HARDWARE SIMULATION OF SPEECH PROCESSOR FOR COCHLEAR IMPLANT

Meenakshi Rajendra Rane[1], Prof. Sangeetha Prasanna Ram[2]

[1]Student, [2]Deputy H.O.D, Department of Instrumentation Engineering, Vivekanand Education Society's Institute of Technology,meenakshirane31@gmail.com

Abstract— In this paper, we are proposing system which will be used for humans who suffer from hearing impairment. Sensorineural hearing loss cannot be corrected with medicines. Hearing aids help the people with conductive deafness which is outer ear deafness by amplifying the sound. The cochlear implant uses its own electrical signals to stimulate the auditory nerve, by completely bypassing the damaged part of the cochlea, allowing the person to hear. Cochlear implant is different from hearing aid because hearing aid system deals with the people who have partial hearing loss and cochlear implant concerns for people whose auditory sensors (hearing cells in the cochlea) are not functional at all.

Hardware of the speech processor is described in detail. In the proposed system, sound is picked up by the microphone and is amplified using microphone amplifier circuit and given to filter which will pass sound in the range of 1kHz-6 kHz and that signal is simultaneously given to three band pass filters for division into sub-bands and then to three LEDs through threshold detectors.

Keywords — Cochlear implant, Cochlea, Speech processor, Sensorineural hearing loss, profoundly deaf, bionic ear, filter.

1. INTRODUCTION

Since speech is man's most important form of communication, all efforts must be done to make speech communication possible. The commonly stated range of human hearing is 20 Hz to 20 kHz. 300 Hz to 3000 Hz is referred to as voice frequency. Ear's sensitivity is best at frequencies between 1 kHz to 5 kHz. There is little energy in the spectrum above 4 kHz in voiced sounds and little energy in the spectrum below 1000 Hz in unvoiced sounds.

According to the Food and Drug Administration (FDA), as of December 2012, approximately 324,200 people worldwide have received implants. Cochlear implants can be provided for children as young as 12 months old, as well as adults [1].

A cochlear implant is a surgically implanted small, complex electronic device that can help to provide a sense of sound to a person who is profoundly deaf or severely hard-of hearing [2].Cochlear implant (CI) could be considered as a new mechanism of hearing when conventional hearing aids are ineffective. If the patient has disease in external or middle ear, it leads to conductive deafness. Medical or surgical treatment can correct this type of deafness.

Currently, the cochlear implant is the only kind of medical technology, which rehabilitates the hearing nerves. It differs from hearing aid devices, which generally extend up the volume. As a result, hearing loss patients, whose hair cells in the inner ear are damaged or destroyed, through a hearing aid, will hear the sound that lacks clarity in some degrees [3].

Hearing aids amplify sounds so they may be detected by damaged ears. On the other hand, cochlear implant bypasses the normal hearing mechanism and stimulates auditory neurons directly [4].

Actually, CI is a functional replacement of the biological sensory hair cells in the cochlea.

The speech processor is the heart of the cochlear implant.The speech processor provides the functional core of the cochlear implant by converting acoustic signal into electric signals [5].

The speech processor performs signal processing in which it selectively filters sound to prioritize audible speech and splits the sound into channels based on its frequency content. The latest speech processor of cochlear implant has 22 channels, we have implemented with 3 frequency bands.

2. WORKING OF NORMAL EAR

2.1 THE HUMAN EAR

Figure 2.1 shows a schematic view of the human ear showing the three distinct sound processing sections, namely: the outer ear consisting of the pinna, which gathers sound and conducts it through the external canal to the middle ear; the middle ear beginning at the tympanic membrane, or eardrum, and including three small bones, the malleus (also called the hammer), the incus (also called the anvil) and the stapes (also called the stirrup), which perform a transduction from acoustic waves to mechanical pressure waves; and finally, the inner ear, which consists of the cochlea and the set of neural connections to the auditory nerve, which conducts the neural signals to the brain[6].

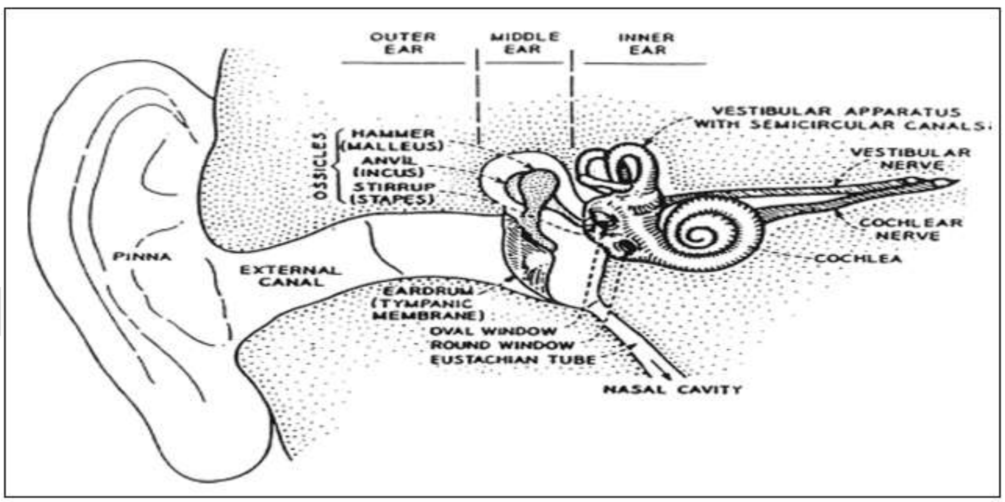

Fig 2.1: Schematic view of the human ear (inner and middle structures enlarged)

Outer ear: What most people think of as the ear is the readily-visible outer ear or pinna. Its shape makes listeners more sensitive to sounds from frontal directions. The outer ear simply funnels incoming speech pressure waves toward the eardrum (at the boundary between the outer ear and middle ear), where these variations are transformed into mechanical vibrations [7].

