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DESIGN, SIMULATION AND CONSTRUCTION OF SOUND ACTIVATED ELECTRONIC HEARING AID DEVICE FOR HEARING IMPAIRED PERSON

[1]A.S Gidado, [2]M. Abdulkadir, , [3]A.B Suleiman and [1]A. Musa

[1]Department of Physics, Bayero University Kano, Kano State
[2] Department of Science Laboratory Technology, Nuhu Bamalli Polytechnic, Zaria, Kaduna State
[3]Department of Physics, Federal University Dutse, Jigawa State

Abstract

A hearing aid is an electronics device comprising of transducer, filter circuit and amplifier circuit to improve the audio signal received. The design in this work involved five stages using relevant theorems and equations. The device was made to be automated using sound sensor or voice detector to trigger the operation of the Filter circuit and Amplifier circuit using Lithium battery (6-9V) d.c. powering. The designed circuit was simulated using Multisim software to determine the A.C signal analysis, transient analysis, noise analysis and distortion analysis which are relevant for successful design. In the AC Analysis, the output amplifier exhibits exponential decrease in magnitude at higher frequency such as 10,000Hz. For transient analysis, the pre amplifier stage was found to be stable at the magnitude of $2.353mV$ per millisecond. The noise analysis shows that the audio amplifier has lower noise at higher frequency. The circuit has an input impedance of 20.7KΩ and an output impedance of 159Ω which make it to have low power consumption. The circuit was constructed based on the designed parameters and specifications. The circuit was found to operate automatically within the designed bandwidth of 100Hz to 20KHz as well, was found to improve the hearing capability of temporarily impeded person but cannot restore the normal decibel hearing level of the person.

1.0 Introduction

Hearing aid is an electronic device that can amplify sound wave in order to help a deaf or earring impeded person hear sound more clearly. It filtered or reduces the signal to noise ratio as well as increase the amplitude of the output signal for higher quality or fidelity [1]. Hearing loss is a prevalent major public health problem and is one of the most common sensory disorders that causes disability. The condition is associated with poor quality of life, with negative effect on interpersonal relationship, communication, socialization. Hearing loss is a major cause of disability with prevalence of over 250 million people suffering various degrees of hearing loss globally with 75% living in sub-Saharan African out of which about 2.8% resides in Nigeria which translate to approximately seven million Nigeria, that is 2.8% of 160 million of total population suffers from one hearing disorder or another [2].

People aged 15 years or more with mild hearing loss cannot detect tones at an average level below 25 to less than 45dB HL (decibel hearing level) while those with moderate loss cannot detect tones at an average of 45 to less than 65dB HL and those with severe hearing loss can not detect tones at 65 or more dB HL in their better hearing ear [1]. Properly designed and fitted hearing aids can improve communication in at least 90% of people with hearing loss as well as improving adult's health related quality of life. While some people are born with hearing problem some others develop it as they grow. This problem can occur as a result of disease, aging, injury from noise or intake of certain medicines. Hearing problems could be that of complete deafness or partially impaired type. Hearing problem could occur after a person learned to talk (post lingual) or those with trilingual deafness that is deafness that occurs before a person learns to talk. Deafness, whatever the degree or course, is generally a source of worry and frustration to the patient concerned as it affects almost all aspects of one's life [3].

Various efforts have been made and still being made in attempt to overcome this ailment both medically and using technology. Measuring devices are now available to enable otolaryngologist's measure aspects of a patient's hearing

Corresponding Author: Gidado A.S., Email: asgidado.phy@buk.edu.ng, Tel: +2348036407447

sensitivity and prescribe remedy. Electronically, hearing aids of various grades, sophistication, sizes and costs have been developed. Hearing aids have the same basic components as any public-address system, but all the components are miniature and the amplified sound is delivered to the ear of only the hearing-aid user. The microphone, amplifier (consisting of transistors and integrated electronic circuits), miniature receiver, and battery of a hearing aid are enclosed either in a chassis or shell, which is worn behind or within the ear or in the stem or temple portion of eyeglasses. A small tube directs the amplified sound from the receiver into the ear canal of the wearer [3].

Giovanni Batista Porta was the first to actually describe one of those early hearing aids. Porta wrote a book entitled "Natural magic in which the wooden aids shape animal ears" in 1627. These hearing aid devices were probably not manufactured in the way we know it today.. In the 17^{th} century, speaking tubes were adopted to a very special sort of hearing problem by puritan couples who were counting. Custom of the times required the two to sit across table from each other, and speaking tubes were used to ensure the privacy of their conservation. Later, anatomical, used to slightly enlarge the sound collection area of the ear may also have been worn by person suffering from collapse of entrance to the external auditory canal. The phonograph was invented by Thomas Alvan in 1832 [4]. [5] designed hearing aid to be automatically calibrated by an Android device was developed to analyze the hearing loss of a person. The calibrated headphones was used for the purpose of analyzing the hearing loss of a person. The Smart hearing aid consists of two modules. The first module analyses the hearing loss and the second module is a hearing aid which performs automatic audio calibration based on the output of the first module. A physically challenged person can undergo hearing loss analysis in their own comfort rather than travelling to hearing test labs which in turn reduces the stress on them and further more reduces the cost. Communication between the Android device and the hearing aid is provided by using GSM (Global system for mobile) technology.

