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Effects of Number of Filters and Frequency Cutoff in Continuous Interleaved Sampling and Frequency Amplitude Modulation Encoding Schemes in Cochlear Implant

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ABSTRACT- Cochlear implants are devices designed to transform sound into electrical signals perceived by the brain, making them vital prostheses for deaf individuals. This study examines two schemes used in cochlear implants, namely Continuous Interleaved Sampling (CIS) and Frequency Amplitude Modulation Encoding (FAME), to compare their performance while varying the number of bandpass filters and cutoff frequencies used. Both schemes were simulated using 8 and 5 bandpass filters, and cutoff frequencies of 2000 Hz and 200 Hz. Results show that the CIS scheme can maintain signal intelligibility despite the loss of some frequency components when the number of bandpass filters is lowered. Conversely, FAME retains more frequency details but presents perceptible delays. With a cut off frequency of 200 Hz, signals processed with CIS loses intelligibility significantly, whereas FAME-processed signals remain intelligible both at 200 Hz and 2000 Hz cut off frequencies. It is therefore concluded that FAME can provide better cochlear implant performance despite the lower number of bandpass filters and lower frequency cutoff.

Keywords: cochlear implant, continuous interleaved sampling, frequency, and amplitude modulation encoding, bandpass filter.

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1. INTRODUCTION

Cochlear implants are auditory prostheses which are widely accepted devices to assist deaf individuals in understanding sound [1]. Cochlear implants consist of two main parts, namely the external and the internal components. The external component consists of an externally charged microphone, a sound processor, and a transmitter. The microphone picks up sound from the surroundings, and the sound processor separates the sound into different frequencies and converts it into a digital signal. These signals are then sent to the transmitter, which subsequently sends them to the receiver in the internal component via a magnetic connection through the skin. The internal component is implanted under the skin behind the ear. This receiver converts the digital signal into an electric current which will subsequently be transmitted to a series of small electrodes that are surgically inserted into the cochlea. A cochlea is a spiral-shaped inner ear structure that converts sound vibrations into electrical signals, which the brain interprets as sound. The cochlea contains tiny hair cells that move in response to sound waves, initiating the conversion process. A cochlear implant is designed to mimic the function of a cochlea. The electrodes in the cochlear implant bypasses damaged hair cells by directly stimulating the auditory nerve with electrical signals, enabling the brain to interpret these signals as sound [2], [3], [4]. Several signal processing concepts are needed to model cochlear implants. The main concept is sound filtering to separate incoming sounds into different frequency bands, to mimic the human auditory process. In the process of human hearing, different parts of the cochlea or inner ear respond to different sound frequencies. By separating incoming sound into different frequency bands, cochlear implants can mimic the way the human ear processes sound. Other important concepts of signal processing used in cochlear implant modeling are envelope detection, modulation, and demodulation. Envelope detection is used to extract variations in the amplitude of the captured sound signal. In the modulation process, the carrier frequency is modulated using envelope signals, and modulated signals are transmitted from the outside of the cochlear implant to the inside.

Continuous Interleave Sampling (CIS) is a stimulation strategy for multichannel implants based on spectral analysis of digital input sound signals, performed by bandpass filter sets [5], [6], [7]. The filter bank has an overall bandwidth from 100 to 8000 Hz, and the number of filters is typically equal to the number of stimulation channels at the interface of the electrode array to the nerve. Each filter is connected to at least one intracochlear electrode according to the position-frequency tonotopic arrangement of the cochlea. The CIS scheme overcomes the channel interaction problem by using interleaved sets of non-simultaneous pulses. Pulses are sent to several electrodes such that only one electrode is stimulated at a time. In CIS scheme, the original sound to be perceived is passed through a set of

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bandpass filters, and the envelopes of all the filtered waves are extracted to produce amplitude pulses that will be sent to the electrodes [8], [9]. Amplitude extraction is carried out using full wave rectification and lowpass filtering, using a certain cutoff frequency. Several studies reported that relying on the amplitude modulated part of a speech would not fully aid speech recognition, due to limited number of filters, especially in noisy environment [7], [10], [11], [12].

Frequency Amplitude Modulation Encoding (FAME) is another technique used in cochlear implants to convert sound signals into electrical signals that can be understood by the auditory nerve [13], [14], [15]. Different from traditional cochlear implants that use Frequency Modulation (FM) coding, FAME combines information about the frequency and amplitude of sound, providing users with more detailed and nuanced hearing information. The incorporation of frequency and amplitude information allows implant users to sense a wider range of sounds and nuances in speech and environmental sounds[16], [17]. This increase in perception can improve the user's ability to understand speech and appreciate various sounds around him. A comparison of cochlear implant simulation models with CIS and FAME will be presented in this study.

