

Continuous Interleaved Sampled (CIS) Signal Processing Strategy for Cochlear Implants MATLAB Simulation Program

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Abstract: The application of cochlear implants in the correction of hearing and speech impediments in deaf persons is known more than 25 years. Although there are so many speech processing strategies, it is necessary to study the performance of the high-rate continuous interleaved sampling (CIS) coding strategy. Research processors utilizing CIS. For fast sampling and responses the FIR filters are used. The total frequency range of audible signals will divide into individual frequency components and to make respond the basilar membrane (cochlea - hair cells) in the inner ear. Research shows, each place of the basilar membrane in the cochlea responds greatly to the frequency where they are most responsive. In order to implement this process, I have to develop the mat lab program for Cochlear implantation by using CIS speech processing strategy.

Index terms —cochlear implant, CIS, Compressed-Analog (CA), Chebyshev II filter, basilar membrane, graphical user interface (GUI),maximum loudness level (MAL).

1. INTRODUCTION

There are approximately 3 million people in the US with hearing loss. The range of loss can vary from mild to profound. Some people with a severe to profound hearing loss wear an ear prosthesis device known as a cochlear implant. The cochlear implant is one of most innovative inventions developed in the field of Biomedical Engineering. This hearing device plays a vital role in restoring partial hearing to people with sensorineural hearing loss. For unknown reasons, some Users are far more successful than others. In order to understand my proposal for the project, I will begin with the background that led to the development of cochlear implants and their signal processing strategies.

2. NORMAL HEARING

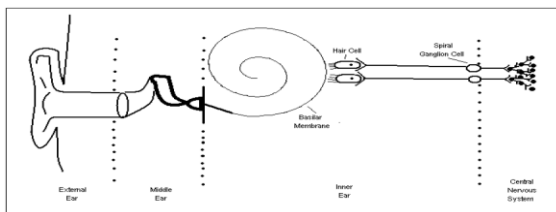


Figure 1. A diagram of the human ear

Figure 1 shows a diagram of the human ear comprising of the outer, middle and inner ear. When sound travels to the ear, it moves through a series of transformations as it passes through the outer ear, middle ear, inner ear, auditory nerve and brain. Sound is acoustic pressure caused by abrupt changes in the air pressure. As sound enters the ear, the outer ear gathers sound in the ear canal. A series of small bones in the inner ear then responds to the incoming sound by vibrating itself. In fact, the middle ear converts sound into mechanical vibrations. The cochlea, a snail-shaped cavity filled with fluid in the inner ear, then transforms these mechanical vibrations to vibrations in fluid. Pressure differences within the fluids of the cochlea lead to fluid displacement of basilar membrane. These displacements contain significant information about the frequency of the sound signal. Hair cells that attach the basilar membrane are bent according to the change of position in the basilar membrane. The bending of the hair cells causes neurons to fire by releasing an electrochemical substance. These neurons transmit information about the sound signal to the brain. The brain detects the excitation in the auditory nerve and then interprets the sound.

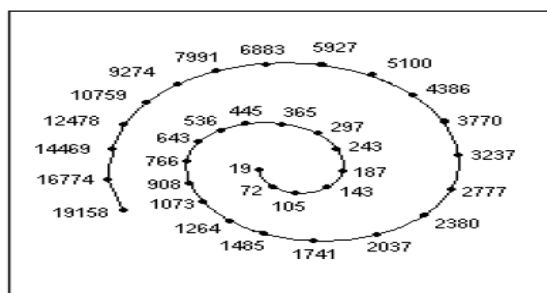
3. HEARING LOSS

The bending movements of the hair cells in the vicinity of the basilar membrane are responsible for converting mechanical vibrations into electrical signals. If the hair cells are damaged, the auditory system is unable to properly convert sound into electrical impulses in the nerve. Because

of this, the result is hearing impairment. Even if an audiogram looks “normal,” there may be hearing loss if the person cannot understand spoken language. Simply, there is a broken link in the whole process. The acoustic wave is able to travel through the outer ear, middle ear and inner ear. However, the wave is unable to reach the auditory nerve and the brain because of the damaged hair cells. There are various types of diseases, drug treatments and other causes that contribute to degeneration in the hair cells. In majority of cases and most research, the common cause of hearing loss is damage to the hair cells, but not the auditory nerve. This problem has led to the development of cochlear implants. Cochlear implants will be explained in detail later. It is more important to describe the common cause of hearing loss that led to the invention of cochlear implants. If a large number of hair cells of a particular person are damaged, the person is diagnosed as having a profound hearing loss.