Middle ear: The eardrum (tympanic membrane) transfers power to the middle ear, which contains the ossicular bones: malleus (hammer), incus (anvil), and stapes (stirrup). Among the smallest bones of the body, these three amplify eardrum vibrations and send them to the oval window membrane of the inner ear. The main amplification effect is due to a large difference in surface area: big eardrum versus small oval window.

Inner ear: The inner ear contains the most important component of the ear, the cochlea, a very hard bony tube filled with lymphatic fluid. It converts mechanical vibrations of the oval window into electrical energy for its neural outputs. Tubes inside the cochlea taper off in diameter, from a wide base to a narrow apex. The tubes contain an important structure called the basilar membrane, that is about 32 mm long, and increases in thickness from the base to the apex (despite the narrowing of the tubes). Approximately 30,000 sensory hair cells connect the basilar membrane to the auditory nerve, which leads to the brain. These cells lie in several rows (inner and outer hair cells, having different functions) along the length of the cochlea.

Cochlea: The cochlea is a snail-shaped, curled tube located in the area of the ear where nerves are contained. Its function is to gather electrical signals from sound vibrations and transmit them to auditory nerve. The hearing nerve then sends these signals to the brain, where they are translated into recognizable sounds. If important parts of the cochlea are not working properly and the hearing nerve is not being stimulated, there is no way for the electrical signals to get to the brain; therefore, hearing does not occur.

2.2 Spatial frequency arrangement of the human cochlea

The cochlea is arranged like a rolled-up piano keyboard, as shown in box A in the figure 2.2. Lining the cochlea are many thousands of hair cells that convert the sound into electrical signals. Cochlear implants only have up to a couple of dozen electrodes, each of which performs a similar function to a hair cell or group of hair cells.

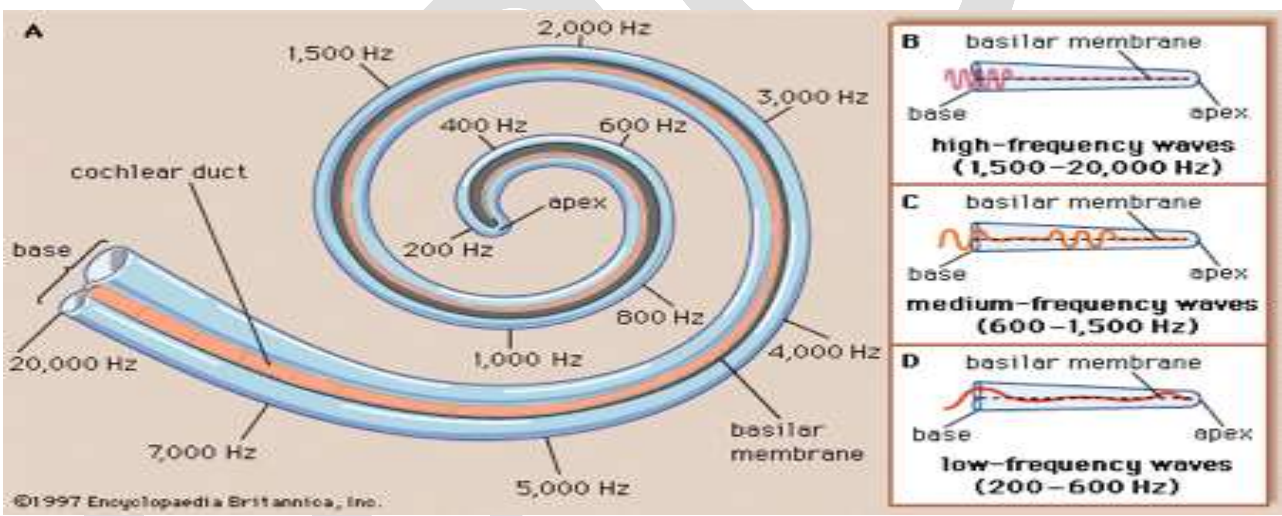

Figure 2.2: Spatial frequency arrangement of the human cochlea

Boxes B, C, and D illustrate the cochlea in an unrolled configuration. The base of the cochlea, which is where the sound enters, responds to the highest pitches. This is illustrated in box B. The apex, or innermost part of the cochlea, responds to the low-frequency tones, shown in box D. The locations in between the base and the apex correspond to the range of frequencies in between the two extremes. Damage of hair cells also results in subsequent degeneration of the adjacent auditory neurons. If the hair-cell and auditory nerve damage is excessive, the connection between the central nervous system and the external world is lost and the person who has such level of loss is recognized as being profoundly deaf.

However, some amount of living auditory neurons can still exist in the cochlea, even with extensive loss of hair cells. Direct electrical stimulus of these neurons can create a sound sensation in profoundly deaf people.

3. COCHLEAR IMPLANT

3.1 Functionality

The cochlear implant's functionality depends on the joint capabilities of the internal and external components. A cochlear implant is also referred to as a "bionic ear". The bionic ear transmits sound to the cochlea in a similar manner to normal hearing; however, it bypasses the outer and middle ear and directly simulates the auditory nerves with electric current. The amount of electric current determines the loudness, and the position of the electrodes determines the pitch.