2. METHODS

2.1. Simulation Model

The cochlear implant block diagram is depicted in *figure 1*, which illustrates the use of a microphone system within the external device to capture sounds. A sound processing method is then employed to adjust the spectrum shape using filters and to optimize the input dynamic range in correspondence to input signal levels [18]. The encoding process refers to the transformation of input sound signal into a pattern of electrical pulses. The electrical pulses are subsequently transmitted using radio frequency from the external device towards the internal device. In the internal device, the signals received are decoded to extract the characteristics needed to stimulate the electrode array.

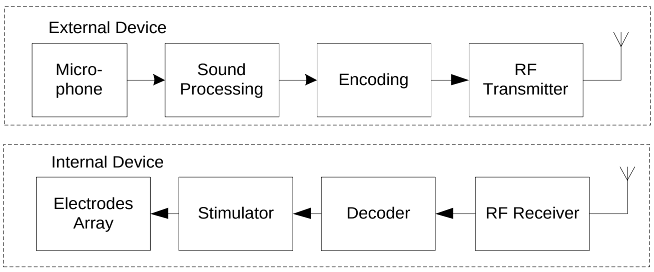

Figure 1. Block diagram for Cochlear Implant

In general, the way of presenting information to the electrodes can be divided into two types, namely analog stimulation, and digital stimulation. When information in analog form is used to stimulate the electrodes, the stimulation is called analog stimulation. The term digital stimulation is used when information is sent in the form of pulses to the electrodes [6].

The CIS schematic block diagram is illustrated in *figure 2*. The filter bank is the most complex component of CIS and require a long execution time. The frequency mapping to the cochlea is nonlinear and most of the information in the acoustic signal is in the low frequencies, so the bandwidth of the filters is distributed in such a way that narrow bands are used for low frequencies and larger bands are used for high frequencies.

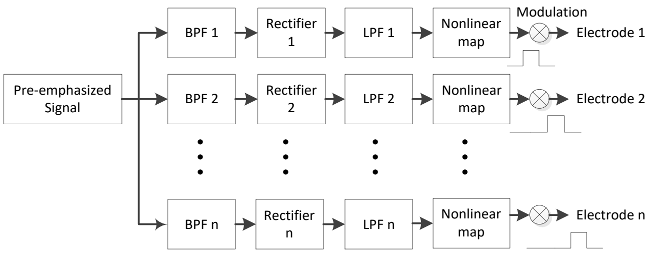

Figure 2. Block diagram for CIS [19]

The simulation model used in this research for CIS simulation is illustrated in *figure 3*. The sound signal captured by the microphone will be forwarded to a set of 8 bandpass filters. The frequency ranges used for the filters in this simulation are 50 - 450 Hz, 450-800 Hz, 800-2000 Hz, 2000–4000 Hz, 2200-4500 Hz, 2800-5100 Hz, 3000-5300 Hz, and 4000-6000 Hz. The natural frequency response of the cochlea is between 20 Hz and 20 kHz, with the lower frequencies typically are perceived better than the higher ones. Therefore, frequencies cutoff used in the subsequent simulations are 200 Hz to represent low frequency and 2000 Hz to represent high frequency.

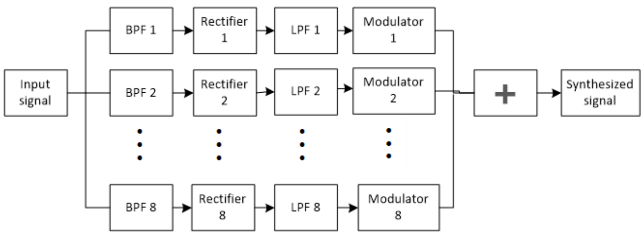

Figure 3. CIS Simulation Model

The FAME simulation model adapted from [20] is illustrated in *figure 4*. The input signal is fed into a bank of bandpass filters, which frequency ranges are the same as the ones used for the CIS simulation. The output of one bandpass filter will be processed for amplitude modulation and demodulation, as well as frequency modulation and demodulation concurrently. The resulting signals from the AM and FM processes are combined to produce a synthesized signal.

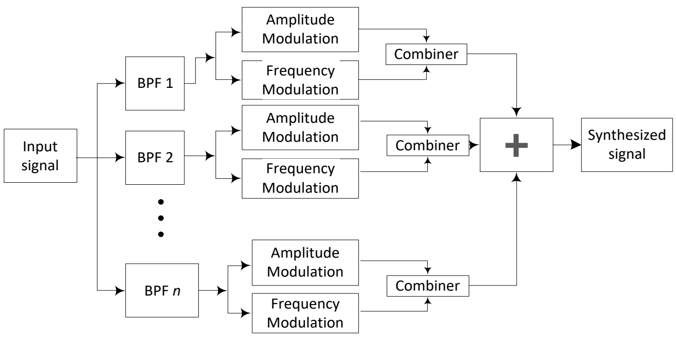

Figure 4. FAME Simulation Model

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The frequency deviation used in the frequency modulator for this study is 50 Hz, while the carrier frequency is 12 kHz. Both the frequency deviation and carrier frequency must be selected carefully, and in real application must be fitted to the user of cochlear implant. The frequency deviation is chosen in this study to ensure the intensity of sounds is within optimal auditory range. The carrier frequency is chosen to ensure the different pitches in the speaker's voice can be perceived.

