TESTING OF THE WORK

The following steps were taken to test the work upon completion, and these include:

The range: This is the distance between the source of the sound and the user of the hearing aid. That is the extent at which the user can hear the sound very clearly from the source. It was measured in metre (m).

The loudness: This is the extent to which the input signal is amplified at the output by the amplification unit so as to accommodate the various hearing levels of impaired user. It was measured in Decibels (dB).

The standby time: This is the length of time the battery can continue to supply power to the hearing aid continuously over a period of time. It was measured in hours (hrs).

The sonic clarity: This describes how clear the sound signal produced at the output is. It is a function of the distance from the source of signal to the user.

Suitability: This test reveals how suitable the device is. In addition, the test deals with what level of deafness the device constructed was suitable for. Unfortunately, it was not possible to get a person with a very high level of deafness to try it upon. But the test was carried out on some people with moderate hearing deficiency.

4.2 RESULT OBTAINED

Table 4.2: Test Carried Out and Results Obtained

S/N	TESTS CARRIED OUT	RESULTS OBTAINED
1	The Range	The test on the range of the hearing aid was carried out and it was discovered that it is 80 metres.But at a distance above 50 metres, the audibility begins to fall.
2	The loudness	With the help of the volume control, it was observed that the loudness of the hearing aid could be adjusted to fit various levels of hearing impairments suffered by various users.
3	The standby time	When the standby time of the device was tested, it was observed that using non-rechargeable battery; hearing aid could be powered for approximately 336hours (2 weeks).But when rechargeable battery is used longer operating time could be achieved.
4	The sonic clarity	When a test on the sonic clarity of the device was conducted, the result obtained was that, the transducers (i.e. the earphones and the microphones) were responsible for the sonic clarity of the device. That is when a high quality earphones and microphones are used, the sonic clarity improved.
5	Suitability of the stereophonic hearing aid.	When this test was conducted, it was observed that the device was suitable because it enable a normal person to hear faint sound. But because of the unavailability of severe deaf person, the test was not properly carried out. Though when carried on people with moderate deafness; it was observed that the volume could be used to adjust to fit various degrees of hearing loss.

4.3 DISCUSSION OF RESULTS

The range of the hearing aid is a very important test parameter to be considered. In this case, at distances above 50 metres, the sound signals picked by the microphones might not be amplified high enough to be heard loud enough, thus at this distance, the audibility of the device tends to fall. Therefore, people tend to raise their voices, thereby compensating for the low signal level.

Sequel to the standby time, the power supply was designed in such a way that a non-rechargeable and rechargeable battery could both be used, with the former having its advantage of easily available and affordable. While the later last much longer, and as a result can be used to power the device for a longer period of time.

The sonic clarity of the stereophonic hearing aid is dependent of the quality of the transducers (microphones and the earphones) used. A good and quality transducer gives a clear sonic quality.

The suitability of the device is of utmost importance. It is required that the hearing aid takes care of as many levels of impairments as possible. The volume control of the device takes care of that. The gain of the device was found to be 46dB which is within the range humans can hear. This also proved the suitability and efficiency of the hearing aid.

4.4 LIMITATIONS AND PROBLEMS

During the testing of this device some limitation and shortcomings were encountered.

1. At a very high volume, a loud sound is produced at the earphones and this was a result of feedback signal from the operational amplifier which generated noise at the earphones.
2. When a gain of 46dB was used, an oscillation was observed at high volumes.

3. Power failure was rampant during the period the project was carried out; as a result the progress of the work was slowed down.

4. Similarly, some of the components due to their high degree of sensitivity to temperature changes were damaged during the process of soldering, and this drastically slow down the progress of the work.

5. Inability to get a severely deaf person the testing.

6. Unavailability of some equipments in the school laboratory such as functional generator and other important equipments. And not forgetting the financial difficulties involved.

4.5 TROUBLESHOOTING TIPS

The following are the troubleshooting tips to be observed by the users when any form of malfunction arise.

