Implementation of Cochlear Implant Speech Processor Architecture in MATLAB

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ABSTRACT

Cochlear implants are electronic devices used to improve hearing in patients with sensorineural hearing loss by bypassing damaged portions of the ear to deliver electrical signals to the brain. Within this architecture, the speech processor plays a pivotal role in converting sound signals to electrical impulses, which can then be sent to electrodes, the auditory nerve, and ultimately, the auditory cortex of the brain. The goal of the present investigation was to implement all components of the speech processor within MATLAB. Specifically, using a pre-recorded speech signal in the form of .mp4, a pre-emphasis filter, 8 channel bypass filters, envelope detection, and modulation were created. Two types of modulation, namely White Noise and Pulse Modulations, were used, after which they were compared for the Peak Signal Noise Ratio. In actuality, this modulated noise can then be sent to the electrodes for further transmission to the auditory nerve. Following implementation, the vocoded/CI synthesized signal was compared with the original signal to identify any differences. Finally, speech output with 20 bandpass filters was checked to determine the optimum number of channels for the filters.

BACKGROUND INFORMATION

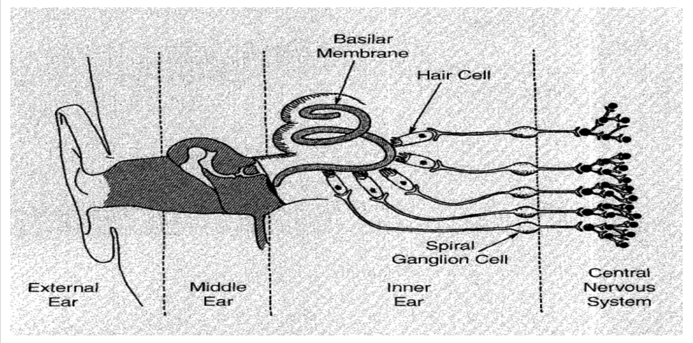

Figure 1: Anatomy of Human Ear

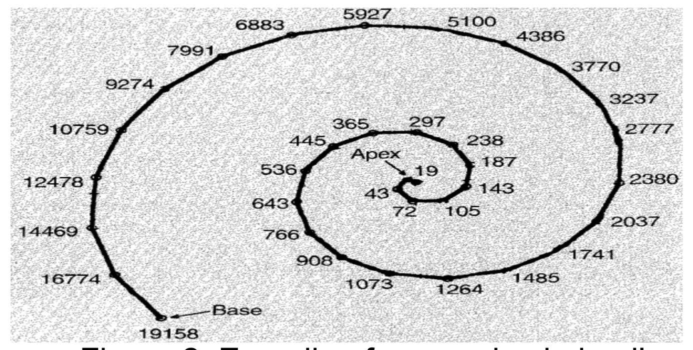

Figure 2: Encoding frequencies in basilar membrane

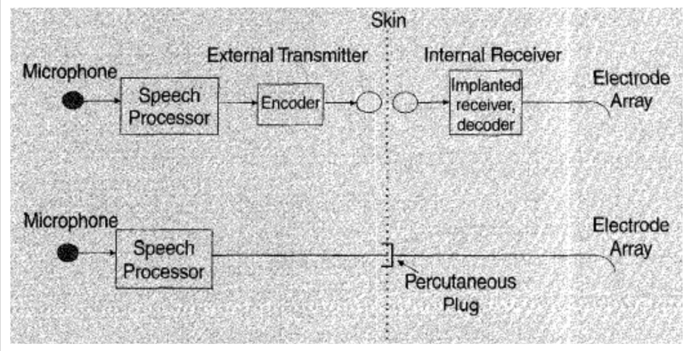

Figure 3: Cochlear Implant Architecture

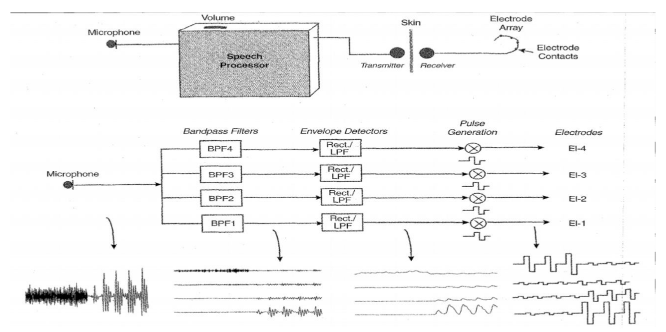

Figure 4: Signal Processing

OBJECTIVE

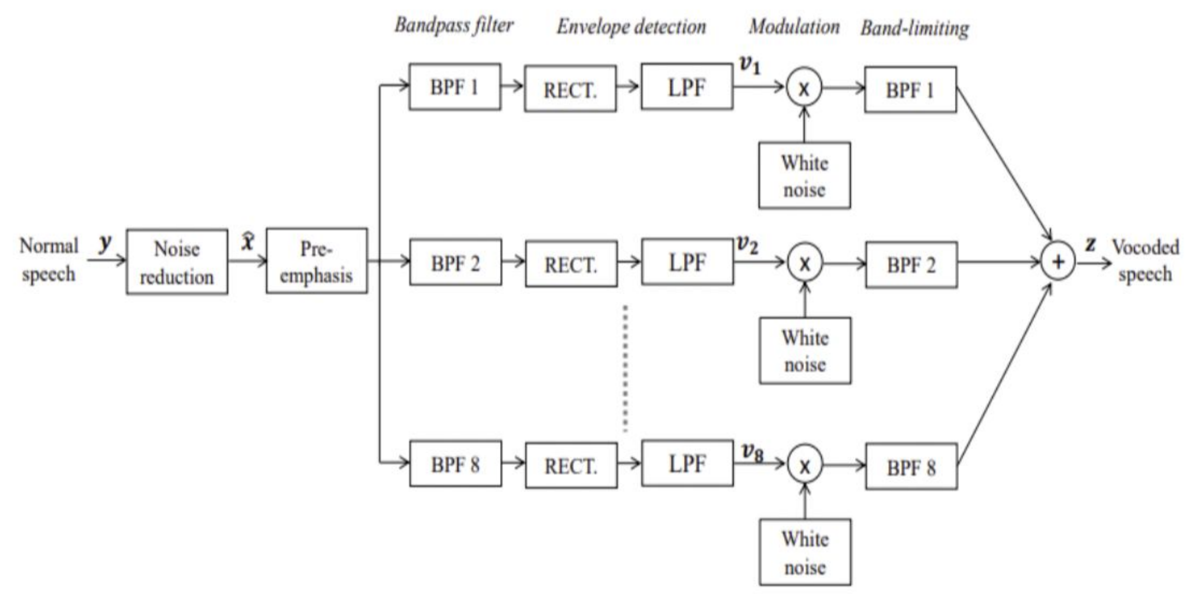

Figure 5: Speech Processor Architecture

This project aims to implement the above cochlear implant speech processor architecture in MATLAB.

MATERIALS AND METHODS

- MATLAB version '9.14.0.2206163 (R2023a)
- Speech.mp4 was used as an input to implement the architecture
- Speech was stored in vector [x, fs] and the number of samples in the signal were 253951
- Sampling frequency was 44100 samples/second and time duration of signal was 5.75 seconds