In 1886, Edisobn applied for a pattern on his carbon microphone or transmitter, which translate sound into electrical signals, allowing it travel through wires and then been translated back into sound, this mark the beginning of first electric signal amplifier. In 1899 Miller Reese and J. Wilson established the evaluation company in Alabama. They held the pattern for the first practical hearing aid which employed a carbon microphone or transmitter, a battery and pair of earphones. The invention of vacuum tube marked the beginning of electronic hearing aid device in which the first one appeared in 1922, but this type did not become practical until 1936. In 1952, integrated circuit (I.C) hearing aids popularly called electronic hearing aid appeared in few models and virtually replaced vacuum tube hearing aid by the end of 1953 [6].

According to [6] "Hearing aid device is a small electronic gadget that is fit in or behind the ear to improve one's hearing and consequently communication ability". They research work involves the designed and developed a hearing aid device with pre-amplifier, an acoustic signal picked-up using a condenser microphone. TDA 2822M IC was configured to produce an audio amplification which was converted to audio signal through a headphone. Domtau, et-al, 2013 also designed and constructed a low-cost hearing aid devic and constructed to produce an amplified sound for people with hearing loss. A 9V dc was used as the power supply [7]. The condenser microphone was used as input transducer to pick up sound from the environment for conversion to electrical signal, NPN transistor (BC548C) along with three capacitors and five resistors were used as pre-amplifier. The integrated circuit (IC) TDA2822M, available in 8-pin mini chip package and specially designed for portable power amplification was used for the amplification function. A 32 ohms ear phone was used as the output transducer to convert the amplified electrical signals back to sound [6].

In this work the hearing aid device was designed to operate automatically within the designed frequency bandwidth of 100Hz and 20000Hz. TDA 2030 IC was used as output audio amplifier active component due to its lower current consumption as well less power dissipation.

2.0 THEORETICAL BACKGROUND

2.1 FILTER

In almost all sensor systems there's a need to use or design filters selectively to admit or reject signals in order to reduce the bandwidth to the required operating region or to reduce noise as well to reject interference by selecting one signal at a certain frequency from other signals at different frequencies [8].

2.1.1 First-Order Filters

The first-order filter is the simplest type and forms the basis of all other filters. Normally, is called the *Butterworth* type. In this circuit note that the op-amp is ideal, i.e., it draws no current, and also it is used in the non inverting mode in order to prevent loading down of the *RC* network. *R* and *C* act as a voltage-dividing network, and hence we [8].

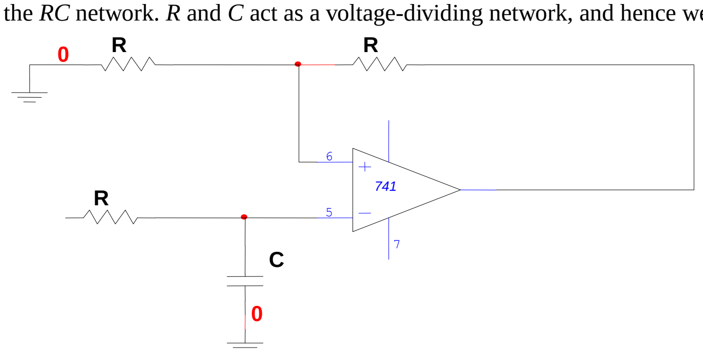

Fig. 1: Low pass Buteworth filter circuit

The filter function for Butterworth transfer function equation is given by [9]

$$\frac{V_o}{V_i} = \frac{A}{1+j\left(\frac{f}{F_{L(3dB)}}\right)} \quad (1)$$

Where

$$F_{L(3dB)} = \frac{1}{2\pi RC} \quad (2)$$

and $A = 1 + \frac{R_f}{R_i}$ (d.c gain of a non-inverting OPAMP) and F is the cutoff frequency.

This has the characteristics of a first-order low-pass filter. When $\omega = 0$ then the passband gain is:

$$\frac{V_o}{V_i} = \frac{R_f}{R_1} = K \quad (3)$$

This is simply the amplifier gain.

2 . 1 . 2 Second-Order Filters

As has already been mentioned, the higher the order of filter the sharper the cut-off. For certain applications, such as radio relay applications and channel separation, it is necessary to have higher-order filters. This work considered only first and second-order filters but many higher orders can be designed by simply cascading these two types. This is one of the big advantages of using the active filter [8].

2.1.3 Low-Pass Second-Order Filters

Consider two low-pass first-order filters with the same cut-off frequencies, but different pass-band gains are represented by their transfer functions as [9].

$$\frac{V_o}{V_i} = \frac{K_1}{1+j\left(\frac{f}{f3dB}\right)} \quad \text{(low pass filter transfer function with gain K)} \quad (4)$$

$$\frac{V_o}{V_i} = \frac{1+K_2}{1+j\left(\frac{f}{f3dB}\right)} \quad \text{(low pass filter transfer function with gain } K_1K_2 \text{)} \quad (5)$$

If these filters are now cascaded, then the overall function becomes

$$\frac{K_1K_2}{(1+j\sigma)^2} = \frac{K}{(1+j\sigma)^2} \quad (6)$$

where $\sigma = 1/f_{3dB}$ and $K = K_1K_2$.