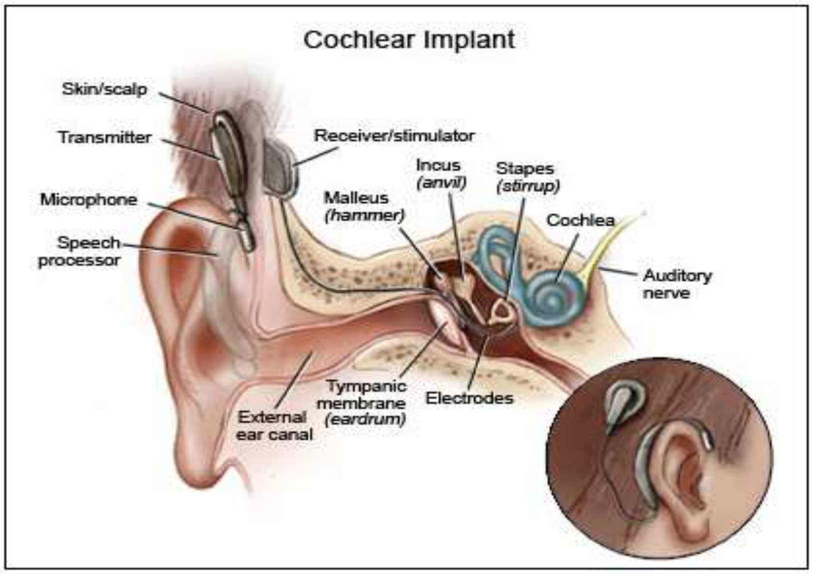

Figure 3.1: Cochlear Implant

The following subsections describe in detail the operation of the two parts.

1) External Functionality*:* The microphone detects sound vibrations, the speech processor performs signal processing and the transmitter transmits encoded signals and delivers power by electromagnetic induction to the internal components. The magnet aligns the external device to the internal implant to ensure high signal quality and power transmission efficiency.
2**) Internal Functionality:** The internal receiver receives signals from the external transmitter, the stimulator sends impulses to the inside of the cochlea, the electrodes stimulate the cochlear or auditory nerve, and the signals are then passed to the brain. Also, the magnet holds the external components in place.The cochlear implant consists of an internal coil, embedded under the skin behind the ear, and a wire (active electrode) introduced into the fluid filled spiral of the cochlea. The implant uses small electrical currents applied through the cochlea to the end of the auditory nerve, bypassing the damaged or missing hair cells.

3.2 How typical modern cochlear implant system works

An implant does not restore normal hearing. Instead, it can give a deaf person a useful representation of sounds in the environment and help him or her to understand speech [9].
An implant has the following basic parts:
• A microphone, which picks up sound from the environment.

• A speech processor, which selects and arranges sounds picked up by the microphone and filters and digitizes the sound into coded signals.

• A transmitter and receiver/stimulator, which receive signals from the speech processor and convert them into electric impulses.

• An electrode array, which is a group of electrodes that collects the impulses from the stimulator stimulate the remaining hearing nerve fibers in the cochlea and sends them to different regions of the auditory nerve and then to the brain for interpretation.

4. SPEECH PROCESSORS FOR COCHLEAR IMPLANTS

Different speech processing strategies are proposed and used successfully in cochlear implant devices [10].

- 3M/House Speech Processor
- Compressed Analog Speech Processor
- Continuous Interleaved Sampling Processor
- Feature-Based Speech Processors
- Spectral Maxima Sound Processor

Clinical studies on human subjects showed that CIS processors provide much better speech perception than CA processors. In some commercial cochlear implant devices, like the Clarion Multi-Strategy Cochlear Implant System from Advanced Bionics Corp., both CIS and CA processing strategies are used.

Spectral maxima sound processor is very popularly used and significantly better than other speech processors.

The proposed speech processor that we designed is different than Spectral Maxima Sound Processor and much simpler than other processors.

4.1 Spectral Maxima Sound Processor

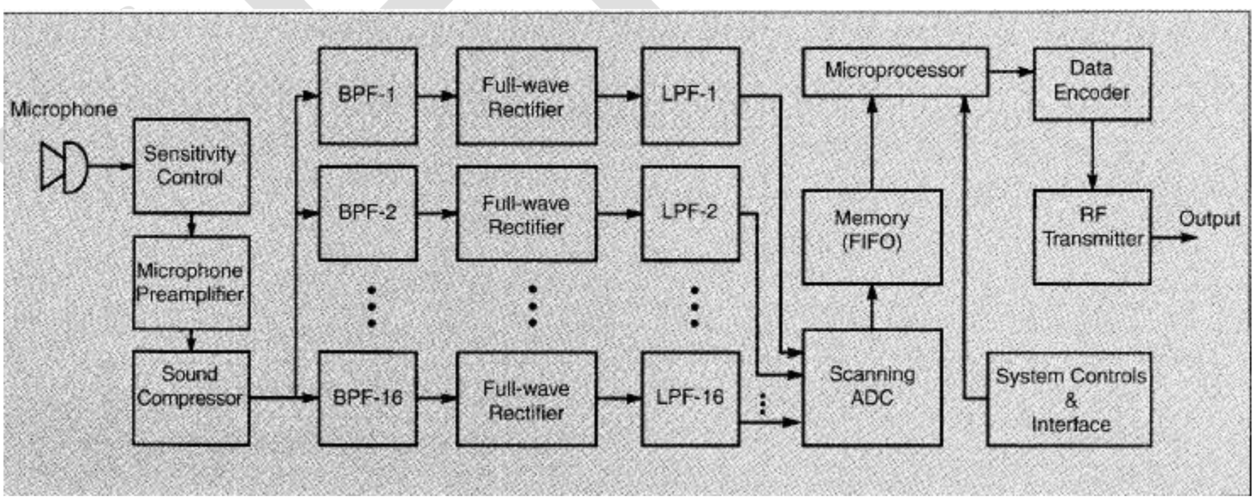

Figure 4.1: Spectral Maxima Sound Processor

The spectral maxima sound processor (SMSP) was first successfully used in 1989. A block diagram of a modified version of the SMSP is shown in figure 4.1. The processor includes sensitivity control, a microphone preamplifier, and a sound compressor followed by 16 band pass filters, full-wave rectifiers; low pass filters for analog signal processing. A scanning analog to- digital converter (ADC) is used to convert band signals into digital form with 8-bit resolution, and digitized signals are stored into a first-in first-out

(FIFO) memory. Digitized spectral information is processed by a microprocessor and the maximum amplitude or amplitudes of the entire speech spectrum is determined. Depending on the external control parameter values, such as loudness and the implantee's stimulus threshold levels, and the position of the spectral maxima, the microprocessor transfers the electrode numbers with stimulus levels to the data encoder. The data encoder converts data frames into pulse streams and sends them to the RF transmitter.