Table 4.3: Trouble Shooting Tips

	Likely Faults	Possible Remedies.
1	Unit does not turn ON	Ensure that the switch is turn on. Check that the battery is properly fitted and is working fine.
2	Unit turns ON but there is no sound.	Check the volume control. Ensure that the earphone is properly plugged in.
3.	Loud whistling sound heard at the earphones.	Reduce the volume Move earphones as far as possible from the microphones on the unit.
4.	The signal at the earphone is too low.	Check the volume control. Check if the battery is too low or needs replacement.
5.	Poor sonic clarity.	Check if the battery is too low. Check the quality of the earphones.

CHAPTER FIVE

CONCLUSION AND RECOMMENDATIONS

5.1 CONCLUSION

The entire work carried out in this project involved the use of various, easily affordable, and easily sourced electronics components such as microphones,pre-amplifiers,audio amplifiers, earphones, capacitors,resistors,diodes etc.These were connected together to produce a stereophonic hearing aid, using the principle of sound amplification.

This process resulted in the design and construction of the device which could cover a range of 80 metres, a loud sound level high enough to amplified faint sound and capable of accommodating various levels of hearing disorders suffered by various individuals. And this was achieved by the use of volume control.

Similarly, the device could boast of a standby time of two weeks, but this feature is dependable on the type of battery used (that is rechargeable and non-rechargeable).A high sonic clarity was achieved by the use of high quality transducers in the form of microphones and earphones.

Most importantly, the device was found suitable for people with various forms and level of hearing disabilities.

5.2 RECOMMENDATIONS

I recommend that a more robust exposures to practical works should be given to students so as to help them carry out their final year project, and to enable them have a better understanding of the various theories taught in class, since SWEP and SIWES programmes are not adequate to achieve this work. Similarly, workshops and laboratories should be adequately furnished with modern equipments and tools so as to enable them get acquitted to the new trend in technology. Since doing this will help the students to compete with their counterparts in the labour market.

To be factual, design and construction of a stereophonic hearing aid is a project that has had much attention both far and near. I would like to recommend to anybody interested to do further research on this work as follows;

1. To embrace miniaturization technology so that the work could be made more compact and portable for users.
2. The use of wireless earpiece should also be looked into.
3. The use of rechargeable battery at the power unit should also be incorporated into the system.
4. Construction of a digital hearing aid and
5. Improvement upon the range of the hearing aid.

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APPENDIX 1

LM386
Low Voltage Audio Power Amplifier

General Description

The LM386 is a power amplifier designed for use in low voltage consumer applications. The gain is internally set to 20 to keep external part count low, but the addition of an external resistor and capacitor between pins 1 and 8 will increase the gain to any value from 20 to 200.

The inputs are ground referenced while the output automatically biases to one-half the supply voltage. The quiescent power drain is only 24 milliwatts when operating from a 6 volt supply, making the LM386 ideal for battery operation.

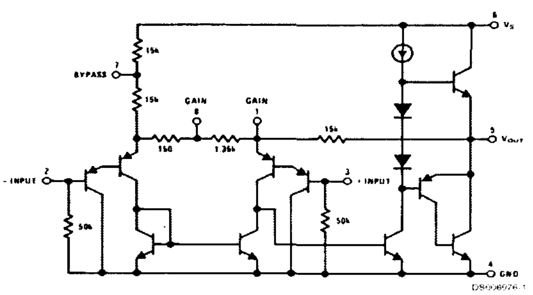
Features

- Battery operation
- Minimum external parts
- Wide supply voltage range: 4V–12V or 5V–18V
- Low quiescent current drain: 4mA
- Voltage gains from 20 to 200
- Ground referenced input
- Self-centering output quiescent voltage
- Low distortion: 0.2% (A_V = 20, V_S = 6V, R_L = 8Ω, P_O = 125mW, f = 1kHz)
- Available in 8 pin MSOP package