Expanding the above expression gives:

$$\frac{V_o}{V_i} = \frac{K}{a^2(j\omega)^2 + 2a(j\omega) + 1}$$

and in general terms the transfer function become

$$\frac{V_o}{V_i} = \frac{K}{a_2(j\omega)^2 + a_1(j\omega) + 1} \quad (7)$$

2.1.4 Audio Amplifier

Class A stages are those in which the transistor(s) are always biased on and never saturated (bottomed). A Class A stage may use a single transistor (a single-ended stage) or two transistors which share the current in some way (a push-pull stage), but the efficiency is low. Percentage efficiency is defined as [10].

$$E = \frac{Power\ dissipated\ in\ the\ load}{Total\ power\ dissipated\ in\ the\ output\ stage} \times 100 \quad (8)$$

and is always less than 50% for Class A operation A Class A stage should pass the same current when no signal is applied as when maximum signal is applied.

Class B audio operation uses a pair of transistors biased so that one conducts on one half of the waveform and the other on the remaining half. Some bias must be applied to avoid 'crossover distortion' due to the range of base- emitter voltage for which neither transistor would conduct in the absence of bias. Class B audio stages can have efficiency Figures as high as 75%, though at the expense of rather higher distortion than with a Class A stage using the same layout. The higher efficiency enables greater output power to be obtained with smaller heat sinks, and the use of negative feedbacks [11]. Class D amplification principle is to use fast-switching transistors with pulse waveforms that have been modulated with an audio signal, and one enormous advantage is that the dissipation in the transistors can be low even for outputs of several hundred watts. This allows higher output power for an IC chip to become a reality, particularly when fast-switching MOSFETs with very low forward resistance can be used [11].

3.0 DESIGN METHODOLOGY

The work involved both experimental and simulation methods. Relevant design equations and component data specification were used in the circuit design. The simulation of the circuit was carried out using Multism 10. The simulated circuit was then constructed and tested. Details of the procedure on the design, simulation and construction of the circuit are as follows.

3.1 DESIGN PROCEDURE

These involved the following stages according to the block diagram in fig 2.

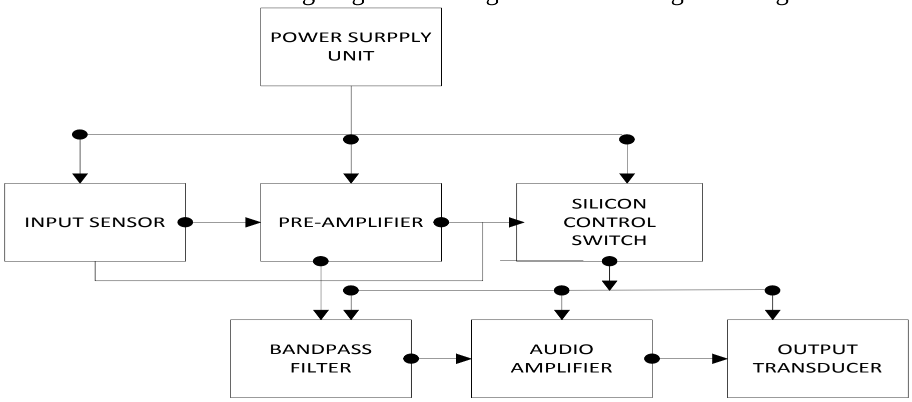

Fig. 2: Block diagram for the design

Stage 1: Sensor / Input Transducer Design

From the relation [12].

$$V_{CC} = i_{dc} R_T \quad (9)$$

R_T is the effective series resistance of microphone and is given by

$R_T = R_1 + microphone\ DC\ resistance(R_{mic})$

From the Best sound electronics [13] datasheet information on Ominidirectional electrets condenser microphone electrical characteristics, we have:

Output impedance: 1.1KΩ to 2.2KΩ

Current consumption for which V_{cc} = 2.0V, R_L = 2.2KΩ is 300µA

Operating voltage 1.5 to 10V [13].

V_{CC} = 9V, I_{dc} = 300µA, $V_{CC} = i_{dc} R_T$.

9V = 300µA (R_1 + 1.1KΩ)

$9V = 300 \times 10^{-6} R_1 + 300 \times 10^{-6} \times 1100$

$$R_1 = \frac{9 - 0.33}{3 \times 10^{-4}} = 28900\Omega = 29\text{K}\Omega$$

The coupling capacitor C_n should have a reactance X_c of few kilo-ohms at the mains audible frequency of low pass value of about 1KHz. In order to have good impedance matching we choose *Xc = 2.2K*Ω which is relatively equal to the output impedance of the microphone [14].

