# AC 2010-1061: THE CODING OF SOUND BY A COCHLEAR PROSTHESIS: AN INTRODUCTORY SIGNAL PROCESSING LAB

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# The Coding of Sound by a Cochlear Prosthesis: An Introductory Signal Processing Lab


## Abstract

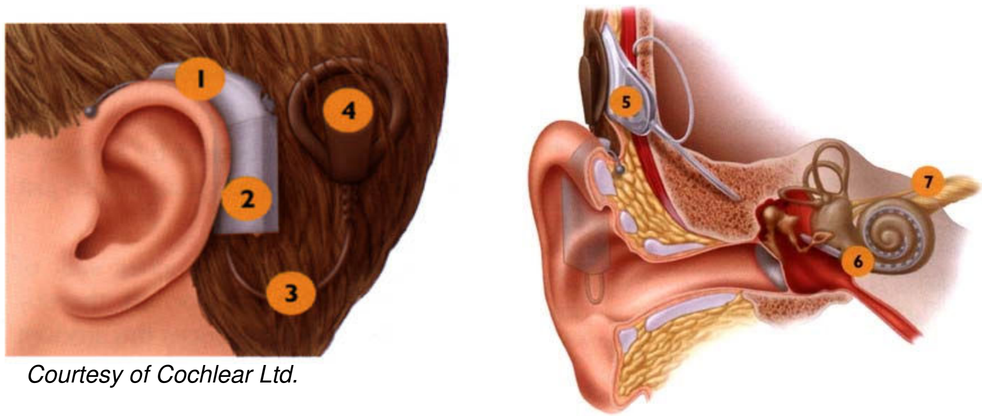
An innovative and pedagogically appealing real-world application—a cochlear implant signal processor—forms the substrate for a laboratory exercise in design, simulation, and qualitative assessment of an engineering problem. In an introductory signal processing course, students are able to write MATLAB code that mimics the operation of a cochlear implant signal processor in which sound information is extracted and then coded for input to a neural stimulator. Fundamental concepts such as sampling continuous-time signals, discrete-time filter design, filter banks, envelope detection, spectrograms and signal reconstruction are explored and formalized in different parts of this project. To promote interaction across disciplines, the students work in randomly assigned teams of two that often pair up Biomedical Engineering (BME) students with Electrical and Computer Engineering (ECE) students. For many students, the lab presents the first exposure to a collaborative engineering design effort, in contrast to the common independent exercise of tackling a "tough homework problem." Although this laboratory project is quite challenging, it was well liked by the diverse population of BME and ECE students. Efforts are underway to integrate an online post-lab survey during the upcoming term to enable a more quantitative means of assessment. In addition, to provide free international access, the laboratory will be disseminated on the Connexions educational website.


## 1. Introduction

At the Georgia Institute of Technology (GA Tech) an introductory signal processing course is required for all electrical engineering, computer engineering and biomedical engineering undergraduates. To provide this foundational material early on in the undergraduate curriculum[1], the course is presented in the sophomore year for ECE students and often taken during the final two years for BME students. Basic integral calculus, linear algebra, familiarity with complex numbers, and MATLAB (MathWorks, Natick, MA) programming experience are the pre-requisites. This rigorous semester-long course consists of three main instructional components: (1) a faculty led bi-weekly lecture, (2) a faculty led weekly recitation section, and (3) a weekly laboratory section co-led by a faculty member and graduate student teaching assistants. Both the recitation and laboratory enrollment is limited to 20 to enhance the level of faculty-student contact during the labs and recitations.

The weekly laboratory exercises consist of a simple pre-lab that the students are to complete on their own, a structured in-lab warm-up section examined by the lab staff, and a more in-depth exercise/project completed by the students outside of the lab section in teams of two. These projects require a lab report which is sometimes a formal report as in the case of the Cochlear Implant Lab (CI Lab). Although the labs have been crafted to build upon and reinforce fundamental concepts presented in the lecture, many students must work hard to integrate the theory from the recitations and lectures with the practical implementations required in the labs. Historically, students often view the material in other signal processing courses as very mathematical and sometimes struggle to see where it can be applied. Labs such as the CI Lab address that shortcoming, and, with 30-40% of the student population coming from the biomedical engineering program, there is a strong need for even more biomedical engineering applications in the signal processing labs.

The goal of the CI Lab project is to engage students in the simulation of a real-world biomedical device—a cochlear prosthesis. Over 180,000 individuals[2] worldwide rely on a cochlear prosthesis to overcome severe to profound sensorineural hearing loss. Through direct electrical stimulation of auditory nerve fibers in the inner ear, the prosthesis conveys the sensation of sound. A signal processor determines what information in a sound signal is of interest, extracts the important signal parameters, and converts them into codes used by the stimulation electronics to output appropriate stimulating current pulses[3] (Fig. 1).