The simulations were programmed and executed on two computers, which specifications are given in *table 1*.

Table 1. Specifications of computers used for simulations

	Computer 1	Computer 2
Operating System	Windows 11	Windows 10
Processor	Intel i7-1165G7	intel i3-6000U
Base clock-speed	2.80 GHz	2.00GHz
RAM	8 GB	4 GB

The input signals are pre-recorded voice samples, as described in *table 2*.

Table 2. Voice samples used in simulations

Label	Speaker	File Type	File Size (KB)
Sample A	Female	WAV	304
Sample B	Female	WAV	742
Sample C	Male	WAV	384
Sample D	Male	WAV	531

Samples A and B are pre-recorded adult female voice, uttering a short phrase "Good Morning". The waveforms of *sample A* in the time domain are shown in *figure 5*. The original signal is then processed using bandpass filters and only the envelopes of filtered signals are used in the CIS scheme, so that the synthesized signals lack certain frequency components present in the original signals. In cochlear implant users, this signal synthesis occurs within the central nerves. *Sample B* is a longer utterance compared to *sample A*, and underwent the same process as *sample A*. The waveforms of sample A in the time domain are shown in *figure 6*.

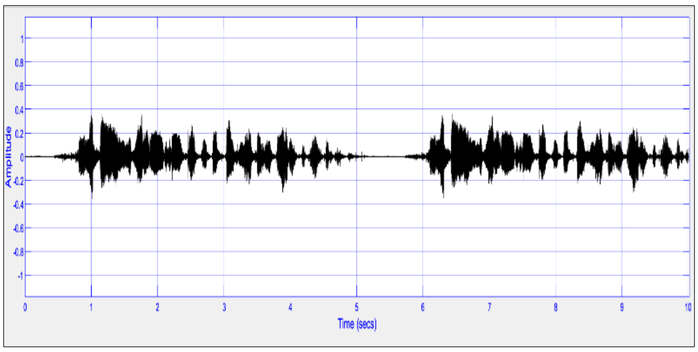

Figure 5. Original voice – Sample A

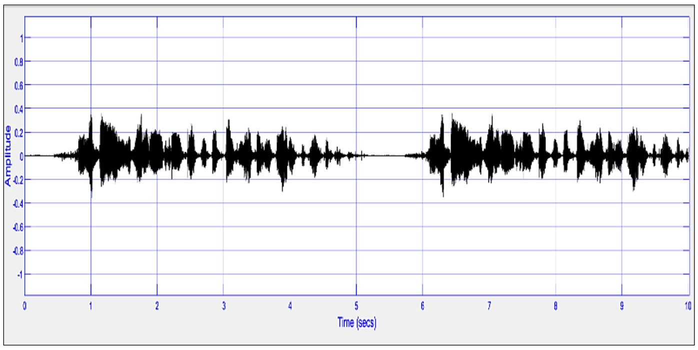

Figure 6. Original voice – Sample B

Samples C and D are both pre-recorded male voices, consisting of a short phrase and a long sentence, consecutively. The original waveforms of samples C and D are given in *figures 7* and *figure 8*, subsequently.