Hence

$$X_c = \frac{1}{\omega C} = \frac{1}{2\pi fc} \quad (10)$$

F = 1000Hz , C = ? $X_c = 2.2\text{K}\Omega$

$$2200 = \frac{1}{2\pi \times 1000 \times C_1}$$

$C_1 = 7.23 \times 10^{-8} F \approx o.1\,\mu F$

Summary: R_1 = 29KΩ, $C_1 = o.1\,\mu F$

Stage 2: Pre-amplifier design

From the data sheet for BC 547 to 550 NPN epitaxial Silicon transistor to operate at 25^0C

DC current gain (h_{fe}) at V_{CE} = 5V, I_C = 4mA ranges from 110 to 800. Taking h_{fe} = 200

$V_{CE(sat)}$: 250 to 600mV $V_{BE(sat)}$: 900mV = 0.9V $V_{BE(on)}$: 720mV = 0.7V

Output capacitance : 3.5 to 6.0pF Input capacitance : 9.0pF $I_{CBO(collector\ cut\text{-}off\ current)}$: 15nA

$I_{C(collector\ current)}$: 100mA to 200mA $P_{C(collector\ power\ dissipation)}$: 500mW

$I_{BM(peak\ base\ current)}$: $200\mu A$ [15].

where β ,α, r_e , r_o , I_c , I_B , V_{BE} are obtainable from manufacturer's data sheet

$V_c = V_{out} = \frac{1}{2} V_{cc}$, if V_{CC} = 9V then V_C = 4.5V

The V_{out} can be obtained from [12].

$$V_{out} = V_{cc} - I_c R_c \quad (11)$$

if V_{out} = 4.5V and $I_c = 4mA$,

$4.5 = 9 - 4 \times 10^{-3} R_C$

$R_C = \frac{4.5}{4\times10^{-3}} = 750\,\Omega$

Since $I_C \approx I_B$ (12)

$V_{BB} = \frac{V_{cc}\ R_c}{R_1 + R_2}$,

$V_{CC} = I_E R_E + V_{CE} + I_C R_C$ (13)

$9 = 4\times10^{-3}\times750 + 4.5 + 4\times10^{-3} R_E$

$R_E = 166\Omega$ for which 100Ω will be conveniently available

Alternatively $R_E \approx \frac{R_C}{10}$ hence $R_E = 75\ \Omega$ (for which 33 Ω was used for effective feedback)

Transistor current relation and gain factor relation are given as [16].

$I_E = I_C + I_B$, (14)

$\beta = \frac{I_C}{I_B}$, (15)

$I_B = \frac{4\times10^{-3}}{200} = 2\times10^{-5}A = 20\mu\text{A}$

But

$I_1 = 10 I_B$ (16)

$I_1 = 200\mu\text{A}$

$I_1 = \frac{V_{cc}}{R_1 + R_2}$ (17)

$200\mu\text{A} = \frac{9}{R_1 + R_2}$

$R_1 + R_2 = \frac{9}{200\times10^{-6}} = 45{,}000\ \Omega = 45\text{K}\ \Omega$

$V_2 = 0.7 + 0.1$

$V_2 = 0.8$V at saturation

$R_2 = \frac{V_2}{I_1} = \frac{0.8}{100\times10^{-6}} = 8000\ \Omega$

R_2 was chosen to be 10KΩ (for voltage divider biasing purpose)

$R_1 + R_2 = 45000\ \Omega$

$R_1 = 45000 - 8000$

$R_1 = 37000\ \Omega$ for which 330KΩ was chosen due to tolerance range (for fixed biasing purpose)

The circuit coupling impedance is calculated as [17].

$Z_{i(min.)} = R_1 /\!/ R_2 /\!/ \beta r_e$ or $Z_i = \beta r_e$ (C-E configuration) $Z_{o(max)} = R_c /\!/ r_o \cong R_c = Z_o$

$r_o = 10 R_c$, $r_e = \frac{26mA}{I_E}$

$r_e = \frac{26\times10^{-3}}{4\times10^{-3}} = 6.5\Omega$

$Z_i = \beta r_e = 200 \times 6.5 = 1300\Omega$

$r_o = 10\ X\ 680 = 6800\ \Omega$

Summary: $R_1 = 33000\Omega$, $R_2 = R_3 = 22\ K\Omega$, $R_C = R_2 = 750\Omega$, $R_E = R_5 = 75\ \Omega$, $C_2 = 10\mu\text{F}$

Stage 3: Switching design

For the silicon control switch (SCS) to operate effectively a typical maximum anode current ranges from *100mA* to *300mA* with dissipating power rating of *100* to *500mW* and Gate turn on/off (GTO) current ranges from *30μA* to *20mA* are to be used [18].

Similarly the forward -on-state voltage typical of [19]

1.4V, if $I_A = 50$mA, $I_{AG} = 0$, $R_{KG\text{-}K} = 10$KΩ and current $I_H = 1$mA were used.