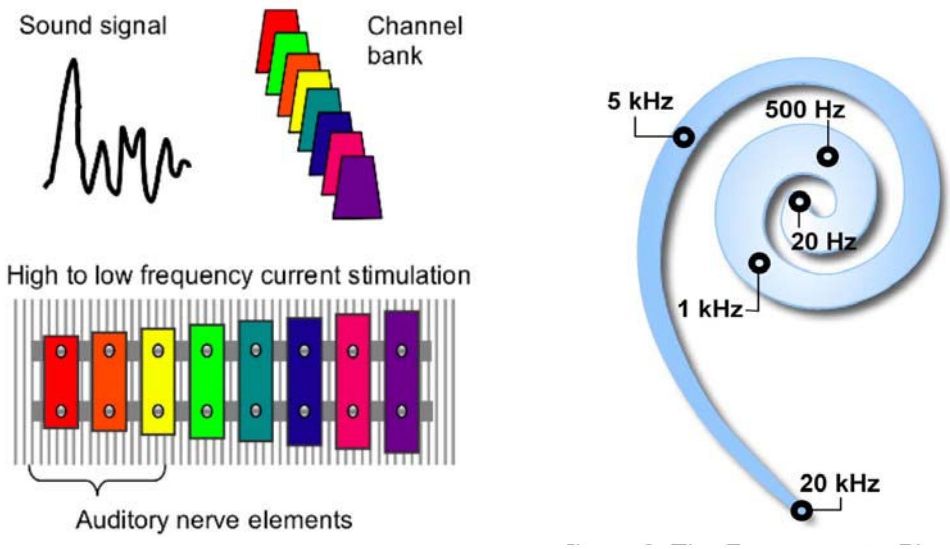

*Courtesy of Cochlear Ltd.*

**Figure 1:** External and Internal portions of a cochlear prosthesis. Sound is collected by a microphone (1) and processed to extract information relevant to speech by the speech processor (2). The information is transmitted via an electrical cable (3). This information is transmitted wirelessly across the skin through a two-way telemetry link (4). The internal receiver (5) uses this information to construct current pulses that are delivered by the electrode array (6) to auditory nerve elements (7) that lead to the perception of sound.

For a lab exercise the students are required to write MATLAB code that mimics the operation of the signal processor. The architecture of the signal processor is a filter bank that decomposes the input sound signal into many frequency channels (Fig. 2a). To verify the correctness of their code, they apply their MATLAB processing to various input acoustic signals, e.g., speech and music. Rather than generate an output for neural stimulation, an acoustic output is produced so that normal listeners can appreciate and evaluate the operation of the processor. Students are directed to compare the output produced from male and female speech as well as music. To understand engineering tradeoffs, the students are instructed to vary the number of bandpass filters used in processing sound and evaluate the resulting signal quality.

## 2. Approach and Pedagogy

The cochlear prosthesis application resides at the intersection of signal processing and BME, and serves as an ideal platform to introduce undergraduate BME students, as well as ECE students, to signal processing concepts. Although the cochlear application has been presented in courses elsewhere[4], it is often in an upper-level course where the students have already had an earlier opportunity to solidify such concepts such as the distinction between continuous and discrete time, z-transforms, as well as digital filter design and implementation. In contrast, this lab is folded into an *entry-level* signal processing course at the sophomore level. The CI Lab is based on the filter-bank architecture, but there have been many other successful labs in the basic signal processing course that have also focused on filter banks. Examples are touch-tone telephone decoding and music decoding. One advantage of using the hearing application for a lab project is that it motivates further study of advanced digital signal processing courses where labs related to speech processing and the auditory system are enthusiastically pursued by the students.




a) b)

**Figure 2: a)** A cochlear implant signal processor serves to separate the sound content in an acoustic signal into distinct frequency bands. Similar to chiming the keys on a xylophone, a cochlear implant activates distinct frequencies in the cochlea through electrical stimulation of auditory nerve elements. The brain perceives these targeted and organized neural activations as sound cues. **b)** Frequency-to-place mapping of the human cochlea. The acoustic frequencies that the human cochlea responds to range from 20kHz at the base to 20Hz at the apex.

### 2.1 Cochlear Implant Signal Processing Overview

In the lecture portion of the course, students are learning theoretical concepts such as signal attenuation, amplification, frequency response, and bandpass filtering which are components of the cochlear prosthesis system. To provide students with specific background for this lab, it is necessary to provide an overview of natural human hearing illustrating the ability of the ear to process sound. In particular, the cochlea performs a function that amounts to a frequency-to-place mapping (Fig. 2b), which can be modeled in signal processing as a filter bank. Cochlear prostheses are so effective because they are able to exploit the cochlea's frequency-to-place mapping. By stimulating at a given cochlear place, the implant is able to target a specific frequency band and thereby deliver cues that the brain perceives as frequency specific.

