SIGNAL PROCESSING FOR COCHLEAR IMPLANT

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Department of Electrical and Information Engineering

Faculty Of Engineering

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EE6209: Introduction to Biomedical Engineering

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4. Introduction of cochlear Implant

Early stages the deafness is unsolvable. After some researches they found a way that stimulation of auditory nerve. But the receiving signal was unintelligible. After more researches they implemented a device is called cochlear implant. The device is implanted inner ear. The device is bypassing the normal ear and stimulate the electrical signal instead of auditory nerve[1].

The device receives the signal and after prosses the signal is sent to the brain through auditory nerve as electrical signal. The cochlear shape is spiral design. There is inside location name is called apex, end point called base. The frequencies are spread apex to base. At apex sense frequency is near 20Hz, the base sense frequency is 20kHz. Each location sensitive for a particular frequency. The same thing has to happen this cochlear implant device also. The coding frequency has motivated no of channels in signal processing that is called volley theory.

And the auditory nerve activates rates proportional to period of signal up to 5kHz frequency range. Overall, at low frequency range a single nerve activates, but larger frequency range group of auditory nerve activate[2].

Fig-1 shows block diagram of cochlear implant. The microphone picks up the sound. The signal processor processes the signal and transmitter sent the signal to implant. Then the signal sent to the nervous system by electrode. If single channel used that enough one electrode. if more channel uses, we want array of electrode.

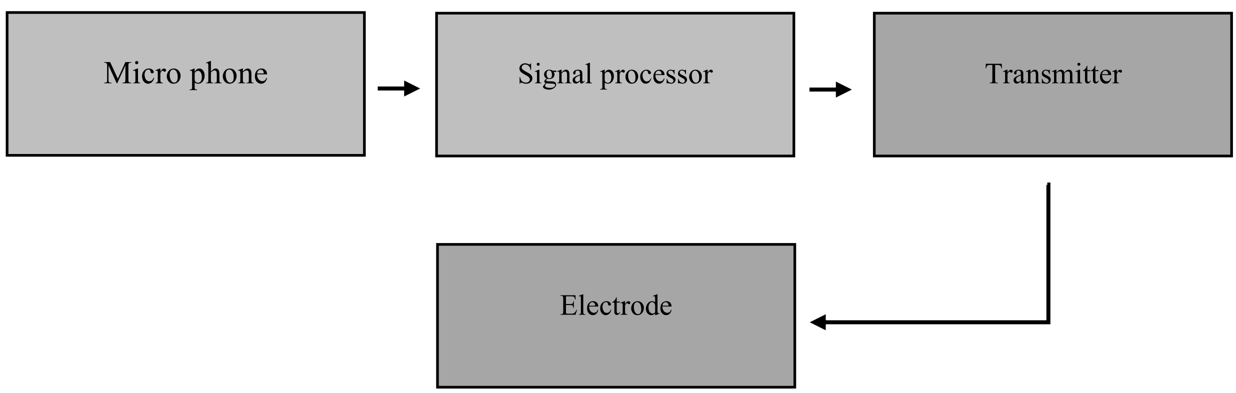

Figure 1: Block diagram of cochlear implant

5. Signal processing

Signal were firstly proceeding through lowpass filter (low-pass below 1200 Hz, -6 dB per octave) and band passed into the 8 frequency bands (channel) by using sixth order butter worth filter. The center frequency and bandwidth are given Table-01. The bandwidth of the filter is at 3dB down from the passband level. The envelop signal detect by lowpass filter and halfwave rectification. The lowpass filter is second order butter worth (cutoff frequency- 400Hz). The sinusoidal wave created with center frequencies of 8 channel frequency bands. The amplitude was found by rms values energy of envelop signal. The frequencies were evaluated by FFT of the audio segment. The prosthetic speech amplitude also adjusted with original audio signal amplitude. The block diagram of speech processor shown below.