$V_{BB} = 2$V, Turn-on-time $t_{on} = 0.25\mu$s, for which $V_{KG\text{-}K} = -0.5$ to 4.5V, $R_{KG\text{-}K} = 1$ KΩ

$V_{CC} = I_{(GTO)} R_T$ (18)

where $R_T =$ Biasing resistance + microphone DC resistance

Also using the relation [16]

$V_{CC} = 20\times10^{-3}(R_{mic} + R)$ R_{mic} *is known,* R = ?

$9 = 1.4 + 20\times10^{-3} R$ $R = \frac{7.6}{20\times10^{-3}} = 380\ \Omega$

Summary: R = R_6 = 380Ω for which 470Ω is appropriate

Stage 4: Indicator design

LED (light emitting diode) maximum current and voltage are given by [12].

$I_{Dmax.} = I_F = 20$mA, Pd across LED = 1.5 to 2V

$V_{CC} = I_D R + V_R$ (19)

There for R = ?

$9 = 2 + I_D R$,

$R = \frac{7}{0.02} = 350\Omega$

Summary $R = R_D = 350\ \Omega$

Stage 4: Filter design

For upper cut off frequency of 20KHz and choosing R = 10KΩ , then from [12].

$$F_{OH} = \frac{1}{2\pi R_{12} C_4} \quad (20)$$

$20 \times 10^3 = \frac{1}{2\times3.142\times10\times10^3\times C_4}$, $C_4 = \frac{1}{20\times10^3\times6.284\times1010^3} = 7.9867 \times 10^{-10}F$

$C_4 = 7.986 \times 10^{-4}\mu F \approx 0.799pF$

For low pass filter [20]:

F = 200Hz , choosing R=10KΩ $F_{OL} = \frac{1}{2\pi R_7 C_3}$

$200 = \frac{1}{2\times3.142\times10\times10^3\times C_3}$, $C_3 = \frac{1}{200\times6.284\times1010^3} = 7.9867 \times 10^{-8}F$

$C_3 = 0.07956\mu F \approx 0.1\mu F$

The voltage gain for filters is obtained from

$$V_0 = (1 + \frac{R_f}{R_i})V_i \rightarrow A_V = 1 + \frac{R_f}{R_i} \quad (21)$$

The filter design require gain in decibel (dB) for which 65dB was chosen (moderate hearing loss) [9].

$$65dB = 20\log(1 + \frac{R_f}{R_i}) \quad (22)$$

$65 = 20log1 + 20log\frac{R_f}{10}$

$45 = 20logR_f$

$R_f = 177.83\ K\Omega$

Summary: $R_7 = R_{10} = 10K\Omega$, $R_f = R_8 = R_{11} = 200\ K\Omega$, $R_9 = R_{12} = 10\ K\Omega$, $C_3 = 0.1\mu F$, $C_4 = 0.1\ \mu F$

Input Impedance

From the relation of input impedance due to parallel resistors R_T and R_1 [20].

$$Z_i = R_T // R_1 \quad (23)$$

= 22kΩ // 330KΩ = 20.7KΩ

Output impedance

The output impedance due to the effective resistance and capacitive reactance is given as;

$$Z_o = \sqrt{R^2 + X_c{}^2} \quad (24)$$

where $X_c = \frac{1}{\omega C} = \frac{1}{2\pi f c}$,

F = 100Hz , C = 10nF

$X_c = \frac{1}{2\pi\times100\times10\times10^{-6}} = 159.13\mathbf{\Omega}$

Given in data sheet R = 4.7 (output fuse resistor)