5. HARDWARE DESIGN

The proposed block diagram of Hardware Simulation of Speech Processor for Cochlear Implant is shown in a figure 5.

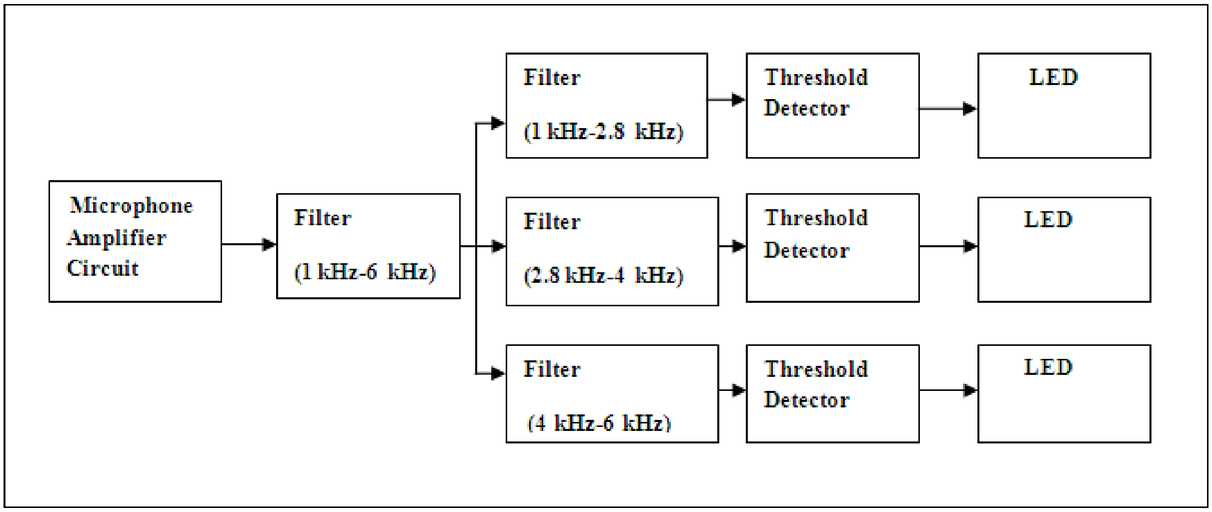

Figure 5: Complete Block Diagram

In this system a Microphone amplifier circuit, fourth order band-pass Butterworth filter (1 kHz-6 kHz), three filters, Threshold Detectors are used which helps to glow LEDs according to input sound signal. The block diagram consists of following blocks

- Microphone amplifier circuit: The basic function of Microphone amplifier circuit is to amplify the input audio signal using condenser microphone.
- Filter (1 kHz-6 kHz): The amplified audio signal from Microphone amplifier circuit is given to fourth order band-pass Butterworth filter having cutoff frequency of 1 kHz to 6 kHz.
- A Microphone amplifier and fourth order band-pass Butterworth filter is followed by bank of filters, Threshold Detectors and LEDs for analog signal processing.

  The 1 kHz-6 kHz frequency is divided into three sub-bands whose ranges are 1 kHz to 2.8 kHz, 2.8 kHz to 4 kHz and 4 kHz to 6 kHz using three fourth order band-pass Butterworth filters and outputs of those three filters are given to three LEDs through Threshold detectors.

5.1 Microphone amplifier circuit

The entire circuit consumes a very small amount of power within the range of 10 millwatts. Furthermore, the voltage Requirement of every major component is within the range of 1.8 volts and 15 volts. Therefore, for portability, a 3V DC battery is used to power the circuit [11].