Table 1: center frequencies and bandwidths of 8 channels

Channel NO	1	2	3	4	5	6	7	8
Center frequency	366	526	757	1089	1566	2252	3241	4662
Band width	131	189	272	391	563	810	1165	1676

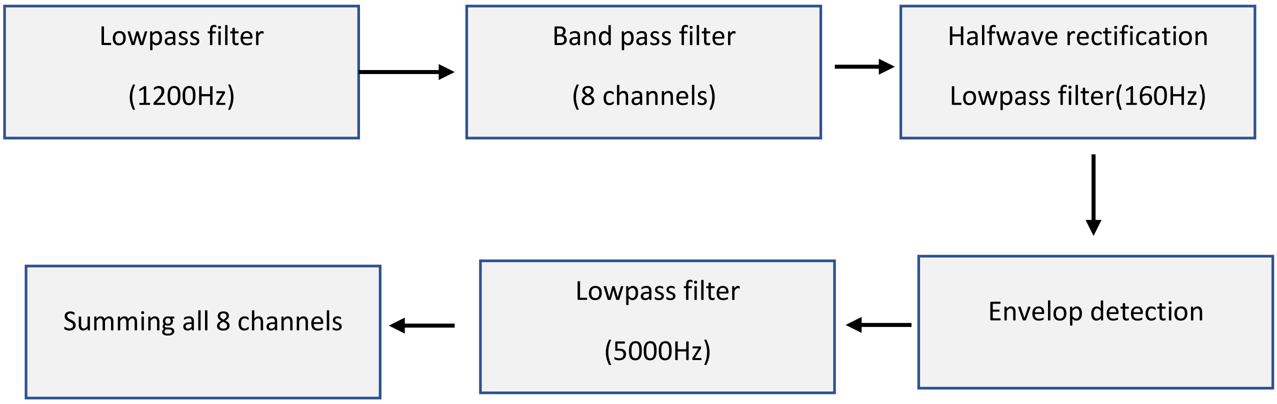

Figure 2: speech processor block diagram

6. Detail and Descriptions

6.1 Digital filters

Digital filters are more popular than analog filters. Analog filters are more cheap, fast and large dynamic range in amplitude and frequency. But the digital filters are more better performance than analog filters. The filters mainly use for two uses signal separation and signal restoration. The signal separation is used for filter the signal from noise, inference. The signal restoration is done for the distortion of signal. Digital filters are more important for Digital signal processing. Digital filters are the reason for DSP more famous. The filters implemented by many ways. The first way is by convolving the input signal with the filter's impulse response. Another way to make the devices called recursion. The filters implemented by convolution. Recursive filters are called infinite impulse response filters (IIR). And filters carried out by convolution are called Finite impulse response filters (FIR).

There are some different between FIR and IIR filters.

FIR filter characteristics

- Finite Response
- Linear phase characteristics
- Stable
- Depends only on input
- Good for higher orders/ tap delays
- No poles, only zeros

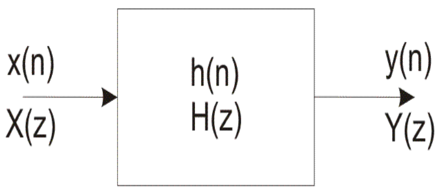

Figure 4: FIR block diagram

IIR filter characteristics

- Infinite Response
- Non-linear phase characteristics
- Recursive, Feedback
- Stability issues
- Computationally efficient; compact; same frequency response with less coefficients
- Less coefficients
- less storage