4. FUNCTION OF COCHLEA

Before explaining cochlear implants, I'd like to elaborate more on how the cochlea functions. Since the basilar



membrane detects the change in fluid of

Figure 2. Diagram of the basilar membrane showing the position of maximum displacement in response to different frequency (Hertz).

the Cochlea due to various frequency of acoustic waves, this fact leads to the basic question, “How the whole length of the basilar membrane does picks up frequencies of the sound wave?” The work by Georg von Bekesy in the 1950’s discovered that the basilar membrane in the cochlea is responsible for detecting and analyzing the input signal into different frequencies. Different frequencies cause maximum vibration rate at different points in the basilar membrane (Figure 2). Low-frequency sounds generate roaming waves in the fluids of the cochlea that cause the basilar membrane to vibrate with largest amplitude of displacement at the apex (farthest end of the basilar

membrane). On the opposite side, high frequency sounds produce roaming wave with largest amplitude of displacement at the base (beginning of the snail-like line) of the basilar membrane. If the sound signal is composed of many different frequencies, then the resulting wave will generate maximum displacement at different points along the basilar membrane. [1] In fact, the cochlea picks up sound waves and separates them into many different frequency components. The cochlea, as a part of the auditory system, is responsible for encoding frequencies of acoustic sound signal. The traveling wave of the basilar membrane in the cochlea depends largely on the frequency of the stimulation. The hair cells attaching to the basilar membrane bends by the displacement of the stimulation. The hair cells are organized based on the frequency where they are most sensitive. Each place of the basilar membrane in the cochlea responds greatly to the frequency where they are most responsive. This determination of placement for frequency is known as place theory. Another theory, known as volley theory, states that frequency is determined by the rate at which the neurons are fired. Simply stated, the auditory neurons fire at rates that are proportional to the period of the input signal of all available frequency for sounds. Concepts of how the ear functions and the main causes of hearing loss build solid foundations for new developments of cochlear implants.

5. COCHLEAR IMPLANTS

Cochlear implants have been developed over the years to provide partial hearing to people with a severe to profound hearing loss. Figure 3 shows a diagram of a typical cochlear implant. The basic functions of cochlear implants are shown: a microphone picks up the sound, a speech processor (also known as signal processor) that transforms the sound into electrical signals, a system consisting of

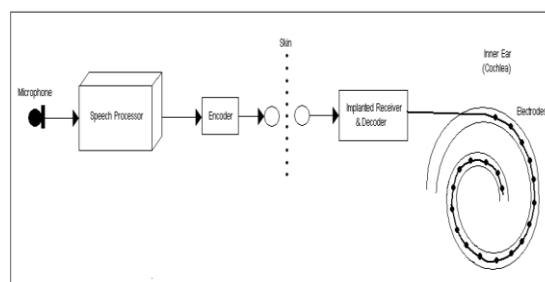


Figure 3. A diagram of a typical cochlear implant.

encoder, receiver and decoder that transmits the electrical signals to the implanted electrodes, and an electrode array that stimulates the auditory nerve in order to transmit