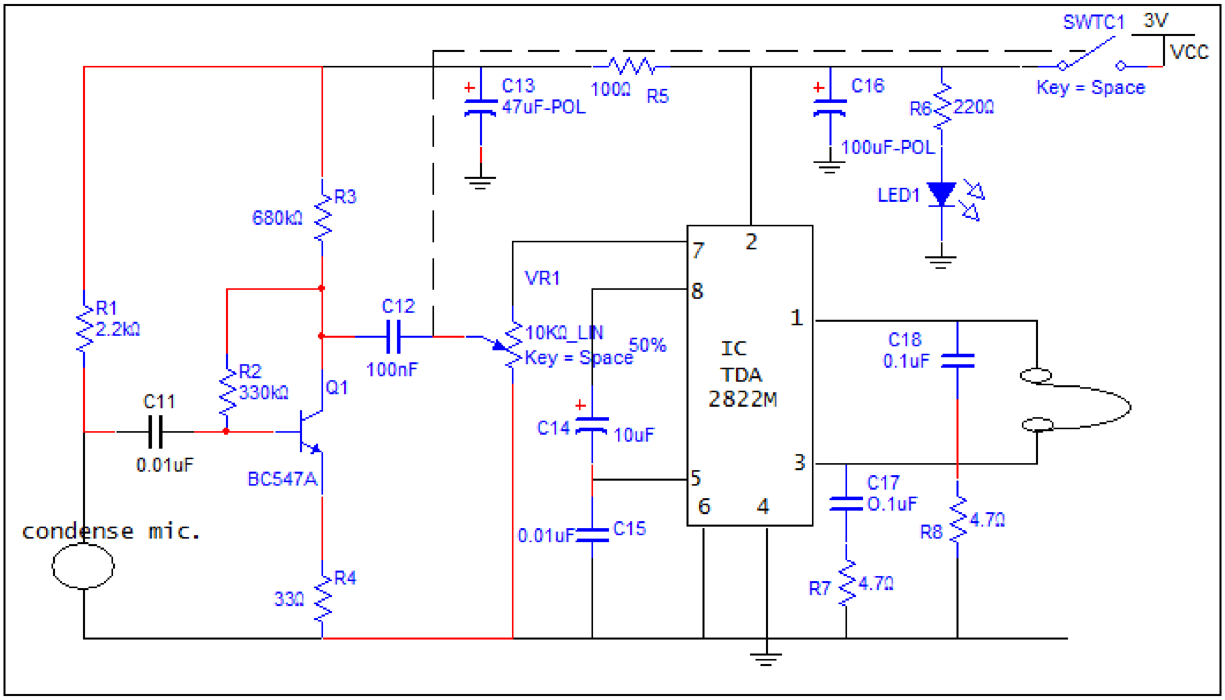

Figure 5.1 Microphone amplifier circuit

A transducer is a device which converts one form of energy into another form. The transducer used in this circuit is the condenser type microphone.

The pre-amplifier stage of this circuit was designed to produce a gain (A) of 500, so that the faint, weak signal produced by the microphone will be amplified 500 times before being further processed. The transistor selected for this purpose is BC547.

The medium power amplifier amplifies the output of the pre-amplifier to an audible level. It comprises of the TDA2822M IC and those external components needed to make the IC function properly.

A 32 ohms earphone is used in the output unit of this circuit as recommended by the manufacturers of the TDA2822M IC.

The aim of this circuit was to design a system that pre-amplifies an acoustic signal Picked up by a condenser microphone.

Capacitors C11 and C12 are called coupling capacitors. Their functions are to block any DC components in the input and outputs of the pre-amplifier. The pre-amplifier comprises of R5 and capacitor C13 which decouples the power supply of the preamplifier stage, while capacitor C12 and resistors, R2, R3 and R4 with transistor T1forms a negative feedback amplifier which stabilizes the overall gain (A). Resistor, R4 is known as an emitter swamping resistor which also adds stability to the amplifier. The medium power amplifier amplifies the output of the pre-amplifier to an audible level. It comprises of the TDA2822M IC and those external components needed to make the IC function properly. This other external components are capacitors C14, C15, C16, C17, C18 and resistors R6 and R7. Resistor, R5 and capacitor, C13 form an RC decoupling circuit which are connected across the power supply to smooth out noise. Finally a 32 ohms earphone is used in the output unit.

5.2 Band Pass Filter

A band-pass filter is a circuit which is designed to pass signals only in a certain band of frequencies while attenuating all signals outside this band. The parameters of importance in a band-pass filter are the high and low cut-off frequencies, the bandwidth (BW), the centre frequency f_c, and the selectivity or Q [12].

The key characteristic of Butterworth filter is that it has a flat passband as well as stopband.A band-pass filter has a passband between two cutoff frequencies f_H and f_L , where $f_H > f_L$ and two two stopbands $0<f<f_L$ and $f>f_H$. The bandwidth of the band-pass filter is, BW = $f_H - f_L$,f_c is the centre frequency since it is approximately at the centre of the passband on stopband.A band-pass filter has a centre frequency f_c and is defined as $\sqrt{f_H f_L}$, where f_H is high cutoff frequency (Hz) and f_L is low cutoff frequency (Hz).

A fourth order band-pass filter is formed by connecting in series or cascading second order high-pass filter and second order low pass filter. As the order of the filter increases, so does its size. Also the accuracy declines, in that the difference between the actual stopband response and the theoretical stopband response increases with an increase in the order of the filter. The overall gain of the filter is equal to the product of the individual voltage gains of the filter section.

A band-pass filter has a passband between two cut-off frequencies f_H and f_L such that $f_H > f_L$.Any input frequency outside this passband is attenuated.

Basically there are two types of band-pass filters-

- Wide band-pass filter
- Narrow band-pass filter

If figure of merit or quality factor, Q<10 it's a Wide band-pass filter and if Q>10 it's a Narrow band-pass filter.

$Q = \frac{fc}{BW}$

BW = $f_H - f_L$

The rate at which the gain of the filter changes in the stopband is determined by the order of the filter. For second order low-pass filter the roll-off rate is 40 dB/decade, and by contrast the second order high pass filter the gain increases at the rate of 40 db/decade in the stopband , that is until $f = f_L$.

To obtain ±40 dB/decade band-pass, second order high-pass and second order low-pass sections are connected in series. Order of the band-pass filter depends on the order of the high-pass and low-pass filter section.

5.2.1 Filter (1 kHz-6 kHz)

Here f_L=1 kHz and f_H= 6 kHz

- C1=C2=C3= C4=0.0047 µF
- $R_1 = R_1'$= 27 kΩ
- $R_2 = R_3$= 33.804 kΩ
- $R_2' = R_3'$= 5.634 kΩ