$Z_o = \sqrt{(4.7)^2 + (159)^2} = \sqrt{25303.09} = 159\Omega$

Summary $Z_i = 20.7K\Omega$, $Z_o = 159\ \mathbf{\Omega}$

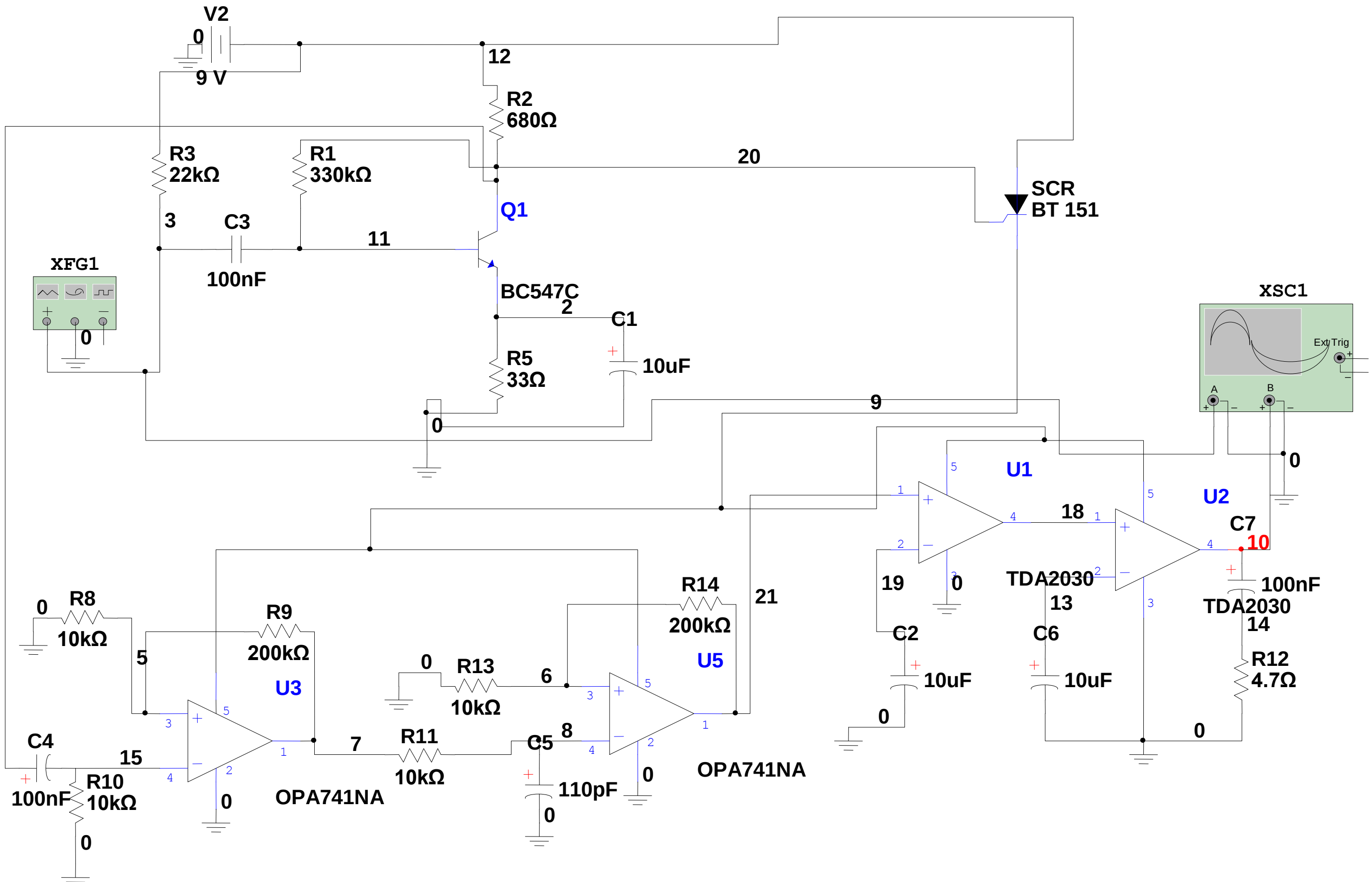

Fig. 3: Sound Activated Electronic Hearing Aid Device Circuit Diagram

4.2 SIMULATION

The Simulation employed the use of Multism 10 software to analyze the operation of designed sub-circuits. The DC Analysis was used to determine the DC operating point of the circuit**.** AC Analysis AC analysis is used to calculate the frequency response of linear circuit, with all input source considered to be sinusoidal. This analysis calculated the AC circuit response as a function of frequency. The Transient Analysis also called Time-domain computes the circuit response as a function of time. The solution of the voltage waveform at a node is determined by the value of that voltage at each time point over one complete cycle [21]. Noise Analysis indicate the noise contribution from each resistors and semiconductor device at the specific mode. Thermal Noise is temperature dependent and caused by the thermal interaction between free electrons and vibration ions in a conductor expressed by Johnson's formula as [22].

$P \ = K \times T \times BW$

where K = Boltzman's constant ($1.38 \times 10^{-23} J/K$), T = resistor temperature (T = 273 + operating temperature) BW = bandwidth. The Fourier Analysis is a method of analyzing complex periodic waveform. It permits any non-sinusoidal period function to be resolved into sine or cosine waves. This permits further analysis and allows to determine the effect of combining the waveform with other signals [24].

4.0 RESULTS AND DISCUSSION

The sound transducer (condenser microphone) converts the sound pressure into electrical signal at low amplitude, this audio signal is then amplify by the common emitter configuration (single collector bias) pre-amplifier. The pre-amplified signal is fed to the bandpass-filter which allow frequency range of 100 to 20000Hz to pass into the input of the main audio-amplifier. The input signal in the bandpass and audio amplifier stage remain dormant until the Thyristor (switching circuit) simultaneously receive a triggering signal from the input transducer in order for the circuit to complete operation. The process is repeated for every signal received by the input transducer.

4.1 AC ANALYSIS

At the input frequency of 100 – 10000Hz , the Pre-amplifier displayed peak amplitude at about 1000Hz (indicating high gain at that frequency) and no phase angle change at the same frequency as in Fig 4. The Bandpass active filter shown in Fig.5 under the same condition shows increase in magnitude as the frequency increases (due to gain ratio) with decrease in phase angle as frequency increases but much stable at about 1000Hz.The audio output amplifier Fig.6 exhibits exponential decrease in magnitude at much higher frequency.