electrical signal to the brain. The implant comprised of receiver, decoder and electrodes are inserted inside the skin above the ear by a surgeon. There are two different designs of implants: single-channel implants and multichannel implants. The multi-channel design is found in all newer cochlear implants. Figure 2 shows different positions for maximum displacement of frequency in the cochlea. The single-channel implant merely produces electrical stimulations from sounds of all frequency in one electrode. Unlike the single-channel, multichannel implants are designed to address these differences of frequency in different locations of the cochlea. Since the main function of electrodes is stimulation of the auditory nerve, the speech processor produces different levels of frequency for each electrode in the cochlea. Frequency is separated in sequence from maximum possible frequency in the base of the cochlea to lowest possible frequency in the cochlea as seen in Figure 3 where there are different electrodes in the cochlea. The most important component of the cochlear implant is the signal-processing strategy used for transforming the speech signal to electrical signal. Various signal processing techniques have been developed over the past several decades. Some of these techniques are aimed at maintaining waveform information, others are aimed at envelope information, and others are for spectral features. The scope of this proposal will focus on waveform and envelope information. CIS (Continuous Interleaved Sampling) strategy is an example of envelope information. Before discussing CIS strategy, I want to introduce what led to the development of the CIS strategy. It is known as Compressed-Analog (CA) Approach. CA approach actually separates the sound into different frequencies. The signal in this strategy is based on waveform information. There are many publications that show huge improvement in speech understanding with the multi-channel over the single channel implant. However, there is an unresolved issue. Channel interaction occurs in electrodes used for stimulation by the CA strategy. If two electrodes are stimulated at the same time, each electrode produces an electromagnetic field around itself. Based on the laws of physics, current flow in the cable produces an electromagnetic field around the cable. Thus, if the two electrodes that are next to each other are stimulated, the electromagnetic field produced by each electrode actually

interacts with the other. This interaction in electromagnetic field causes distortion in the sound signal, impeding the progress in nerve response to a certain sound. This effect results in reduction of speech intelligibility. Because of the channel interaction, CIS strategy was developed to solve this issue.

6. SIGNAL PROCESSING OF CIS STRATEGY

CIS (Continuous Interleaved Sampling) strategy is one of signal processing techniques on preserving waveform information for cochlear implants. Figure 4 shows the diagram of the whole process of CIS strategy.

How does CIS strategy process sound signals?

When a person is talking, his voice is picked up by the microphone. The microphone converts sound waves into electrical signals. The signal is first amplified before going into a “box” of band pass filters. Band pass filters then separate sound signals into different frequencies. After passing through the band pass filters, full wave rectifiers and low-pass filters (typically with 200 or 400 Hz cutoff frequency) extract the filtered waveforms into “envelopes.” Then, the envelope outputs are compressed. The compression is a nonlinear function. It is used to verify that the envelope outputs fit the dynamic range of electrically evoked hearing based on patient’s preference. After the compression, the signals are used to modulate biphasic pulses. A diagram of these pulses is shown in Figure 5. The amplitudes of the pulses are proportional to the envelopes.

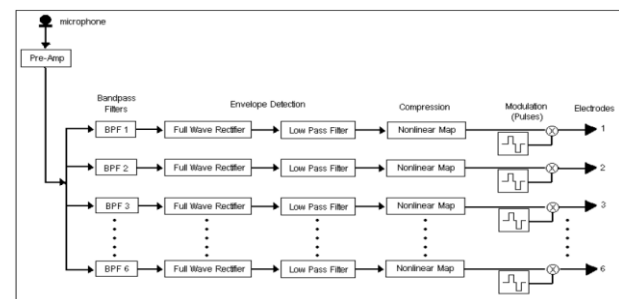


Figure 4. Diagram of the Continuous Interleaved Sampling (CIS) strategy.

They are delivered to electrodes in a non-overlapping fashion as seen in Figure5. In many studies, high pulse-rate stimulation yields better performance than low pulse-rate stimulation. This is the basic concept of the CIS strategy.

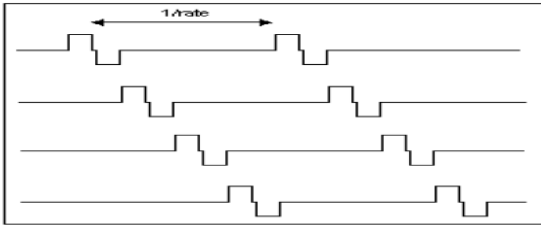


Figure 5. Interleaved Pulses found in CIS strategy.

The MATLAB simulation program shows how the CIS strategy functions.

My Objectives

- Use the MATLAB program to simulate the signal processing strategies
- Use simulation to understand how sound signals are manipulated and processed from the original sound form to the final signal that stimulates electrodes
- Long Term Objective: use MATLAB simulation to study the neural and electrical responses produced this senior thesis is only a small part of my long-term objective. Using the program that shows pictorial graphs of CIS strategy signal processing can help me to work towards an ideal goal of one hundred percent understanding of speech.