PRE-EMPHASIS FILTER – 1ST STAGE

- Aims to amplify the gain of high frequencies
- Implement using "High Pass Filter" with cutoff frequency of 1200 Hz
- Utilize Butterworth function to get butter coefficients (pre-a, pre-b) in order to mathematically determine new values of gain (Figure 6)
- Utilize Filter function to apply the butter coefficients on the 'x' signal
- Store filtered function after pre-emphasis on pre-y vector

BANDPASS FILTERS – 2ND STAGE

- 8-channels were used to filter pre-emphasis signal
- Select lower and upper cutoff frequencies (determine by using center frequencies and bandwidth for each channel)
- Normalize frequencies using Sampling Theorem (normalize to 0.99 if greater than 1, 0.01 if less than 1)
- Utilize Butterworth function to get butter coefficients for each channel for the lower and upper cutoff frequencies (Figure 7)
- Apply butter coefficients 'a' and 'b' of each bandpass filter on the pre-y vector to filter the gain of the signal for each of the 8 cutoff frequencies of the bandpass filter
- Repeat with number of filters = 20 to compare output differences

ENVELOPE DETECTION – 3rd STAGE

- Comprised of Rectification and implementation of Low Pass Filter for each of the 8 channel signals
- Rectification helps to suppress the negative gains of each of the 8 channel or filtered signals
- Extract 'Temporal Envelope' of each of the 8 filtered signals
- Implement Temporal Envelope using Low Pass Filter using butter function with cutoff frequency of 960 Hz
- Apply Filter function using the butter coefficients on each of the 8 rectified signals (Figure 8)

MODULATION – 4th STAGE

- Modulation helps to generate the signal by multiplying the envelope signal for each of the 8 channels with White Noise Modulation or Pulse (Sinusoidal) Modulation
- White Noise or Pulse Modulation represents the "Unvoiced Signal"; thus, multiply with envelope
- Utilize Random function to generate White Noise corresponding to the length of the sample (Figure 9)
- Develop Alternatively Modulated Noise using sinusoidal pulse detection (this noise can then be sent to the electrodes)