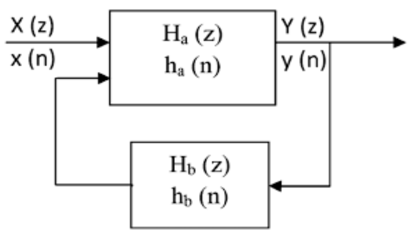

Figure 3: IIR block diagram

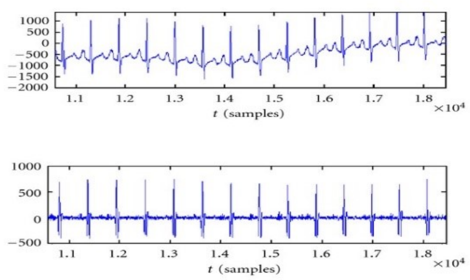

Figure 5:Example After filtered signal by 2 channels

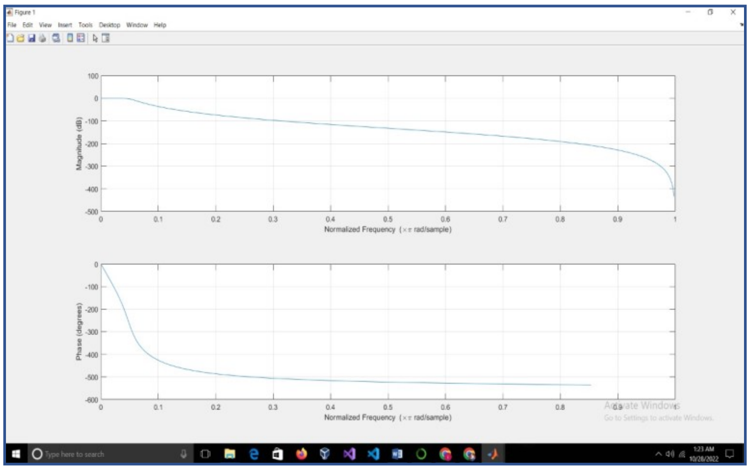

Figure 6: magnitude and phase response of lowpass sixth order butter worth filter (1200Hz)

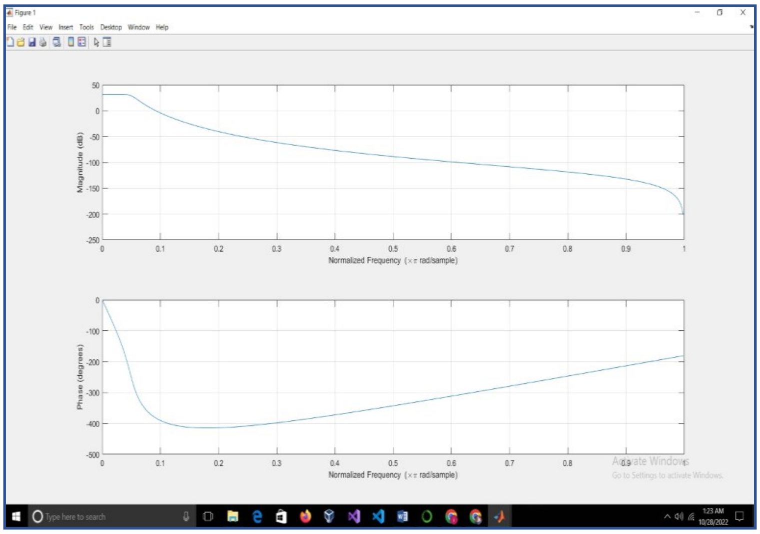

Figure 7: magnitude and phase response of lowpass second order butter worth filter (160Hz)

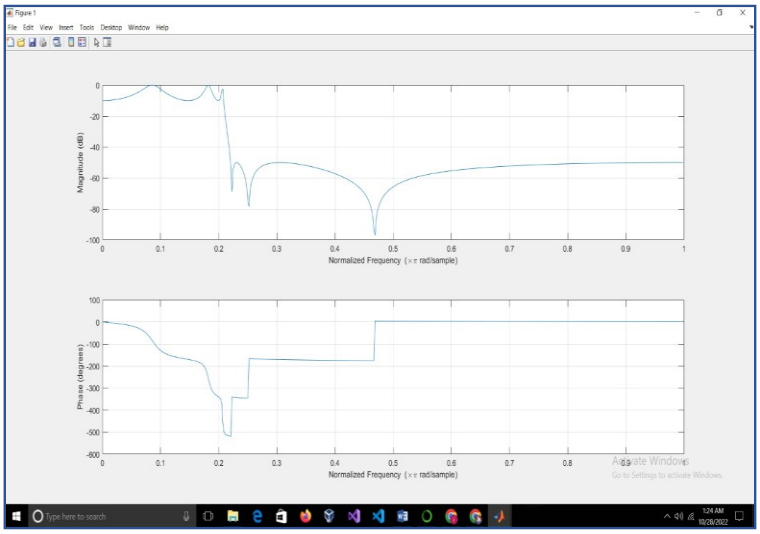

Figure 8: magnitude and phase response of lowpass sixth order butter worth filter (5000Hz)