7. MATLAB SIMULATION PROGRAM

A wave file of sounds is necessary to make the program work. Anyone can create a new wave file with his or her own sound. I'm using the sound of counting from 1 to 5 as an example for the MATLAB simulation. The transcript of the sound file is "OneFTwoFThreeFFourFFive." This is the sound file that used as a sample for the output from the microphone. The microphone simply picks up sound which is the abrupt change in air pressure, and then converts to an electrical signal that looks like a wide waveform in Figure 6. In fact, speech consists of vibrations produced in the vocal tract. The vibrations themselves can be represented by speech waveforms as seen in Figure 6. However, it is not possible to read the phonemes in a waveform, but if we analyze the waveform into its frequency components, we obtain a spectrogram which can be deciphered. The spectrogram is shown in Figure 6. It shows how much of frequency of the waveform is used. The vertical axis represents frequencies up to 20,000 Hz, the horizontal axis shows positive time toward the right, and the colors represent the most important acoustic peaks for a given time frame, with red representing the highest energies, then

in decreasing order of importance, orange, yellow, green, cyan, blue, and magenta, with gray areas having even less energy and white areas below a threshold decibel level.

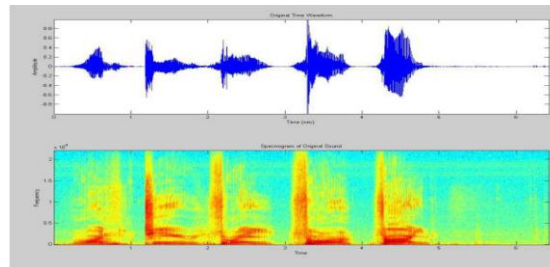


Figure 6. Waveform and spectrogram of the original sound.

Since the cochlea comprised of different frequency placements for vibrations to be transmitted, a waveform filtering system is needed to mimic this frequency placement effect. The waveform filters can be distinguished by different frequency ranges. Since the frequency range is established, a filter design is needed to transform the waveform into different frequency ranges. Chebyshev II filter is the technique used for the band pass filters. Band pass filters bank filters the waveform into channels of different frequency ranges. In the MATLAB program, the code for this filter design can be seen as follows:

$$[b1, a1]=cheby2(3, 40,[50/d 450/d]); \quad (1)$$

This code designs an 3th order band pass digital Chebyshev II filter with the stop band ripple 40 decibels down and stop band edge frequency range from 50 Hz to 450 Hz. In order to show the plot of the frequency spectrum of different channels, a transforms is necessary in order to convert the band pass filters to the frequency domain. I have added the following MATLAB code for each channel:

$$h1 = freqz(b1, a1, 1024); \quad (2)$$

This code produces the output of 1024-point complex frequency response. It is necessary to convert to the frequency domain from the time domain. The graph of the band pass filters are shown on Figure 7. It is a plot of Frequency Spectrum for all 10 channels. It should allow users to see how wide each channel is and verify the gap

between channels for the most optimal delivery of waveform to each electrode.

Then, the program takes out the filter of the sound using the following MATLAB code:

```
out1=filter (b1, a1, [sound file]);
```

The filtered waveform is shown on Figure 8. The sound is played out from each channel in order to see what the waveform looks like as well as hear what it sounds like from each channel. In fact, the lowest frequency goes through the first channel whereas the highest frequency goes through the last channel. In the electrode design, the highest frequency goes to the base of the cochlea while the lowest frequency goes to the apex of the electrode.

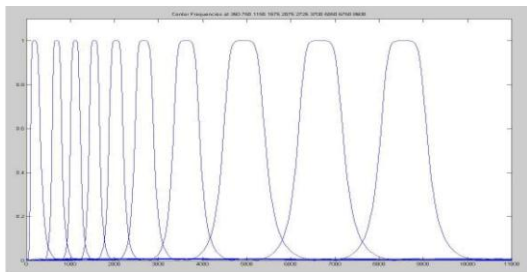


Figure 7. Graph of the 3rd order Chebyshev II filters with the stop band ripple 40 dB down.