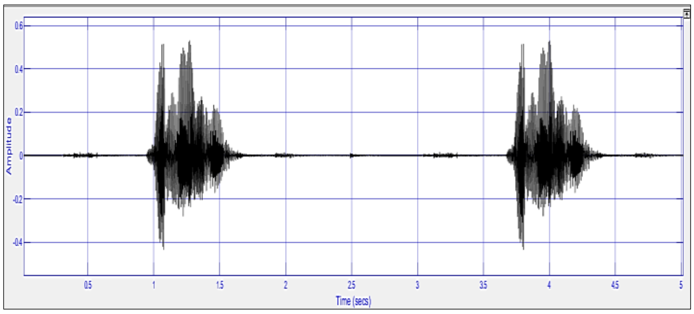

Figure 7. Original voice – Sample C

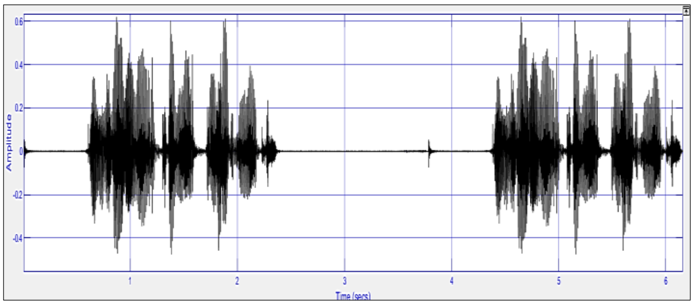

Figure 8. Original voice – Sample D

3. RESULTS AND DISCUSSION

3.1. Effects of the number of filters for CIS and FAME Schemes

3.1.1. Results of Simulations Using 8 Filters

The effects of the number of filters for cochlear implants were observed by using 8 and 5 bandpass filters for the CIS and FAME schemes using all voice samples described in the previous section. The frequency cutoff for these simulations is 2000 Hz which is considerably higher than the frequency cut-off typically used for cochlear implants (200 – 400 Hz). *Figure 9* and *figure 12* shows samples A, B, C and D after being processed with 8 filters, with both CIS and FAME schemes.

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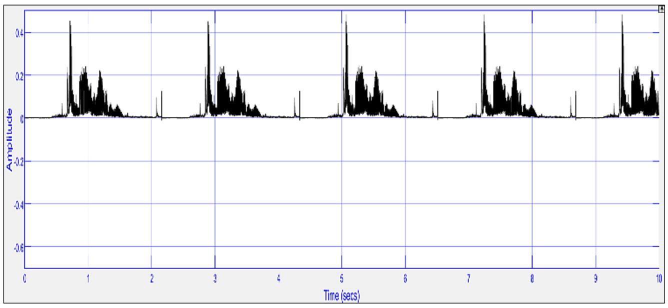
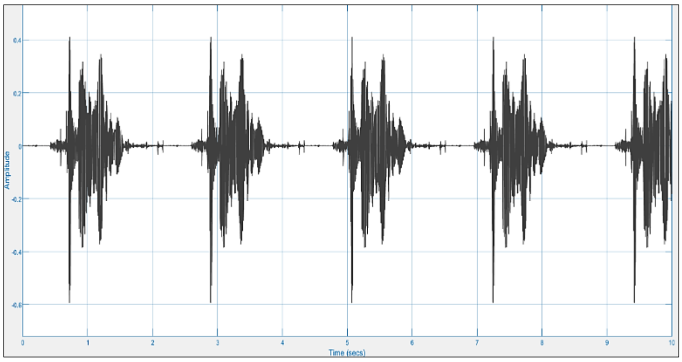
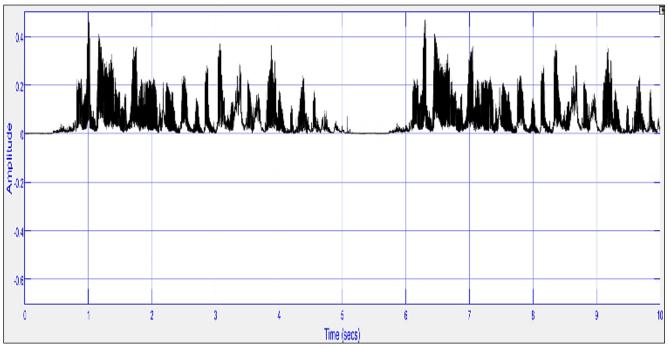


(a)



(b)

Figure 9. (a) Sample A processed with CIS scheme, 8 filters (b) Sample A processed with FAME scheme, 8 filters



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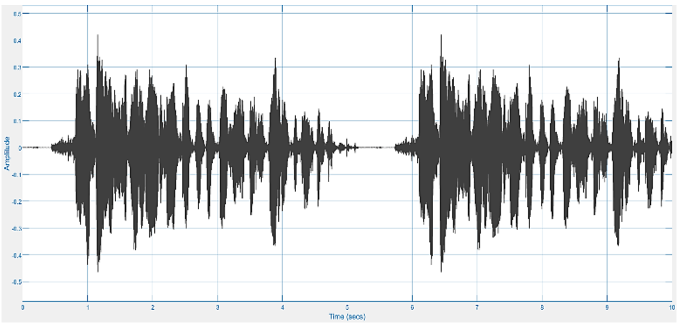
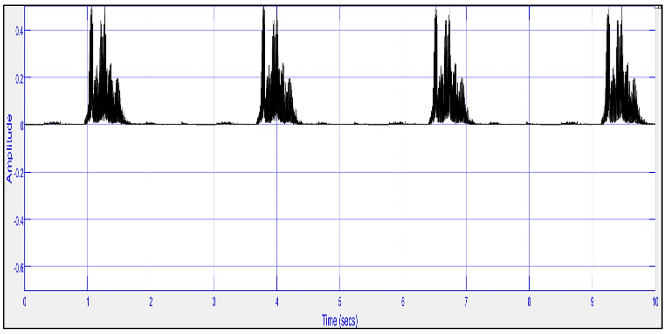


(b)