6.2 Envelop detection

The envelop detection find the outline function of the modulated signal. The detector detects original signal from a high frequency modulated input signal. The device is electronical circuit corrects the input signal and release the new signal as an envelop for the original signal. Some detectors have diode therefore, the signal rectified then the output is single signal unless there are two signals can be found. Fig 4 shows electronic circuit of envelop detector.[3]

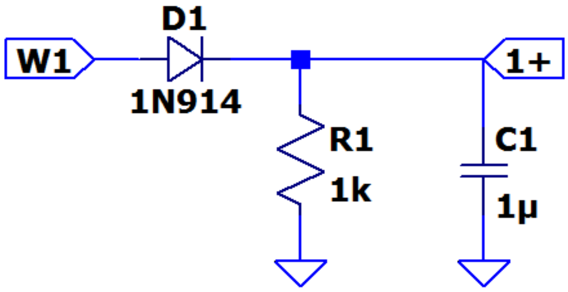

Figure 9:electronic circuit of envelop detector

The diode rectified the signal and there is a low pass filter. Low pass filter is filtered out the high frequency signal. Therefore, the device can detect the original signal. The rectifier and capacitor can detect peak point of the input signal. After detection low pass filter find the envelop of the input signal.

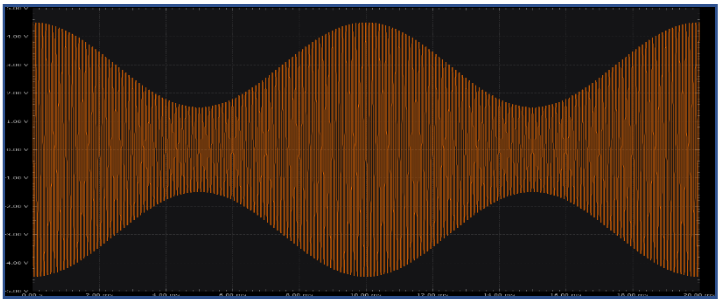

Figure 11:Example Input signal in time domain

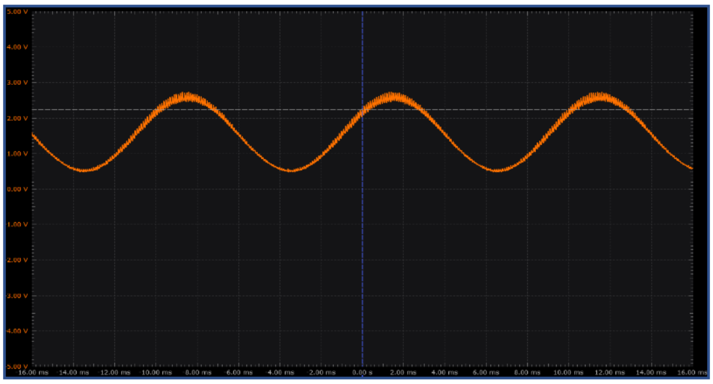

Figure 10: Example diagram Envelop signal in time domain with halfwave rectification

6.3 Rectify the signal

Here we use the half wave rectifier. There are more noises and interference can be happened. The input signal there are more negative signal. The negative part has to be eliminated otherwise in envelop detection there are two outputs can be found. The rectifier diode already in the envelop detector circuit.

