## The Coding of Sound by a Cochlear Prosthesis: Faculty Project Description

**Summary:** The goal of this project is to engage first year electrical engineering (EE), computer engineering (CE), computer science (CS) and electrical engineering technology (EET) students in improving an implantable biosystem that overcomes profound hearing loss in the human auditory system through electrical activation of auditory nerve fibers, that is, a cochlear prosthesis. To achieve this goal, we outline the following procedure: (1) describe the form and function of the auditory system from a systems engineering perspective; (2) introduce students first to fundamental engineering concepts such as the distinction between time domain and frequency domain, frequency spectrum, frequency response and Bode plots, and then reinforce these concepts through operation of a LabVIEW (National Instruments Corp., Austin, TX) graphical user interface (GUI); and (3) guide students through the functional blocks of a cochlear implant signal processor by using another LabVIEW GUI that exercises a virtual instrument based on MATLAB (MathWorks Ltd., Natick, MA). Students will transition from an *exploration* of a cochlear prosthesis to team-based *investigation* by modifying critical parameters for the signal processor, and assessing the output both aurally (sound) and visually (plots). Each group will face an engineering challenge of choosing parameters, for example, channel count and listening environment, to achieve the best performance—essentially understanding the design tradeoffs when building a cochlear implant speech processor.

This project will serve as a real world exercise illustrating the fundamental contribution by engineers in improving the quality of life for profoundly deaf individuals. The expected **technical outcomes** are: (1) development of a fundamental knowledge base in signal processing, (2) introduction to small group problem solving both by understanding a problem statement and exploring potential solutions, and (3) examining engineering tradeoffs in cochlear signal processor performance vs. system complexity and power. The expected **broader outcomes** fall in three categories: (1) multidisciplinary—the examination of human auditory system as a signal processor, (2) societal—understanding the challenges and solutions associated with sensorineural deafness, and (3) international—cross-cultural comparison of signal processor performance inputting a variety of languages and music.

Due to the level of signal processing knowledge required to modify the cochlear implant simulation (beyond setting the channel count), this project emphasizes the real world aspect of making higher-level design decisions that have a direct societal impact. The strategy is to first provide students with a basic background, and then engage students in critical design. The idea is to continue building upon the technical challenges using this first year project as the students progress through their respective programs. Expanding upon this project is addressed in the Summary Lecture.

What follows is a roadmap for implementing "The Coding of Sound by a Cochlear Prosthesis." First, the components and mechanics of understanding and exercising the coding of sound by a cochlear prosthesis with references to the Overview Lecture Slides are addressed. Second, this is followed by a description of the student project. The syllabus may be tailored from one to two weeks based on the signal processing knowledge of the students. Table 1 (Slide 2). outlines the flow of topics, concepts, engineering challenges and approximate time investment.

### Background Material

*Speech generation from a sound source (Slide 3)*

To understand how a cochlear implant signal processor works, it is beneficial to first understand the creation of speech signals and how they are propagated. There are two classes of speech signals characterized by the means of how they were produced. Unvoiced speech is created by forced air. An example is the sound /th/ created by pressing the tongue against the teeth and forcing air through. These sounds are non-periodic. The second class is voiced speech. These signals are created by vibrations of the throat and are periodic (they have a repeating pattern). Slide 3 illustrates a voiced speech signal. The plotted signal is a voltage produced by a microphone after it detects pressure waves created by the voiced vibrations and translates them into voltages. Notice that the voiced signal repeats every 9 milliseconds. Therefore the signal is periodic with a fundamental 111Hz.

*Propagation of sound (Slides 4-6)*
Speech signals are transmitted through a medium, usually air, as pressure waves that are sensed by the eardrum. Air particles are in random motion, but very small pressure variations occur about a steady state pressure. These variations are compressions and rarefactions. When a sound source sets up a vibration, an air particle moves about its steady-state position along an axis parallel to the direction of the pressure wave. The wave propagates because particles, that do not move very far, transfer the pressure wave to other particles. The velocity of sound propagation in air is about 344m/sec, the pressure wave travels at that speed. Pressure waves are also characterized by a period which is the time between two successive compressions or rarefactions. The frequency is the inverse of the period, that is, the number of compressions or rarefactions in one second. The units of frequency are Hz (1/second). The frequency range of human hearing is 20Hz to 20,000Hz. Sound waves move out spherically from a point source, and become less intense as they move. Their intensity drops off as the square of the inverse of the distance from the source. When sound waves reach an obstruction they are both reflected and transmitted.

*Natural detection of sound: form and function of the peripheral auditory system (Slides 7-10)*
In detecting sound, the outer ear plays a significant role. The outer ear is designed to amplify (increase) the intensity of sound and sort out the parts of the signal that are important for speech and basic survival. Sound pressure waves cause a thin membrane, eardrum, to vibrate. Connected to the membrane are a series of three small bones, the ossicular chain—malleus, incus and stapes. The ossicular chain translates pressure changes into mechanical vibrations. Together the action of the middle ear serves to minimize sound reflection through impedance matching between the air in the canal and the fluid in the cochlea. Between the eardrum and the cochlea, sound waves are amplified 200 times.