BANDPASS FILTER + VOCODED SPEECH – 5th STAGE

- Stage is utilized to check if signal (after synthesis of the 8 channel signals) is the same signal as the original sample
- Utilize Bandpass Filter with same butter coefficients (a and b) for each channel to filter the Modulated Envelope for each channel
- Add filtered signal for each of the channels to derive the vocoded speech
- Filter vocoded speech using a Low Pass Filter for cutoff frequency of 4000 Hz using butter and filter coefficients to get synthesis of speech signal
- Adjust synthesized speech signal for amplitude difference in the original signal
- Play final synthesized signal to compare with original speech signal (Figure 11)

COMPARISON OF NOISE VOCODER AND SIN WAVE VOCODER

- Check Peak Signal to Noise Ratio for noise vocoder vs sin wave vocoder

RESULTS

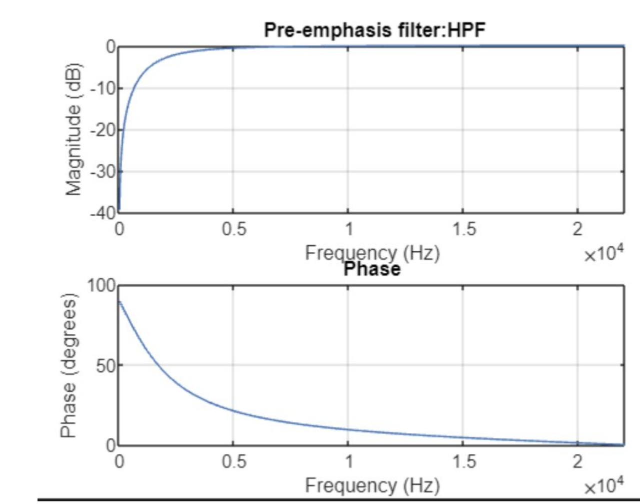

Figure 6: Pre-emphasis filter with 1200 Hz Cutoff Frequency

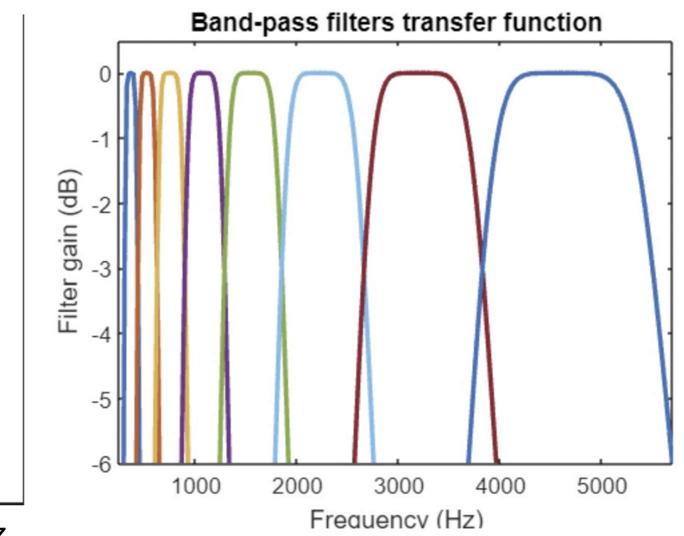

Figure 7: Bandpass Filter Transfer Function for each of the 8 BPF's

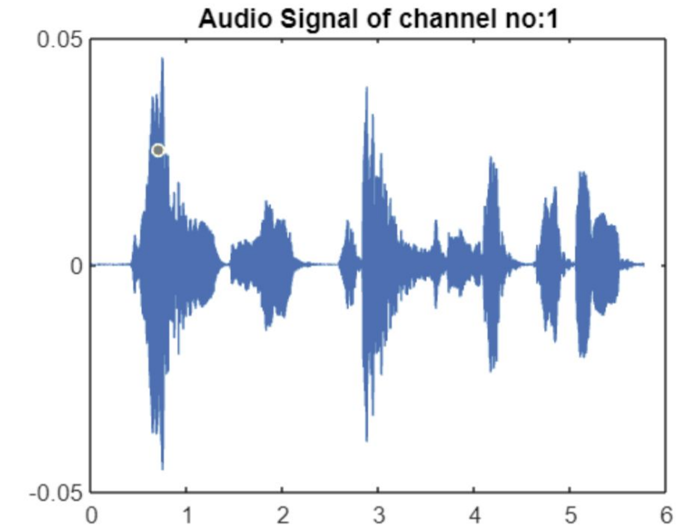

Figure 8: Plot of Gain on time domain filtered for frequencies of BPF 1

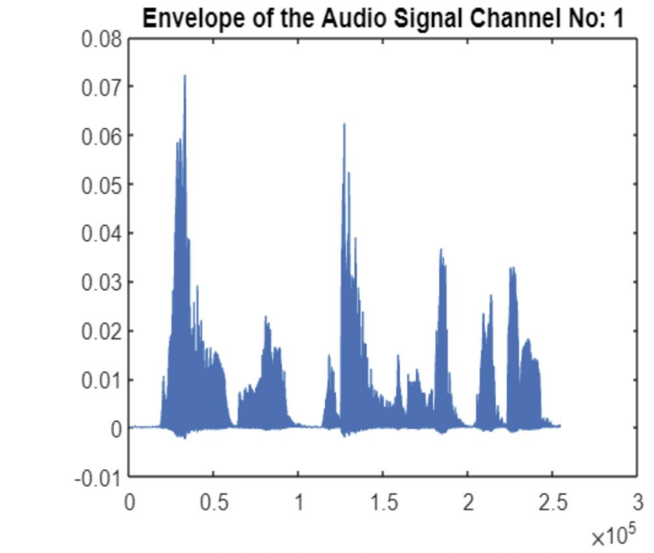

Figure 9: Plot of Temporal Envelope of Gain on time domain for channel 1

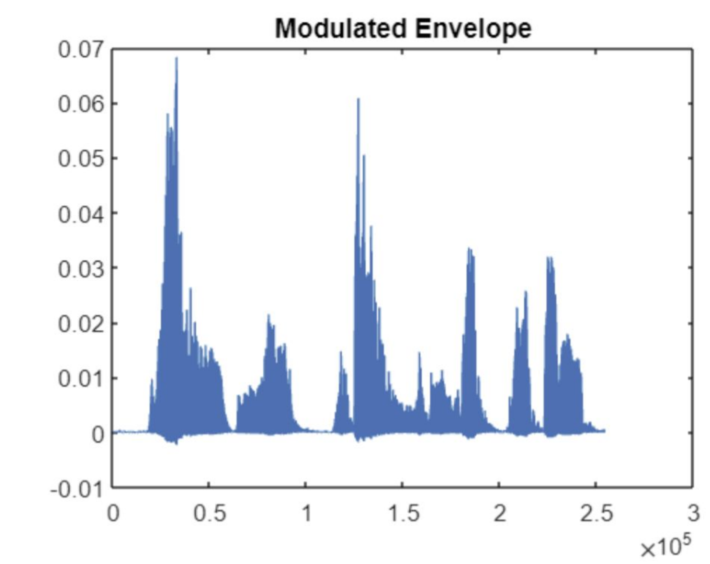