Figure 10. (a) Sample B processed with CIS scheme, 8 filters (b) Sample B processed with FAME scheme, 8 filters



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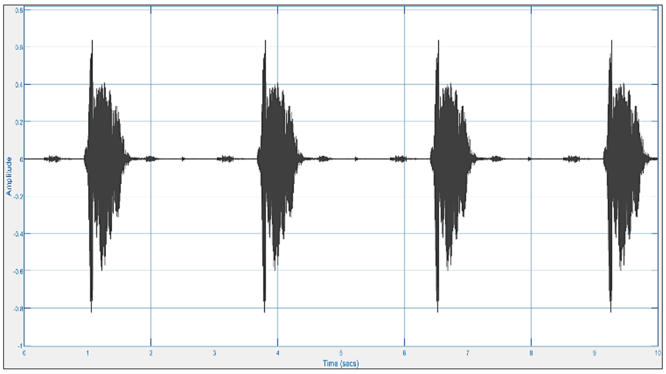
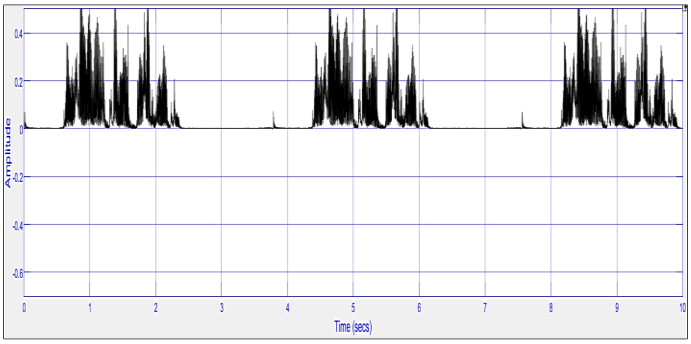
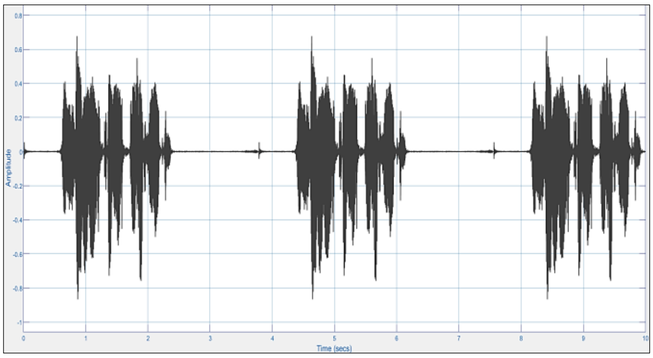


(b)

Figure 11. (a) Sample C processed with CIS scheme, 8 filters (b) Sample C processed with FAME scheme, 8 filters



(a)



(b)

Figure 12. (a) Sample D processed with CIS scheme, 8 filters (b) Sample D processed with FAME scheme, 8 filters

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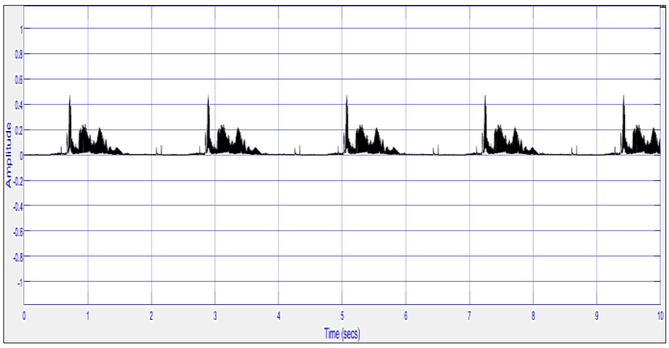
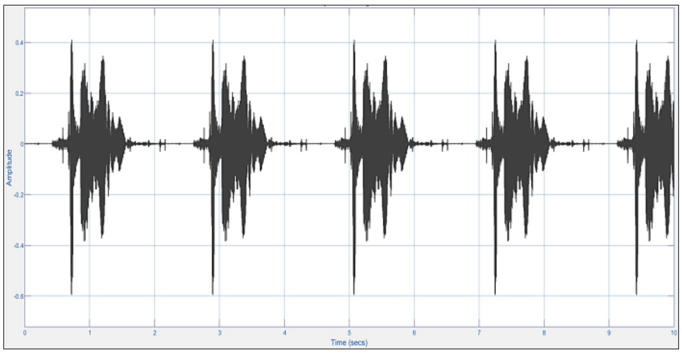
Using the CIS scheme, some frequency components are missing as full wave rectifiers are used to extract the envelope of the original voice signal. However, the synthesized voice signals of samples A, B, C and D are all intelligible. From *figures 9- figure 12*, it can also be observed that the frequency components that are lost when the original voice signals are processed using CIS, can still be preserved when the FAME scheme is employed. The voice signals processed with FAME scheme are also intelligible. However, there is a perceptible delay in the signals processed with FAME. This is because in the FAME scheme, the signal extraction consists of three stages. The first stage is the separation of filtered signal to be processed with amplitude modulation and frequency modulation. This stage is then followed by combining the signals produced by the amplitude modulation and frequency modulation processes for each filter, and lastly, the signals are combined to construct the synthesized signal. This delay is minor and should be assessed for real-time applications. One research has investigated interaural delay and found that cochlear implant users show less sensitivity to this delayed compared to normal-hearing individuals [21].