The stapes couples these mechanical vibrations to the fluid-filled spiraling cochlea. The cochlea is divided into an upper and lower chamber by the basilar membrane. Along the 2.5-turns of the human cochlea, the basilar membrane houses sensory cells—hair cells. In each human ear there are 30,000 hair cells that function to translate, or transduce, fluid vibrations into electrical signal conducted by nerves to the region in the brain dedicated for hearing. Lining the surface of the hair cells are hair-like projections called stereocilia. As the basilar membrane vibrates up and down, the stereocilia undergo shear forces and initiate a series of electrochemical reactions resulting in an electrical signal being conducted by nerve fibers. The basilar membrane exhibits a variable stiffness such that it responds maximally to low frequencies in the apex, and responds maximally to high frequencies near the base. Essentially, each traveling wave in the cochlea exhibits a peak that is directly related to frequency of sound. If a sound signal contains multiple frequencies, then multiple locations along the cochlea will respond to the traveling wave. The cochlea serves to separate the frequencies in the sound signal—the frequency spectrum. The characteristic is referred to as the cochlea's tonotopic mapping and was discovered by Georg von Békésy who received a Nobel Prize in Physiology or Medicine in 1961 for this discovery.

*Sensorineural hearing loss (Slides 11-12)*
Sensorineural hearing loss occurs when the sensory cells in the basilar membrane are unable to transduce the fluid vibrations in the cochlea into electrical signals. A variety of causes may lead to sensorineural hearing loss. Congenital defects, head trauma, excessive noise and one

particular class of antibiotics (aminoglycosides) may all lead to sensorineural hearing loss. Fortunately, although hair cells are damaged, the auditory nerves do respond to electrical stimulation. A cochlear implant can directly stimulate auditory nerves.

*Assessing hearing loss (Slides 13-14)*

An audiogram is a graphical representation of an individual's perception to single tones (single frequency sound signals) applied to the auditory canal. The reference point for making these measurements is the minimum pressure change that humans can detect, $2x10^{-5}$ Newtons/$m^2$ at a frequency of 1000Hz. Regarding frequency, humans are able to detect sounds separated by 1Hz or greater. Since sound waves have such a broad range of frequencies, an appropriate scale is required to represent the signals. A logarithmic scale, decibels (dB), is used. One decibel represents an intensity increase of a factor of 10. For an individual to become a candidate for a cochlear implant, he/she must exhibit a sensorineural hearing loss of 90dB or more.

*System level description of a cochlear prosthesis (Slides 15-18)*

A cochlear prosthesis consists of an external portion and an internally implanted portion. First, external sound is collected by a microphone where the pressure waves are transduced into analog voltages. A signal processor extracts the frequency information from the sound signal that is important for speech and outputs them as coded digital data streams. A radiofrequency transmitter sends the digital waveforms through the skin to an implanted receiver/stimulator. Internally, the stimulator uses the digital codes representing speech to construct current waveforms that convey speech information to the brain via the auditory nerve. Contemporary cochlear implants provide 16-22 metal sites along a molded silicone array inserted by a surgeon into the cochlea. Using frequency information extracted and coded by the signal processor, electrical current delivers charge to the tissue at 16-22 appropriate locations along the cochlea. When compared with the 19,980Hz range of frequencies spanning the cochlea, it is remarkable how effective cochlear implants are with just 16-22 stimulation sites.

*Cochlear implant signal processor: functional block diagram and acoustic simulation (Slide 19-28)*

Once the sound is converted into voltage by the microphone, the signal processor takes command. Its essential role is to transform speech signals into electrical stimuli. We will examine the most widely used cochlear implant signal processing strategy, continuous interleaved sampling (CIS). In this method, electrodes are stimulated in a non-simultaneous fashion to prevent channel interaction, which results when the currents in the cochlea activate more neurons than desired resulting from parallel stimulation.

The first functional block of the signal processor is the preemphasis filter. It serves to accentuate high frequency, low-energy signals. Effectively, the filter brings the high frequency content, especially important for understanding consonants, to a level commensurate with low frequency, high-energy signals.

Following preemphasis, the speech signals are separated into a predetermined number of frequencies. This is achieved through bandpass filters. Each bandpass filter serves to let in a given range of frequencies called a frequency band. The next step is to determine how strong the speech is in each frequency band. This strength is called the energy content. The energy content is determined through processes called full wave rectification and low-pass filtering. The loudness of the resulting signal is modified to fit the dynamic range of the patient.

In this exercise, the energy content in each band will be used to set the amplitude of a corresponding sinusoid representing the particular frequency band. The overall speech signal is generated from the sum of these sinusoids.