Figure 12:Example diagram Half wave rectified wave before envelop detection

7. Results

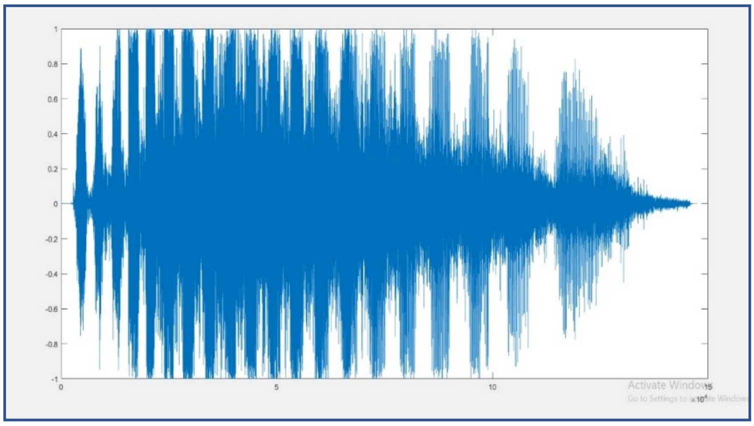

Figure 13: Input signal of the audio file

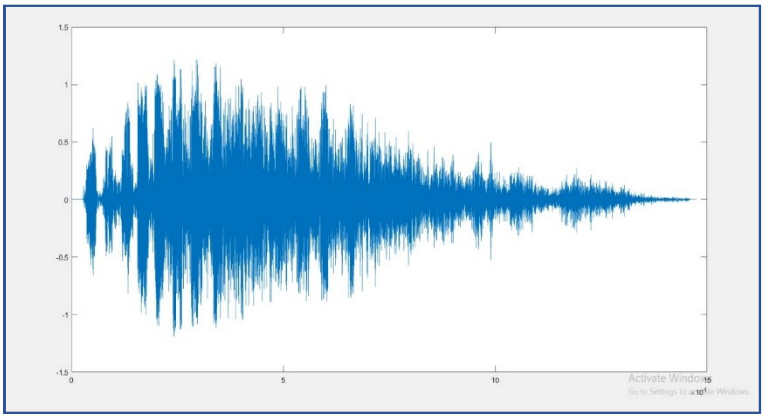

Figure 14: output after the lowpass filter cutoff frequency-1200Hz

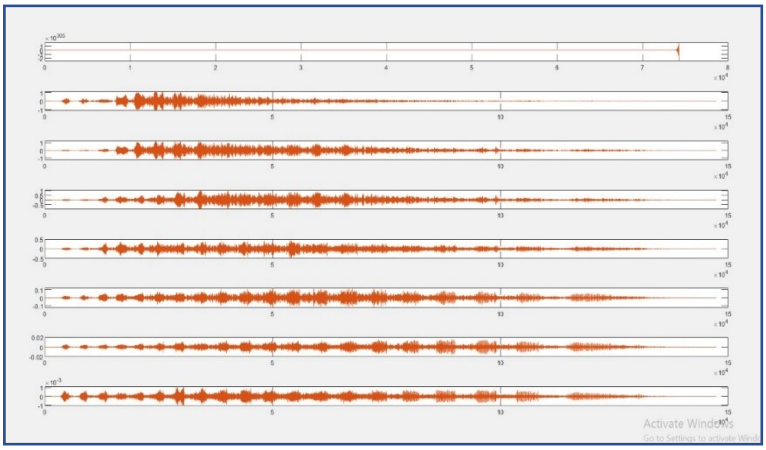

Figure 15: output of the sixth order bandpass filter into 8 channels

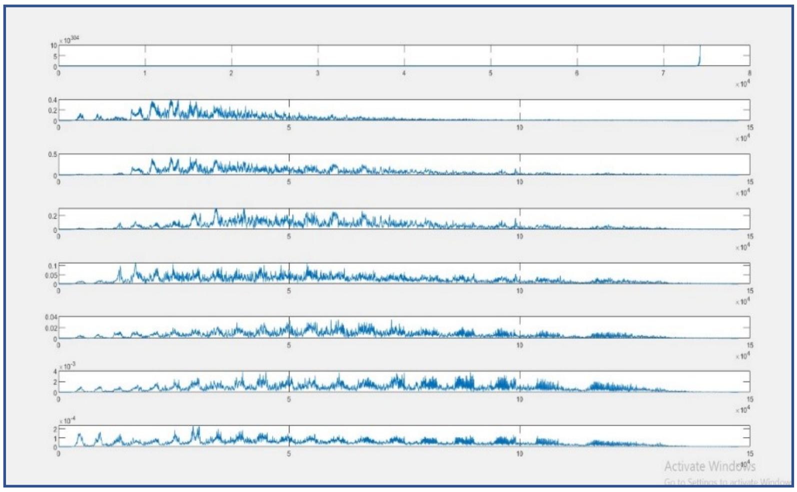

Figure 16: Rectified curve in time domain

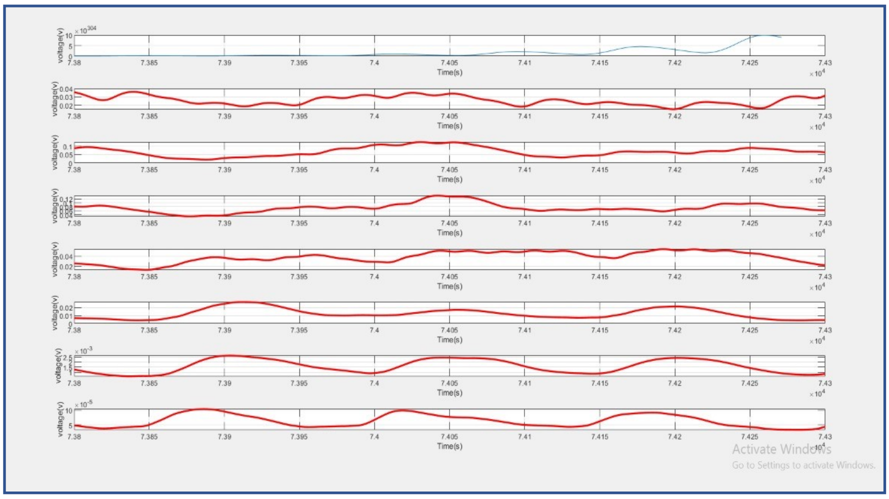

Figure 17: Envelop signal

8. Discussion and conclusion

Cochlear is more sensitive organ in the body. The project is designing a speech processor for a cochlear implant. In nature there are small different frequencies between sound signals. The device must be higher accuracy with small number of channels. If we want to clearer in the audio, the no of channels has to be increased. And also, we unable to increases electrodes. The best cochlear is small number of channels with more efficient signal processor. But the difficulty is identifying the different word in same pronouns. At least eight channels must need to approach prosthetic sound performance.

The process recommended from the readings was followed and the output of the speech processor for different input were observed. The output was much clear and audible with a headset on. The performance might vary depending the frequency range of the input. Changes in the low pass filter cut-off frequency can also result better performances based on the type of input. Therefore the performance can vary depending on the input, thus the speech processor has to be improved such that it works well with a good range of inputs.