The CI Lab implements the most widely used cochlear implant signal processing strategy, Continuous Interleaved Sampling, CIS[5]. The signal processor consists of four steps in order to process a sound signal: (1) pre-emphasis, (2) multiple bandpass filters, (3) envelope detection and (4) current pulse generation (Fig. 4). The students are tasked with emulating steps 1-3 in MATLAB. Step 4 is replaced with an acoustic output, which is formed by using the detected envelopes to modulate sine waves. Each modulated sine wave is representative of a single acoustic channel. One of the design trade-offs in a commercial implant device is complexity versus sound quality which depends primarily on the number of channels.

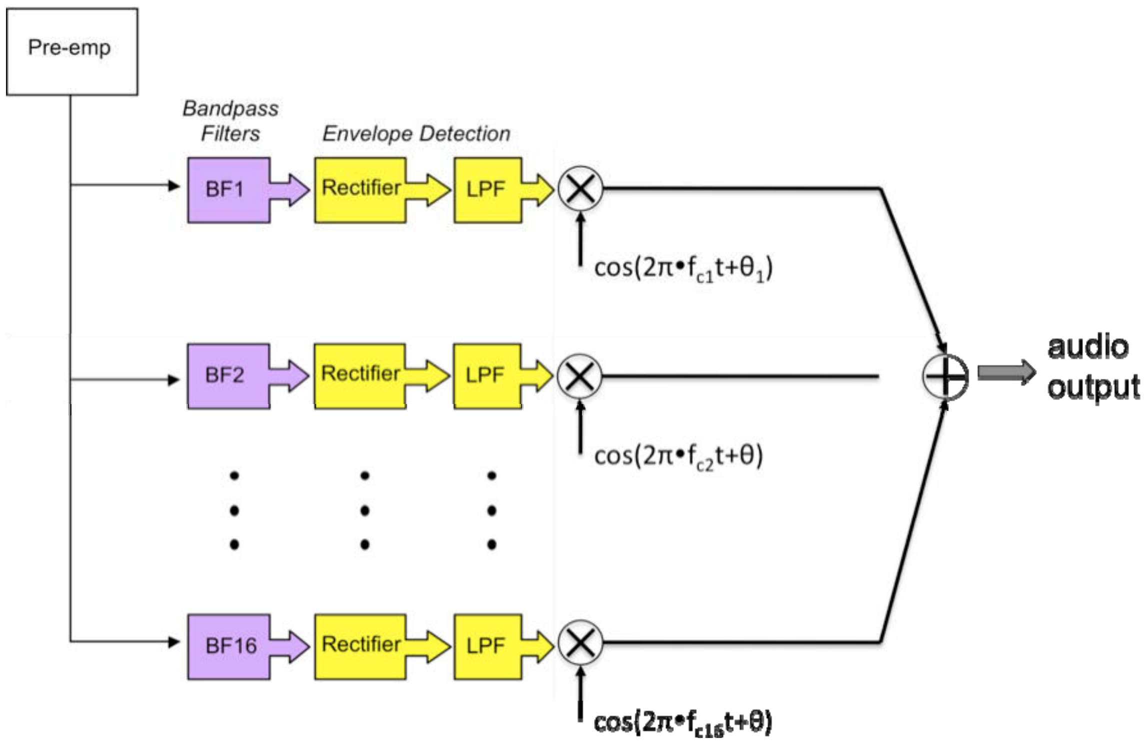



**Figure 3:** The four steps taken to simulate a cochlear implant signal processor: (1) pre-emphasis, (2) multiple bandpass filtering, (3) envelope detection and (4) sinewave modulation.

## 2.2 Cochlear Implant Signal Processing Laboratory Exercise

Two features of the CI Lab distinguish it from many of the other labs in the course. First, since the CI Lab relies on detailed knowledge of the human auditory system, a full lecture is dedicated to describing the anatomy and physiology of normal human hearing, and how a cochlear implant serves to overcome sensorineural hearing loss. These details should be particularly interesting to the BME students. While the students are not formally tested on the biological component, the exposure to the frequency-to-place mapping done by the basilar membrane of the cochlea is a direct illustration of how the function of human hearing can be modeled with fundamental signal processing concepts, e.g., bandpass filtering of sound by the cochlea. In other words, the central theme of the filter-bank architecture is reinforced by biology.