**Project Description and Implementation** *(Slides 29-34)*
The project consists of a Pre-lab section to be completed independently, and an Engineering Challenge section to be completed in teams of 3-4 students. The Pre-lab consists of Part 1 where students exercise a Basic_Filters LabVIEW GUI. Simple filters are explored. Part 2 provides students with the ability to simulate a cochlear implant signal processor. Using a virtual instrument panel (or GUI, see below) the student can input and audio .wav file, set the number of bandpass filters (channels), and listen to an acoustic simulation. Part 2 also requires the students to vary the listening environment, e.g. quiet or noise, and assess speech intelligibility.

Delving further into the signal processing aspect is beyond the scope of a first year project for EE, CE, CS and EET students. The underlying Matlab code may be provided to the instructor if desired. As an alternative to coding, the students are challenged as teams to consider improving performance of cochlear implants for speech recognition or music appreciation. In addition, international students may consider cochlear implant performance in the context of their respective native language.

To implement the project precompiled executable files for the GUIs should be downloaded from an IEEE server and exercised in LabVIEW (details outlined in the *Student Project Assignment*). The instructor has two options for providing the LabVIEW platform. For Option 1 the instructor must provide the students with LabVIEW installed on personal computers. For Option 2 the instructor can direct the students to download and install a *free* evaluation version of LabVIEW form National Instruments' website. The only drawback to Option 2 is the limited 30-day trial period, but ideally the project should be completed within that time. The executables have been verified on National Instruments LabVIEW 8.5 and LabVIEW 2009. Table 1 summarizes the requirements for the two options.

**Table 1: Real World Project Implementation Requirements**

| | Option 1: Purchase LabVIEW Software | Option 2: Free LabVIEW eval. software |
|---|---|---|
| Hardware | Processor: Compatible Pentium<br>RAM memory: 16MB<br>Screen resolution: 800 x 600 pix.<br>Sound card: Compatible with Windows<br>Microphones and headphones | Windows 2000 or later specifications<br>Microphones and headphones |
| Software | Microsoft Windows XP<br>Internet Explorer 7.0<br>LabVIEW 8.5 or LabVIEW 2009 | Microsoft Windows 2000 or later<br>Internet Explorer 7.0<br>LabVIEW Evaluation Software |
| Required code | Basic_Filters_GUI.exe<br>Cochlear_implant_GUI.exe<br>-or-<br>Basic_Filters_GUI.vi<br>Cochlear_implant_GUI.vi<br>*Available on request from author* | Basic_Filters_GUI.exe<br>Cochlear_implant_GUI.exe |

**Supplementary and follow-up material**

Table 2 provides a glossary of terms to assist the students in the "language" of signal processing. As the students expand their respective skills in signal processing, a number of online resources can be accessed. Table 3 lists such resources. In addition, an illustrative youtube animation of basilar membrane motion is listed and an extensive text on cochlear implants.

| **Table 2: Glossary** | |
|---|---|
| Analog | A continuous signal |
| Aperiodic | A signal that does not exhibit any repetitive pattern. |
| Bode Plot | A standard method for examining the gain (in dB), and phase of a system as a function of frequency (in log f). |
| Decade | A factor of 10 |
| Decibels | A relative scale employed to measure the increase, or decrease, of a signal in reference to a given base: 1dB = $20\log(A_2/A_1)$ |
| Digital | A discrete signal |
| Dynamic Range | The range of intensities that can be perceived, acoustic or electric<br>For electric hearing it is the range between the T-level and the MC-level |
| Fourier Spectrum | A frequency domain representation of a signal that includes all the frequency components of a signal |
| Gain | The amplification of a signal by a system: Output/Input |
| Logarithmic Scale | A scale used to represent signals that exhibit substantial changes. As a result, the signal must be compressed to view it's behavior with a reasonable scale |
| Maximum Comfort (MC) Level | The maximum level at which a patient can tolerate an input (acoustic or electrical) |
| Period | A signal that repeats itself with time. |
| Pre-amplifier | This boosts signals from the microphone to a level that can be handled by the speech processor. |
| Pre-emphasis | A filter that accentuates high-frequency sounds that are of low intensity, which are important for understanding consonants.<br>A typical frequency response is a high-pass filter with a cutoff frequency of 1200Hz and a roll-off of 6dB/decade |
| Roll-off | The slope, or rate, at which a signal decreases (increases) |

| **Table 3: Reference and Resource Description** | |
|---|---|
| http://www.youtube.com/watch?v=dyenMluFaUw&feature=related | Youtube clip illustrating a traveling wave down the basilar membrane of the cochlea. |
| http://www.utdallas.edu/~loizou/cimplants/ | Tutorial: Good overview, but more appropriate for second year students.<br>Listening demos: This exercise was inspired by these demos. |
| http://www.mathworks.fr/products/sigprocblockset/demos.html?file=/products/demos/shipping/dspblks/dspcochlear.html | Matlab cochlear implant exercise |
| Clark GM. (2003), Cochlear implants: Fundamentals and Applications. New York: Springer Verlag | Graeme Clark's book, a very inclusive reference text for the instructor |