Figure 10: Modulated Envelope for Gains on Channel 1 frequencies multiplying the Temporal Envelope and White Noise Modulation

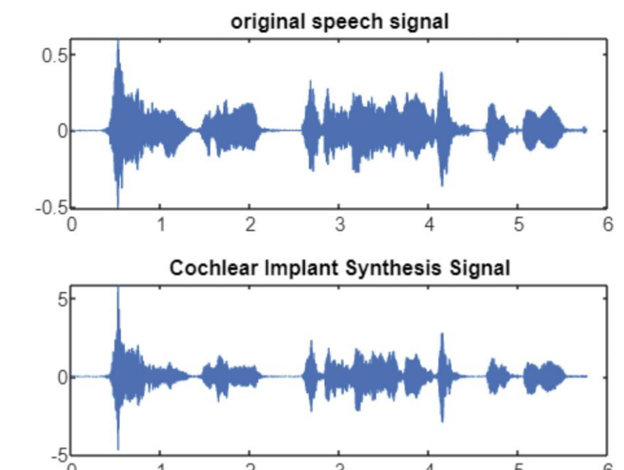

Figure 11: Comparison of original speech signal and CI synthesis signal

CONCLUSION

- Speech processor implementation was tested for comparison of original speech and synthesized speech; both were similar to each other
- Architecture implemented can be used to send the modulated noise to electrodes implanted in the cochlear fluid
- PSNR ratio of White Noise Modulation was 22 dB compared to the PSNR ratio of sin wave vocoder, which was 21 dB
- White Noise Modulation is a better approach to modulation compared to Pulse Modulation
- Speech signal was filtered with 20 Bandpass filters or channels to check for any difference in the cochlear implant synthesized signal; output was similar to the 8 bandpass channels
- 8 is an optimum number for bandpass filters

REFERENCES

Dorman MF;Loizou PC;Rainey. "Speech Intelligibility as a Function of the Number of Channels of Stimulation for Signal Processors Using Sine-Wave and Noise-Band Outputs." *The Journal of the Acoustical Society of America*, pubmed.ncbi.nlm.nih.gov/9348698/. Accessed 31 July 2023.

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