3.1.2. Results of Simulations Using 5 Filters

The simulation scenario for CIS and FAME schemes using 5 filters is like that provided in the previous section. The frequency ranges used in the following simulations are 50 - 450 Hz, 450-800 Hz, 800-2000 Hz, 2000–4000 Hz, 2200-4500 Hz. Figure 13-16 shows samples A, B, C and D after being processed with 5 filters, with both CIS and FAME schemes.

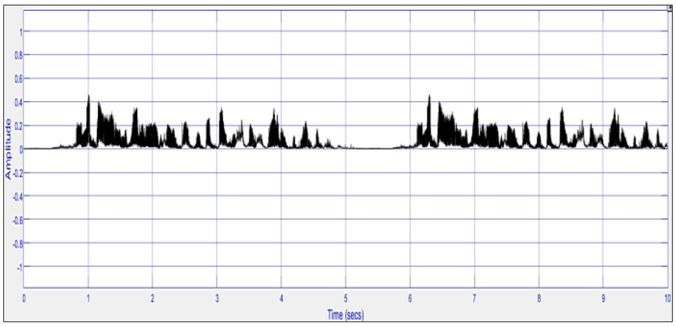


(a)

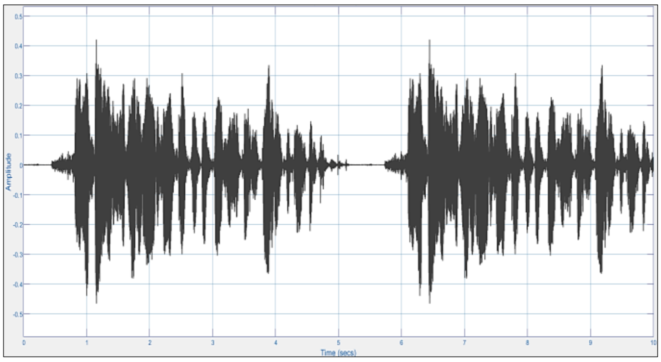


(b)

Figure 13. (a) Sample A processed with CIS scheme, 5 filters (b) Sample A processed with FAME scheme, 5 filters

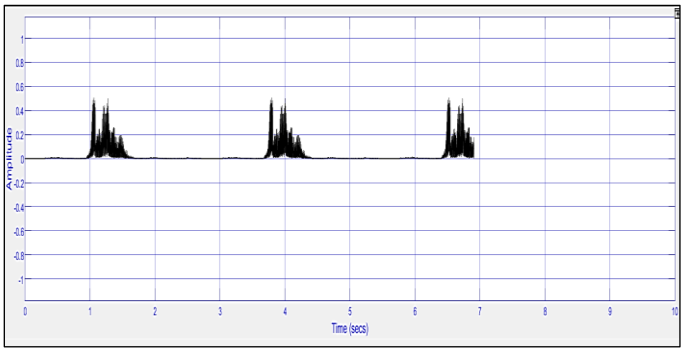


(a)

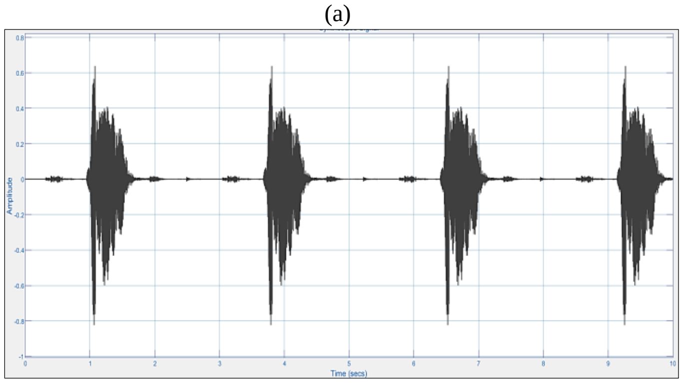


(b)

Figure 14. (a) Sample B processed with CIS scheme, 5 filters (b) Sample B processed with FAME scheme, 5 filters

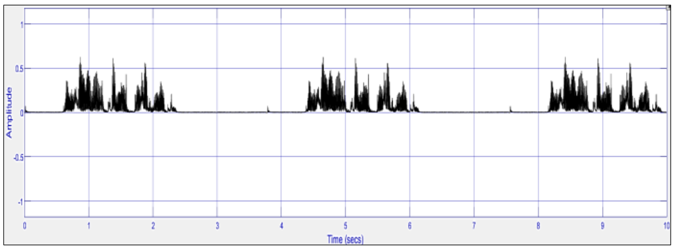


(a)