In order to transform the waveform into envelope signal, the next step from the filtered sounds is applying full wave rectifiers and low pass filters with the cut off frequency at 200 Hz for each channel.

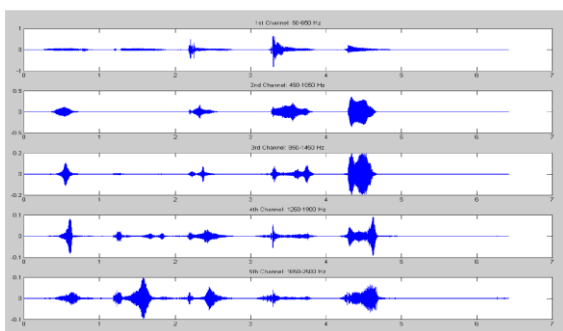


Figure 8. Waveform of the filtered sounds from each channel of different frequency range.

This is done by putting absolute value on each channel of the filtered waveform. The absolute value actually filters out all of waveform that goes below the horizontal axis. Therefore, there will be only waveform shown on the positive axis. This is known as full wave rectifier. Once

each channel is rectified, it is filtered with low pass filter to generate envelope signals. The MATLAB codes for this process are shown as follows:

```
[blow, alow]=cheby2 (8, 30,200/d); (3)
```

This code creates an 8th order low pass Chebyshev II filter with cut off frequency at 200 Hertz in order to generate envelope signals.

```
rou1=abs (out1); (4)
```