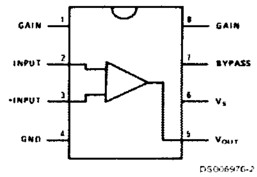
Applications

- AM-FM radio amplifiers
- Portable tape player amplifiers
- Intercoms
- TV sound systems
- Line drivers
- Ultrasonic drivers
- Small servo drivers
- Power converters

Equivalent Schematic and Connection Diagrams



Small Outline, Molded Mini Small Outline, and Dual-In-Line Packages



Top View
Order Number LM386M-1, LM386MM-1, LM386N-1, LM386N-3 or LM386N-4
See NS Package Number M08A, MUA08A or N08E

Datasheet of LM386 Power Amplifier.

intersil

CD4066BMS

December 1992

CMOS Quad Bilateral Switch

Features

- For Transmission or Multiplexing of Analog or Digital Signals
- High Voltage Types (20V Rating)
- 15V Digital or ±7.5V Peak-to-Peak Switching
- 125Ω Typical On-State Resistance for 15V Operation
- Switch On-State Resistance Matched to Within 5Ω Over 15V Signal Input Range
- On-State Resistance Flat Over Full Peak-to-Peak Signal Range
- High On/Off Output Voltage Ratio
  - 80dB Typ. at FIS = 10kHz, RL = 1kΩ
- High Degree of Linearity: <0.5% Distortion Typ. at FIS = 1kHz, VIS = 5Vp-p, VDD - VSS ≥ 10V, RL = 10kΩ
- Extremely Low Off-State Switch Leakage Resulting in Very Low Offset Current and High Effective Off-State Resistance: 10pA Typ. at VDD - VSS = 10V, T_A = +25°C
- Extremely High Control Input Impedance (Control Circuit Isolated from Signal Circuit): 10^{12}Ω Typ.
- Low Crosstalk Between Switches: -50dB Typ. at FIS = 8MHz, RL = 1kΩ
- Matched Control Input to Signal Output Capacitance: Reduces Output Signal Transients
- Frequency Response, Switch on = 40MHz (Typ.)
- 100% Tested for Quiescent Current at 20V
- 5V, 10V and 15V Parametric Ratings
- Meets All Requirements of JEDEC Tentative Standard No. 13B, "Standard Specifications for Description of "B" Series CMOS Devices"

Applications

- Analog Signal Switching/Multiplexing
  - Signal Gating
  - Squelch Control
  - Chopper
  - Modulator
  - Demodulator
  - Commutating Switch
- Digital Signal Switching/Multiplexing
- Transmission Gate Logic Implementation
- Analog to Digital & Digital to Analog Conversion
- Digital Control of Frequency, Impedance, Phase, and Analog Signal Gain

Description

CD4066BMS is a quad bilateral switch intended for the transmission or multiplexing of analog or digital signals. It is pin for pin compatible with CD4016B, but exhibits a much lower on state resistance. In addition, the on-state resistance is relatively constant over the full input signal range.

The CD4066BMS consists of four independent bilateral switches. A single control signal is required per switch. Both the p and the n device in a given switch are biased on or off simultaneously by the control signal. As shown in Figure 1, the well of the n channel device on each switch is either tied to the input when the switch is on or to VSS when the switch is off. This configuration eliminates the variation of the switch transistor threshold voltage with input signal, and thus keeps the on-state resistance low over the full operating signal range.

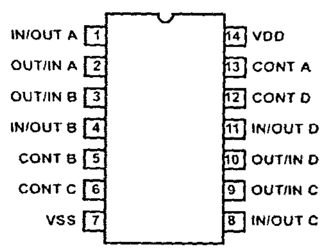
The advantages over single channel switches include peak input signal voltage swings equal to the full supply voltage, and more constant on-state impedance over the input signal range. For sample and hold applications, however, the CD4016B is recommended.

The CD4066BMS is supplied in these 14-lead outline packages:

Braze Seal DIP	H4Q
Frit Seal DIP	H1B
Ceramic Flatpack	H3W

Pinout

CD4066BMS
TOP VIEW



CD4066 CMOS Quadilateral Switch