9. References

[1] Philipos Loizou, Assistant Professor, Department 42 of Applied Science, University of Arkanas at little rock. : “Introduction to cochlear implant”, This article is adapted from a version that appeared in the September 1998 issue of IEEE Signal Processing Magazine, (vol. 15, no. 5, pp. 101-130).

[2] Philipos C. Loizoua) Department of Electrical Engineering, University of Texas at Dallas, Richardson, Texas 75083-0688 Michael Dorman Department of Speech and Hearing Science, Arizona State University, Tempe, Arizona 85287 Zhemin Tu Department of Applied Science, University of Arkansas at Little Rock, Little Rock, Arkansas 72204-1099 ~Received 5 December 1998; revised 7 April 1999; accepted 21 May 1999. PACS numbers: 43.72.Ar, 43.71.Es [JMH], © 1999 Acoustical Society of America. [S0001-4966(99)01810-X]

[3] https://wiki.analog.com/ -copyright 1995-2022 Analog devices, -Inc. All Right deserved.

10. Contribution chart

Table 2: Contribution Chart

Section Details and total percentage of marks allocated for each section	Each member's % contribution		
	Ranjith B.	Mazar M.S.M.	Total per selection
Part A 10%	30%	70%	100%
Report			
Cochlear implant and signal processing 10%	70%	30%	100%
Description and detail of the Digital filters 10%	70%	30%	100%
Results 30%	70%	30%	100%
Discussion and conclusion 10%	30%	70%	100%
Code			
Filter Frequency Response 5%	40%	60%	100%
FFT plots and Quantization 5%	40%	60%	100%
Speech Processor 15%	30%	70%	100%