- $R_F=R_F'= 22\ k\Omega$

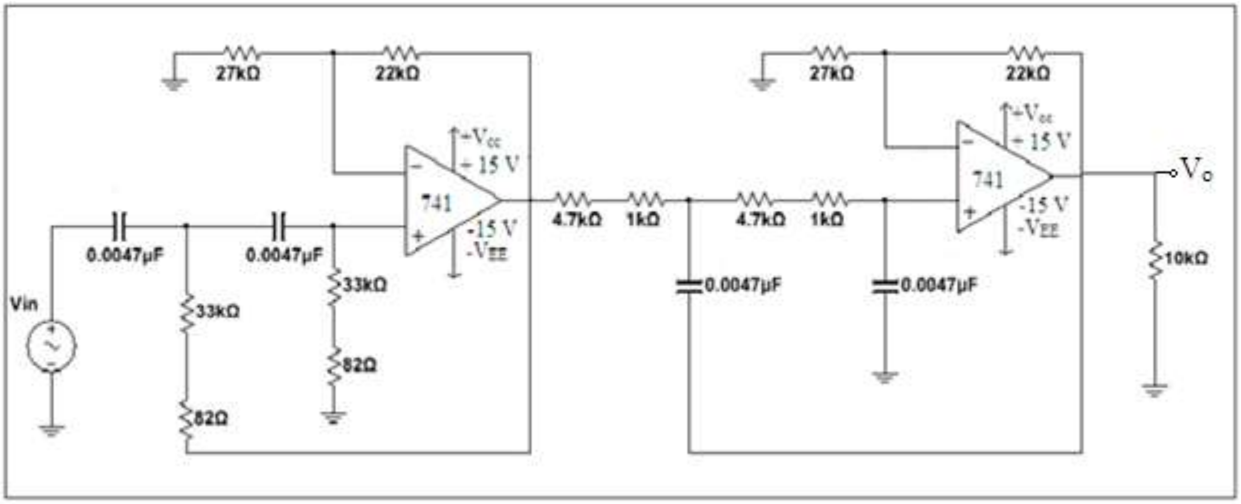

Figure 5.2 fourth order band-pass butterworth filter (1 kHz- 6 kHz)

The above fourth order band-pass butterworth filter is a combination of second order high pass butterworth filter and second order low pass butterworth filter.

And frequency response is,

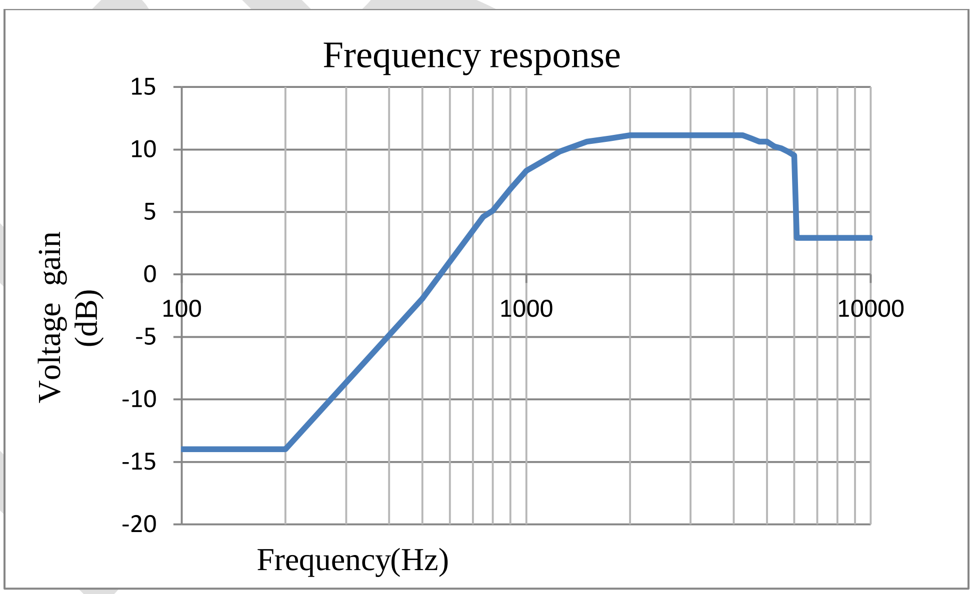

Figure 5.2.1: Frequency response of a filter (1 kHz- 6 kHz)

5.3 Bank of Band-pass filters

All three filters that are used in the circuit are fourth order band-pass butterworth filters, designed in such a way that ,these filters will divide frequency range 1 kHz- 6 kHz into 3 sub-bands .They are listed as,

- Frequency band (1 kHz - 2.8 kHz)

- Frequency band (2.8 kHz- 4 kHz)
- Frequency band (4 kHz – 6 kHz)

5.3.1. Second order high pass butterworth filter for Frequency band (1 kHz - 2.8 kHz)

Here f_L=1 kHz and f_H= 2.8 kHz

Design remained the same as shown in Figure 5.2; values of components are changed as follows,

- C1=C2=C3= C4=0.0047 µF
- R_1=R_1'= 27 kΩ
- R_2=R_3= 33.804 kΩ
- R_2'=R_3'= 12.093 kΩ
- R_F=R_F'= 22 kΩ

And its frequency response is,

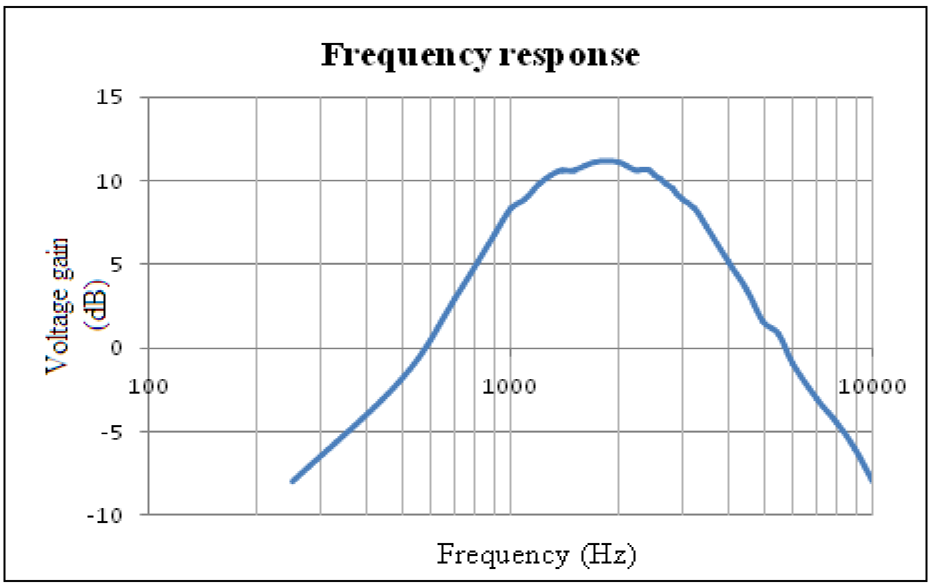

Figure 5.3.1: Frequency response of a filter (1 kHz-2.8 kHz)