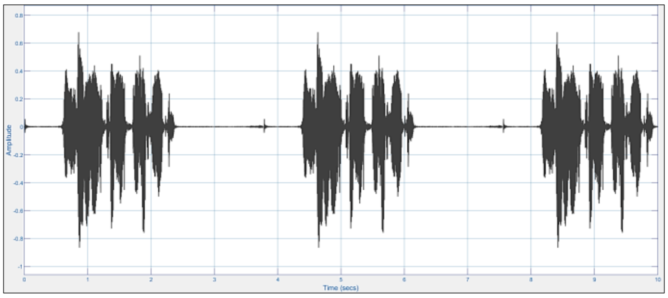


(b)

Figure 15. (a) Sample C processed with CIS scheme, 5 filters (b) Sample C processed with FAME scheme, 5 filters



(a)



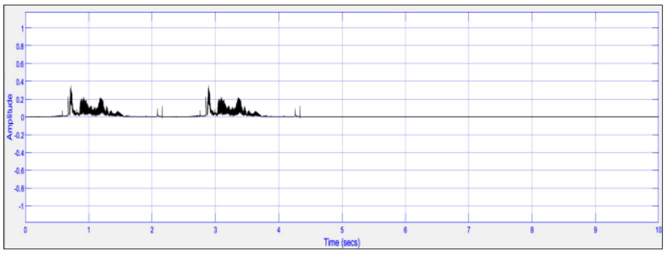
(b)

Figure 16. (a) Sample D processed with CIS scheme, 5 filters (b) Sample D processed with FAME scheme, 5 filters

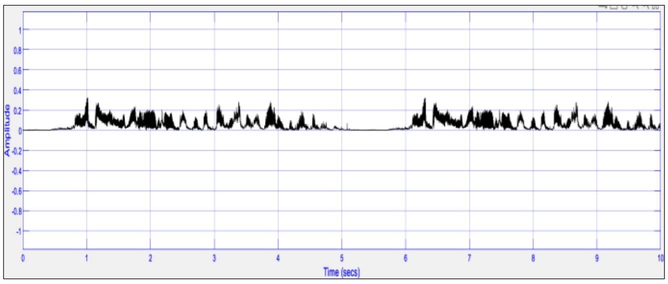
The voice samples processed with 5 filters are intelligible. The use of 5 filters instead of 8 allows the simplification of the cochlear implant design. However, it is noted that the use of 5 filters produce voice samples which sound unnatural and almost monotone compared to the use of 8 filters.

3.2. Effects of frequency cutoff for CIS and FAME Schemes

Simulation results using Low Pass Filters (LPF) with cutoff frequency (f_c) = 200 Hz are given in *figures. 17(a)-figure 17(d)* for cochlear implant model that uses CIS scheme with 8 filters.



(a)



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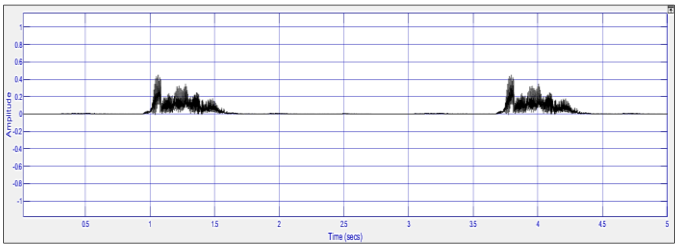

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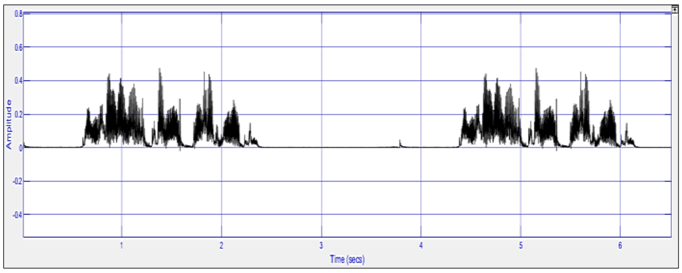
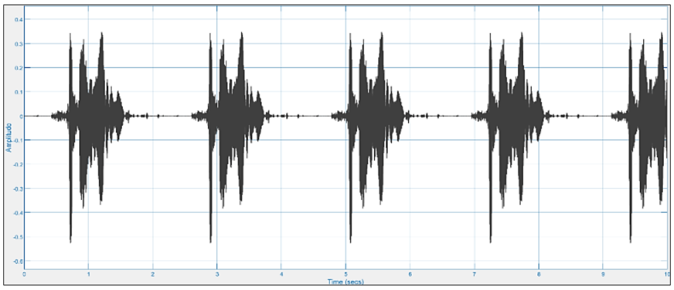