This code puts absolute value on the waveform for full wave rectifier.

```
low1=filter (blow, alow, rou1); (5)
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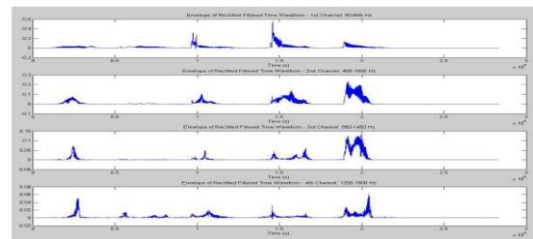


Figure 9. Envelope of Rectified Filtered Time Waveform from each channel of different frequency range. In this figure, they are the display of envelope signals through 5 channels range from 50 Hz to 1900 Hz.

This code actually filters the rectified signal with the low pass filter code as shown above. The resulting envelope signals from these codes are seen on Figure 9 and 10. Envelope is a line that traces over the positive peaks of the waveform.

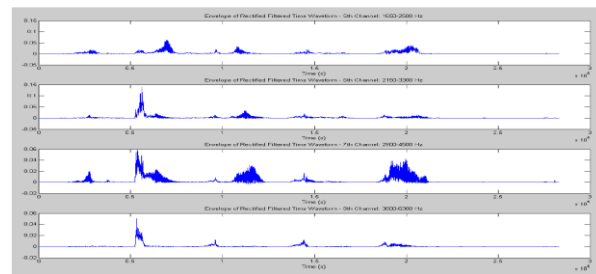


Figure 10. Envelope of Rectified Filtered Time Waveform from each channel of different frequency range. In this figure, they are the display of envelope signals through 4 channels range from 1650 Hz to 6300Hz.

As shown in the last part of the block diagram, a biphasic pulse train generator is used to modulate with the envelope to produce the desired output for simulation of electrodes.

The pulse train generator is comprised of a series of numerous biphasic pulses of the same size and rate. The pulse rate depends on the size of the sampling rate. The pulse duration is the maximum length of pulse train divided by sampling rate of the wave file, which is 44,100 Hertz. With a series of numerous biphasic pulses, they are modulated with the envelope signal in each channel. As a result, the size of the peak of each pulse depends on the maximum peak of the envelope at a certain time. Figure 11 show good example of how biphasic pulses are modulated with the envelope signals.

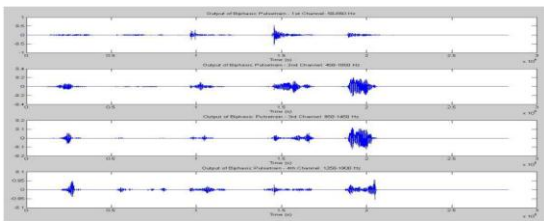


Figure 11. Modulation of Pulse train and envelope for stimulation of electrodes.

The resulting output from Figure 11 is the actual signals that go to electrodes for simulation of the auditory nerve. This simulation is the summary of how the whole CIS signal processing strategy works. The MATLAB programming code used for this project is in Appendix A on the back of the page. Discussion and Issues with the MATLAB Simulation I was able to create most of features of CIS signal processing strategies. However, there are several issues that need to be addressed in order to have a complete functional MATLAB simulation of the CIS signal processing strategy. Originally, I proposed to include a compression to each channel to meet the dynamic range chosen by the cochlear implant user. Due to time constraints, I did not include the compression to the MATLAB program. When the time comes when I can add the compressions, the program will compress the envelope signal by using the equation of power-law functions as shown below:

$$Y=A.xp+B,(P<1) \tag{6}$$

$$A=(MAL-THR)/(Xpmax-Xpmin) \tag{7}$$

$$B=THR-A Xpmin \tag{8}$$

THR= Threshold

MAL=Maximum Acceptable Level

Due to limitations of the dynamic range, the signal must be compressed that ensures the original signal is going through the full range without cutoff at peaks. The program will show the waveform graph as a result of compression. It will enable me to compare the signal between the original waveform and the compressed waveform. In order to maximize the function of the compressions, a graphical user interface (GUI) will be used to setup a dynamic range for channel. Thus, the GUI will be used to help user to find out the dynamic range between threshold and the maximum loudness level (MAL). The threshold is the minimum step of current level that is audible enough for cochlear implant user to hear. On the other hand, MAL is the maximum step of current level that is acceptable to the cochlear implant user. The GUI and dynamic range set up will be added as well. Since the specification for the CIS strategy calls for pulse train to be interleaved.

Compare Figure 5 on page 7 and Figure 12 below. In Figure 12, note at 1.43rd second in each channel, a single pulse train is shown at the same time in all channel. This is a clear example of non-interleaved pulses compared to interleaved pulse sequence in Figure 5. This problem is due to a single fixed pulse train generator in the program. The pulse train generates a same single pulse train at the same time for each channel. Due to time constraints, I did not attempt to solve this issue. However, I intend to come up with a solution in the near future.

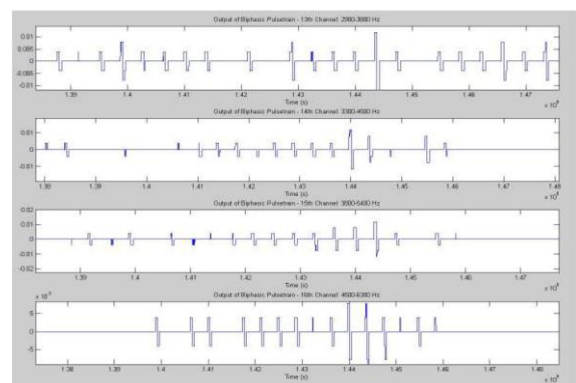


Figure 12. Close up version of the Biphasic pulse train of each channel. Notice that the biphasic pulse is generated at the same time in all channels.

8 CONCLUSION

Conclusion and Pathways to the Future It is hoped that this project greatly contributes to the field of cochlear implants, as well as my future research in signal processing. Understanding the whole signal processing technique and

neural responses will lead to development of an ideal signal processing technique that will hopefully give users one hundred percent understanding of speech, music and environmental sounds in all situations. The goal is that the cochlear implant user is able to "hear" the same as those with normal hearing. Besides the mentioned necessities to fix the MATLAB program for the CIS signal processing strategies, there are several things that I would like to do with this MATLAB program in the future:

Add SPEAK and ACE coding strategies, and other available strategies used for cochlear implants to the program. Add and expand the neural response model program and study how nerve responses to electrical stimulations of different signal processing techniques. Develop a program that connects to the speech processor from the program to test the sounds of different signal processing strategies. Convert the program from MATLAB to C/C++ environment for interface with other software. Expand the program to reach an ideal teaching tool for instructors in educating audiologists and other hearing professionals how cochlear implant recipients hear sounds with various types of signal processing strategies

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