5.3.2 Second order high pass butterworth filter for Frequency band (2.8 kHz- 4 kHz)

Here f_L=2.8 kHz and f_H= 4 kHz

Design remained the same as shown in Figure 5.2; values of components are changed as follows,

- C1=C2=C3= C4=0.0047 µF
- R_1=R_1'= 27 kΩ
- R_2=R_3= 12.093 kΩ
- R_2'=R_3'= 8.465 kΩ
- R_F=R_F'= 22 kΩ

And its frequency response is,

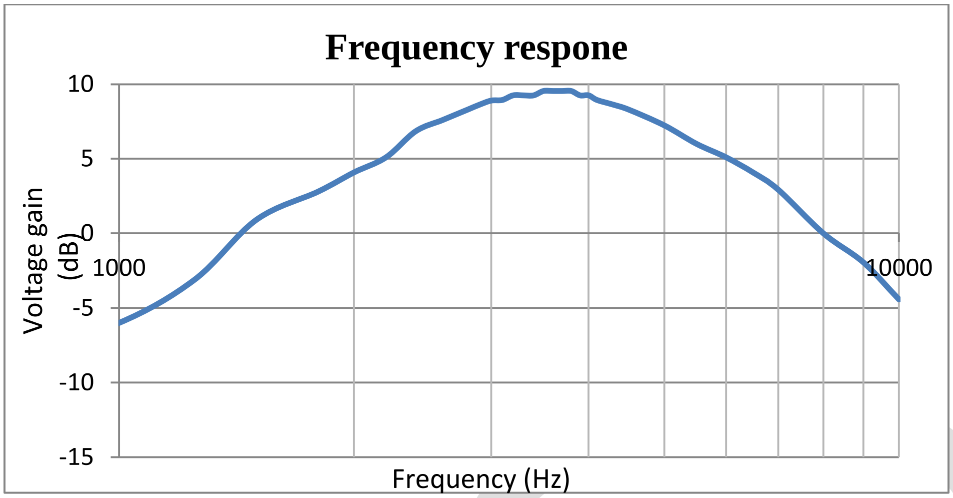

Figure 5.3.2: Frequency response of a filter (2.8 kHz- 4 kHz)

5.3.3 Second order high pass butterworth filter for Frequency band (4 kHz - 6 kHz)

Here f_L=2.8 kHz and f_H= 4 kHz

Design remained the same as shown in Figure 5.2; values of components are changed as follows,

- C1=C2=C3= C4=0.0047 µF
- $R_1=R_1'$= 27 kΩ
- $R_2=R_3$= 8.465 kΩ
- $R_2'=R_3'$= 5.634 kΩ
- $R_F=R_F'$= 22 kΩ

And frequency response is,

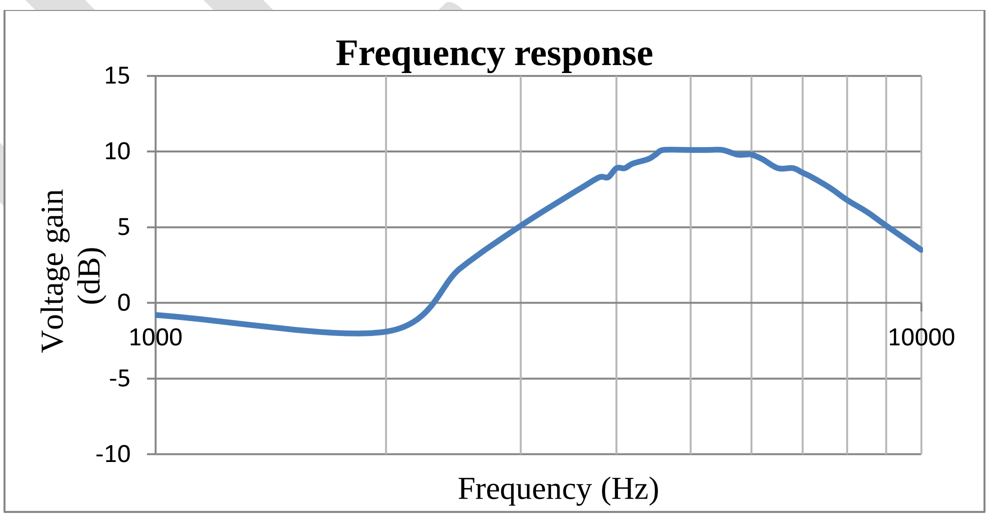

Figure 5.3.3: Frequency response of a filter (4 kHz- 6 kHz)

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5.4 Threshold Detector

Non-inverting Comparator

Comparators are used in circuits such as voltage level detectors. Comparator as its name implies, compares a signal voltage on one input of an op-amp with a known voltage called the reference voltage on the other input. A comparator finds its importance in circuits where two voltage signals are to be compared and to be distinguished on which is stronger.

It is called a non-inverting comparator circuit as the time varying signal voltage V_{in} is applied to the non-inverting terminal and that signal is output from the respective filter. The fixed reference voltage V_{ref} is give to the inverting terminal (-) of the op-amp. When the value of the input voltage V_{in} is greater than the reference voltage V_{ref} the output voltage Vo goes to positive saturation. This is because the voltage at the non-inverting input is greater than the voltage at the inverting input.