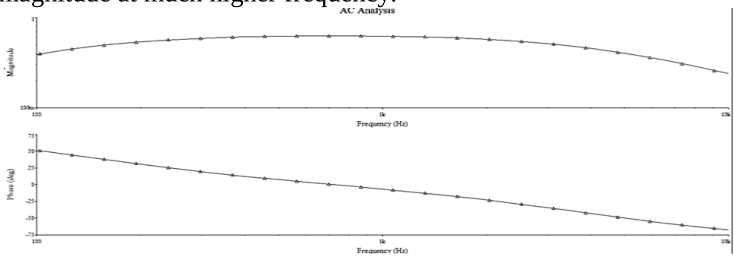

Fig.4: Graph of AC analysis for pre-amplifier

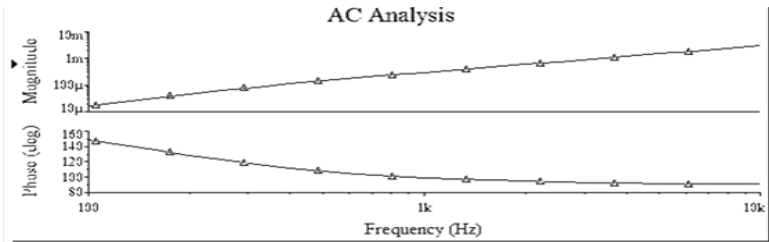

Fig. 5: Graph of AC analysis for bandpass filter

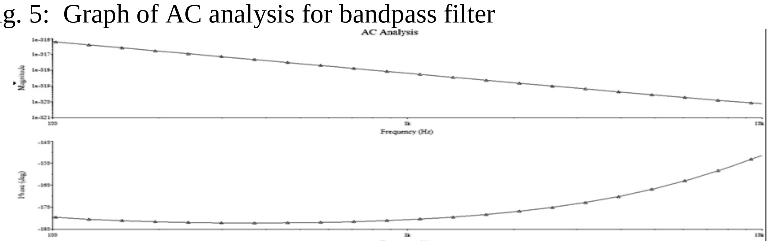

Fig. 6: Graph of AC analysis for Audio amplifier

4.2 TRANSIENT ANALYSIS

Transient analysis computes the circuit response as a function of time by determining the value of voltage at each time over one cycle. The pre-amplifier graph shown in Fig. 7 is found to be stable at the magnitude of 2.352mV per millisecond. While for the bandpass active filter Fig. 8; the output voltage increases linearly (directly proportional to increase in time). The output audio amplifier in Fig.9. has a close to zero response time because the TDA 2030 IC has a faster response time (nanosecond).

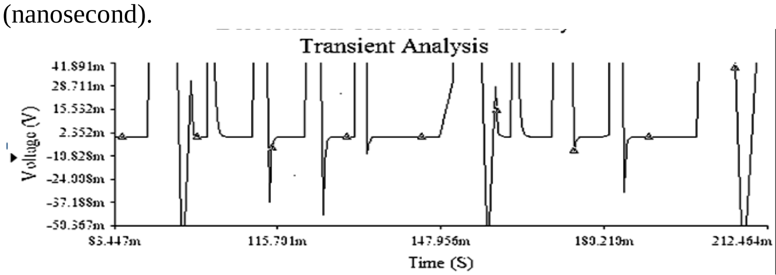

Fig. 7: Graph of Transient analysis for pre-amplifier

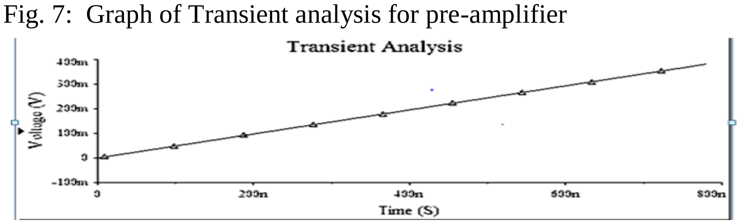

Fig. 8: Graph of Transient analysis for band pass active filter

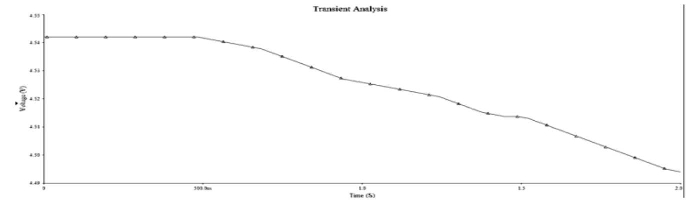

Fig. 9: Graph of Transient analysis for output audio amplifier

4.3 NOISE ANALYSIS

Noise analysis indicates the noise contribution from each resistors and semi-conductor at the specified node. At the pre-amplifier stage Fig.10 it indicates good operation with less noise interference between the frequency range of 1000Hz and 10,000Hz. The bandpass active filter Fig.11 have nearly constant noise effect at the operating frequency. The audio amplifier graph shown fig.18 exhibits lower noise at higher frequency.