Second, the in-depth project portion of the lab has some flexibility, especially with respect to the filter design. The bandpass filters in the filter bank can be either FIR (Finite Impulse Response) or IIR (Infinite Impulse Response) digital filters, but the filter specifications would be given in terms of the desired frequency response and the constraints for matching the passbands and stopbands. In other words, the specifications are soft rather than hard, so students are required to make some decisions about the parameters needed to carry out their own filter design. We have found that having a preceding lab devoted solely to filter design is a good way to develop these skills. For example, in the FIR case the crucial parameter is the filter length, which depends on how tight the passband and stopband constraints are, and on the closeness of the passband and stopband edges. Learning these tradeoffs ahead of time is crucial to making decisions when designing all the bandpass filters for the filter bank. This part of the lab project is demanding and sometimes frustrates students as they transition from a focus on *skill mastery* as the endpoint of their studies to a focus on a goal such as the *implementation to realize the application*.

This lab project is one of two major projects during the course, so a formal lab report is required. Table 1 presents a typical timeline for integrating this lab into an introductory signal processing course. To add rigor to the exercise, several tests may be carried out from a working MATLAB implementation. When making tradeoffs, the primary metrics for assessment are intelligibility for speech, and sound quality for music. Even though the system is designed for speech, it is interesting to study how it performs for music. The students are required to vary the number of channels in the filter bank from four to sixteen, as well as the center frequencies of the bandpass filters, and then assess the reconstructed acoustic signal. Parameters of the filter design can also be varied to study the output sound quality versus filter precision. Some of these tradeoffs can be left to the imagination of the student teams. Fortunately, this approach prepares them well for subsequent courses both within signal processing and other scientific disciplines. In addition, requiring the students to examine design tradeoffs as a team prepares them well for an industry setting as well as research opportunities at the university.

**Table 1. Laboratory Project Timeline**

| ***Task*** | ***Objective*** | ***Duration (3-weeks total)*** |
|---|---|---|
| Filter design lab | *Establish* skills in filter design constructing IIR or FIR filters. | In lab exercises (1.5-hour)<br>Informal lab write-up (2-hours per team) |
| Introduction to cochlear implants | *Build and equalize* knowledge base of ECE and BME students with respect to biological aspects and cochlear implants. | 1 course lecture (1-hour) |
| Cochlear implant lab: pre-lab | *Introduce* students to the respective building blocks in a supportive setting. For example, coding the pre-emphasis filter. | In lab exercises (1.5-hour) |
| Formal cochlear implant lab | *Challenge* students with developing MATLAB to simulate a cochlear implant signal processor. | Formal lab write-up (4-hours per team)<br>In-lab demonstration and discussion of testing the working code on audio clips (15-minutes per team) |

## 2.3 Vertical Integration and International Dissemination

At GA Tech an entry-level signal processing course feeds into an upper division digital signal processing lecture course, as well as a senior-level project/laboratory course. The cochlear implant signal processor lab serves as an excellent platform to further investigate applications in speech processing such as models for speech synthesis, time-domain speech processing, short-time Fourier analysis, design of speech coders and man-machine communication.

In earlier courses at the freshman level, some parts of this material may be introduced to engineering students during their *first* year through an IEEE Real World Engineering Project that examines the coding of sound by a cochlear prosthesis. Through this internationally and freely accessible portal, the students are able to exercise a LabVIEW (National Instruments Corp., Austin, TX) graphical user interface where they are introduced to basic filters (high pass, low pass and bandpass), and can vary the number of filters in the filter bank from two to sixteen[6]. By running audio files through the cochlear implant simulation and listening to the output the students gain an exposure to simple signal processing ideas, but are shielded from writing MATLAB code. To support course instructors, materials such as lecture slides, a project description and a project solution are also readily accessible.

In the spirit of international and free dissemination, the CI Lab presented here will be made available at the Connexions website[7]. Connexions is a mature site that is well-known internationally in the signal processing community, and it should provide an ideal venue for packaging all the components needed in this lab. A LabVIEW virtual instrument is under development that exercises all functions of the cochlear implant signal processor.

## 3. Student Response and Conclusion

Student responses for the first offering of the cochlear lab were very positive. For example, a student commented that s/he would like "More labs (like speech processing and cochlear implant) that apply directly to "real-life" examples." Another student commented that "the diagrams on the second formal lab (cochlear implant) helped us out a lot in creating a code."

The CI Lab is the culmination of efforts in filter design and enables the students to exercise a real-world example of signal processing. Although implementing the CIS algorithm follows a standard design flow, there is some flexibility in filter design. For example the first semester the CI Lab was offered, the bandpass filters were designed as FIR filters; in the second the bandpass filters will be designed as IIR filters. Additionally requiring the students to compare processing of different speakers, male vs. female, or child vs. adult, or speech vs. music is a means to vary the exercise from one term to another.

Although the students are from diverse backgrounds, the knowledge base is very similar (integral calculus, linear algebra, familiarity with complex numbers, and MATLAB). The BME students and ECE students perform equally well on the labs, homework assignments and quizzes in this course. No anecdotal or quantifiable difference in the two populations has been observed.

During the current offering of the course, an online post-lab survey will be given as part of the lab project. This will provide some quantitative data for assessment.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