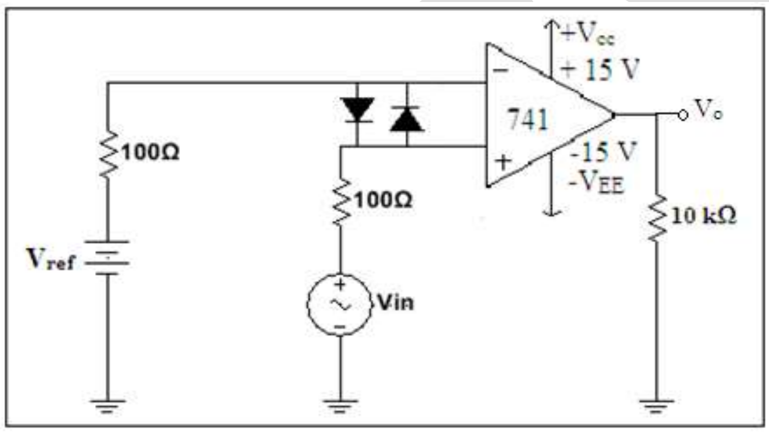

Figure 5.4: Non-inverting Comparator

When V_{in} is less than V_{ref}, the output voltage V_o is at $-V_{sat}$ because the voltage at (-) input is higher than at the (+) input. On the other hand when V_{in} is greater than V_{ref}, the (+) input becomes positive with respect to the (-) input and V_o goes to $+V_{sat}$. Thus, output voltage Vo changes from positive saturation point to negative saturation point whenever the difference between Vin and Vref changes. The comparator can be called a voltage level detector, as for a fixed value of Vref, the voltage level of Vin can be detected.

It is nothing more than an open-loop op-amp, with two analog inputs and a digital output V_o.

In Figure 5.4 the diodes D1 and D2 protect the op-amp from damage due to excessive input voltage V_{in}. The resistor R =100 Ω in series with V_{in} is used to limit the current through diodes D1 and D2.To reduce offset problems, a resistance R=100 Ω is connected between inverting terminal and V_{ref}.

In total three threshold detectors are used in a circuit.

Design of all three threshold detectors is similar. The value of V_{ref} is set differently in those circuits.

For first threshold detector value of V_{ref} is set as 2.8 V.

For second threshold detector value of V_{ref} is set as 2.7 V.

For third threshold detector value of V_{ref} is set as 2.9 V.

Three LEDs are placed at the outputs of threshold detectors.

RESULTS

The speech processor described here was successfully tested in laboratory.

Input	1st LED(1 kHz- 2.8 kHz)	2nd LED (2.8 kHz- 4 kHz)	3rd LED (4 kHz- 6 kHz)
2.5 kHz pure sine audio signal	Yes	No	No
3.6 kHz pure sine audio signal	No	Yes	No
5 kHz pure sine audio signal	No	No	Yes
random ringtone	Blinked		

Frequency of some words was found using spectrum analyzer, and for particular set of words having same frequency, same LED glowed every time, the same set of words was given as input to condenser microphone.

Results seem to be 99 % true.

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CONCLUSIONS:

In this paper a simple system has been reported for hardware simulation of speech processor for use in a typical cochlear implant system.

An acoustic signal is picked up by a condenser microphone and then further amplified, and that signal is given to filter (1 kHz- 6 kHz), then simultaneously given to bank of three band-pass filters through Threshold detectors to three LEDs. The designed and constructed circuit is tested successfully.

So it can be seen from the results shown above, only 3 LEDs are used corresponding to 3 different frequency bands. We are planning to increase the number of bands in due course so that many words can be implemented using speech processor.

REFERENCES:

[1] M. Manrique, A. Ramos, C. Morera, C. Cenjor, M. J. Lavilla, M. S. Boleas, F. J. Cervera-Paz, Analysis of the Cochlear Implant as a Treatment Technique for Profound Hearing Loss in Pre and Postlocutive Patients, Acta Otorrinolaringol Esp 2006; 57: 2-23

[2] NIDCD Fact Sheet: Cochlear Implants, NIH Publication No. 11-4798,March 2011

[3] N. Saimai, C. Tantibundhit, C. Onsuwan, N. Thatphithakkul, Speech Synthesis Algorithm for Thai Cochlear Implants,2012 IEEE,pg.1

[4] Shaoyang Wu, Songping Mai, Chun Zhang, FPGA Implementation of CIS Speech Processing Strategy for Cochlear Implants,pg.1, 2011 IEEE

[5] Susan B.Waltzman,J.Thomas Ronald Jr. ,Cochlear Implants,Second edition,Thieme Medical Publishers

[6] Lawrence R. Rabiner and Ronald W. Schafer ,Introduction to Digital Speech Processing, Foundations and Trends in Signal Processing Vol. 1,pg.26

[7] Li Deng Douglas O'Shaughnessy, SPEECH PROCESSING A Dynamic and Optimization- Oriented Approach,pg 109

[8] Ariel Moctezuma, Jane Tu,An Overview of Cochlear Implant Systems,pg 1-4, BIOE 414, Spring 2011

[9] Fan-Gang Zeng, Senior Member, IEEE, Stephen Rebscher, William Harrison, Xiaoan Sun, and Haihong Feng, Cochlear Implants: System Design, Integration, and Evaluation,pg 4-5 VOL. 1, 2008

[10] Suat U.Ay,Fan-Gang Zeng,Bing J Sheu,Hearing with Bionic Ears,May 1997 IEEE,pg 3-6

[11] Yusuf M.A, Zainab U.S, M.I Ilyasu, I.B. Shehu, I H Jibrin, S. Abdullahi and A. Adedeji, Design and Construction of Hearing Aid Device, (IOSR-JESTFT), Volume 5, Issue 3 (Jul. - Aug. 2013)

[12] Ramakant A. Gayakwad, "OP-AMPS AND LINEAR INTEGRATED CIRCUITS",3RD EDITION, Prentice-Hall India 2011