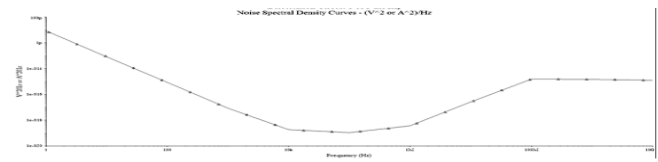

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Fig. 10: Graph of Noise analysis for Pre-amplifier

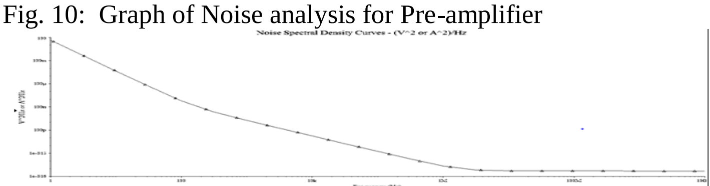

Fig.11: Graph of Noise analysis for Bandpass filter

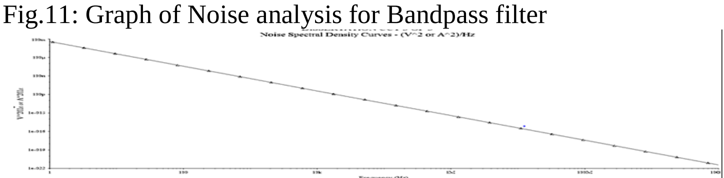

Fig.12: Graph of Noise analysis for Audio amplifier

4.4 DISTORTION ANALYSIS

Distortion analysis is the result of gain non-linearity or phase non-uniformity in the circuit. The pre-amplifier in Fig. 13 indicates no distortion in both magnitude and phase angle because the output signal is the replica of the input signal. The same was observed for bandpass active filter and output audio amplifier.

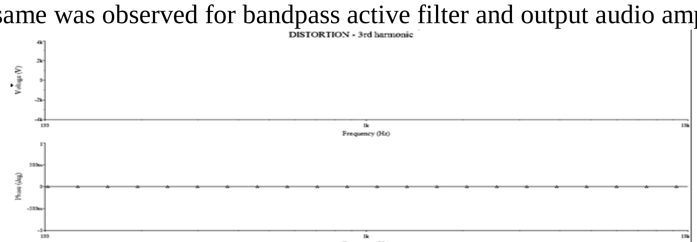

Fig.13: Graph of Distortion analysis for pre-amplifier

4.5 FOURIER ANALYSIS

In Figures. 14 & 15, there are low harmonic distortion because the output signal is a replica of the input making it pure sinusoidal while Fig. 15. Shows high harmonic distortion due to the band sampling hence making it appear more like digital signal

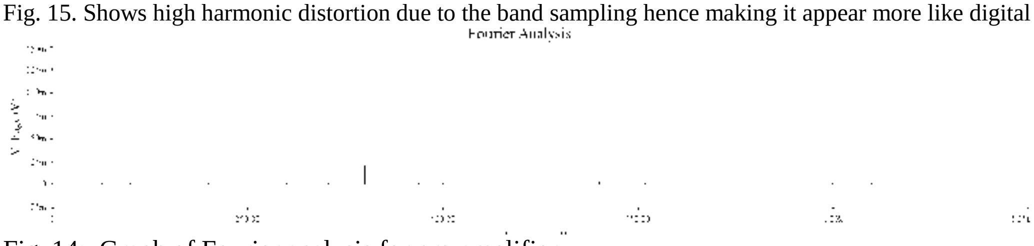

Fig. 14: Graph of Fourier analysis for pre-amplifier

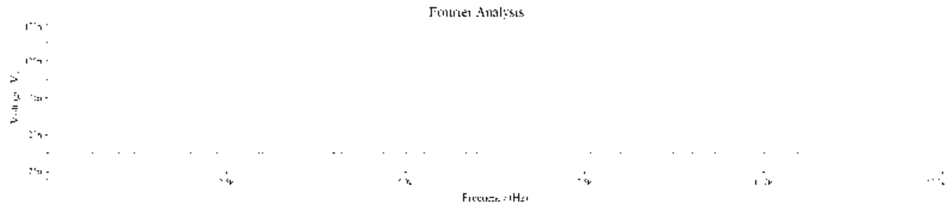

Fig. 15: Graph of Fourier analysis for bandpass filter

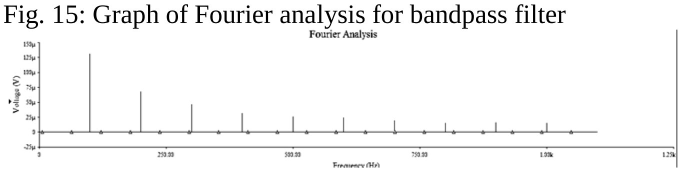

Fig. 16: Graph of Fourier analysis for output audio amplifier

5.0 CONCLUSION

On successful design using appropriate mathematical equations and theorems, the electronic hearing aid device operates automatically by detecting audio signal between the range of 100Hz to 20KHz. The circuit simulation results show that ; there is low signal distortion, low noise, low power consumption, low heat dissipation and the circuit has high fidelity and stability. It has a relatively moderate input impedance of 20.7KΩ which prevent overloading effect and its output impedance of 159Ω making it low heat and power dissipation. Hence this will improve the health, economical and personal confidence of the hearing impaired person. The constructed circuit was tested and found to be working.

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