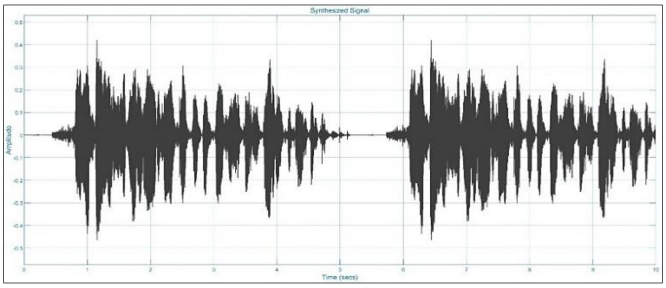
(d)

Figure 17. Processed signals with CIS scheme, fc = 200 Hz (a) Sample A, (b) Sample B, (c) Sample C, (d) Sample D

The simulation results when FAME scheme is used and the number of filters is 8 are given in *figures. 18(a) –figure 18(d).*



(a)



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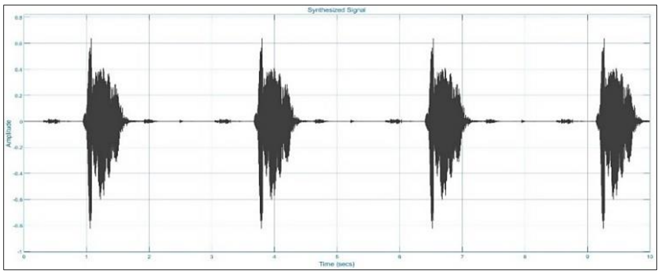

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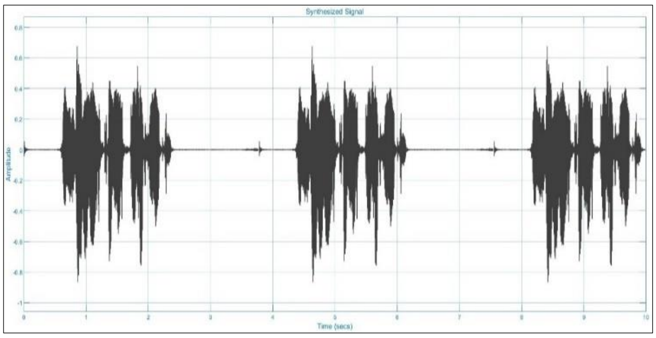

(d)

Figure 18. Processed signals with FAME scheme, fc = 200 Hz (a) Sample A, (b) Sample C, (c) Sample C, (d) Sample D

For CIS scheme, by comparing *figure 17 (a)-figure 17(d)* to *figures. 9(a), figure 10(a), figure 11(a) and figure12(a)* where fc is 2000 Hz, it is apparent that the use of low fc results in more frequency components being lost. The most affected frequency component is at time t = 1.5 to 2 s depicted in *figure 17(a)*, where almost no frequency component is detected. The synthesized sound signal with fc 200 Hz becomes significantly less intelligible than when fc = 2000 Hz. The most noticeable distinction observed when utilizing cutoff frequencies of 2000 Hz and 200 Hz is in the amplitude of the processed signal. Sound samples processed with lower cutoff frequencies generally exhibit reduced signal amplitudes, due to the attenuation of certain signal components.

With FAME scheme and fc = 200 Hz, it is observed that the spectrums of the processed voice samples depicted in *figures 18(a)- figure 18(d)* are very similar to the those of the processed voice samples with fc = 2000 Hz depicted in *figures. 10(b), figure 11(b), figure 12(b), and figure 13(b)*. With the FAME scheme, the use of fc = 200 Hz provides intelligible sounds which are comparable to the sounds resulting from the use of fc = 2000 Hz. As the use of low fc is preferable in cochlear implants, the FAME scheme is therefore having an advantage compared to the CIS scheme.

4. CONCLUSIONS

The CIS and FAME schemes for cochlear implants have been simulated with bandpass filters banks containing 8 and 5 filters. Using 5 filters instead of 8 in processing voice samples results in intelligible yet unnaturally monotone samples, simplifying cochlear implant design but sacrificing naturalness. The CIS scheme, employing full wave rectifiers for envelope extraction, omits some frequency components, yet maintains intelligibility in the synthesized voice signals. In contrast, the FAME scheme, while preserving lost frequency components, introduces a noticeable delay in the processed signals. However, when low frequency cutoff is used, the FAME scheme outperforms the CIS scheme in terms of intelligibility. Therefore, cochlear implant designs should take into consideration a FAME scheme which is refined to the specific requirements or physical conditions of the users. The choice of CIS or FAME schemes should remain open and the chosen scheme should be optimized to the auditory needs of the user.

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