Department of

Electrical & Electronics Engineering

Abdullah Gül University

Lab 15c: Cochlear Implant Simulation with a Filter Bank

EE3001 TELECOMMUNICATION SYSTEM DESIGN

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OBJECTIVE

The main goal of this lab was to simulate a cochlear implant system on MATLAB in software. The main aim was to model the way modern cochlear implants process sound to deliver the correct electrical signals to the auditory nerve. First, the simulation pre-emphasis filtering, after that multi-channel bandpass filtering, then detected envelopes through full-wave rectification and low-pass filtering and finally, modulated sinusoidal carriers with the extracted envelopes. It meant that the system could imitate what cochlear implant users hear, as the signal is reconstructed only using amplitude features from multiple bands. The project also looked at understanding the intelligibility of the reconstructed signal and tried to see how various filter settings affected how the speech sounded.

BACKGROUND

Cochlear implants are used in patients with deep or severe hearing loss, since regular hearing aids don't help them. Unlike hearing aids which only increase sound, cochlear implants attach to the auditory nerve through a series of electrodes instead of relying on the hair cells. By analyzing the speech input in frequency bands, the brain stem speech processor compares the amplitude of the sounds and turns these amplitudes into an electrical stimulation pattern.

The basilar membrane in the cochlea of a healthy human performs a mechanical analysis of sound frequencies by causing activity first at the apex for low frequencies and then at the base for high frequencies. That structure is also present in cochlear implants by organizing the bandpass filters in a logarithmic manner.

The important elements in a cochlear implant's signal processing chain are:

- Boosting high frequencies is called pre-emphasis which helps you understand complex consonants clearly.
- The function of a filter bank is to break audio into different frequency sections.
- The amplitude changes of the subband signals are found by going through envelope detection.
- A sinusoid in each band is modified by an envelope based on the shape of its envelope modulation.

By generating speech only with these envelopes and fixed carriers, we make it sound like speech heard by cochlear implant users, with a better sense of the rhythm and changes in volume, at the cost of pitch and quality. $\mathrm{H(z)} = 1 - \mathrm{z} - 1$

DESIGN AND TEST PROCEDURES

1. **Gathering and preparing audio files:**

I recorded my own speech sample in MATLAB's audio recorder at 22.05 kHz and saved it as myvoice.wav. I made the track mono and normalized it, if needed. A pre-emphasis FIR filter is used.

Function $\mathrm{H(z)} = 1 - \mathrm{z} - 1$ was then set up. Such a filter blocks low sounds that interfere and increases attention to higher ones that improve speech clarity.

2. **Filter Bank Design**

An 8-channel IIR bank of bandpass filters was designed. The filters were all placed at random locations, all centered on a frequency fc and had their own bandwidth B as described by cochlear frequency-place mapping.

- The center frequencies are: [394, 692, 1064, 1528, 2109, 2834, 3740, 4871] Hz.
- The available bandwidths are [265, 331, 413, 516, 645, 805, 1006, 1257] Hz.

In every channel, the poles were located on the z-plane by using:

- Pole radius: $r = 1 - \frac{B}{f_s/2}$
- Pole angle: $\theta = 2\pi f_c / f_s$

Zero and complex conjugate pole filters were used at ±π and radius r with angle θ.

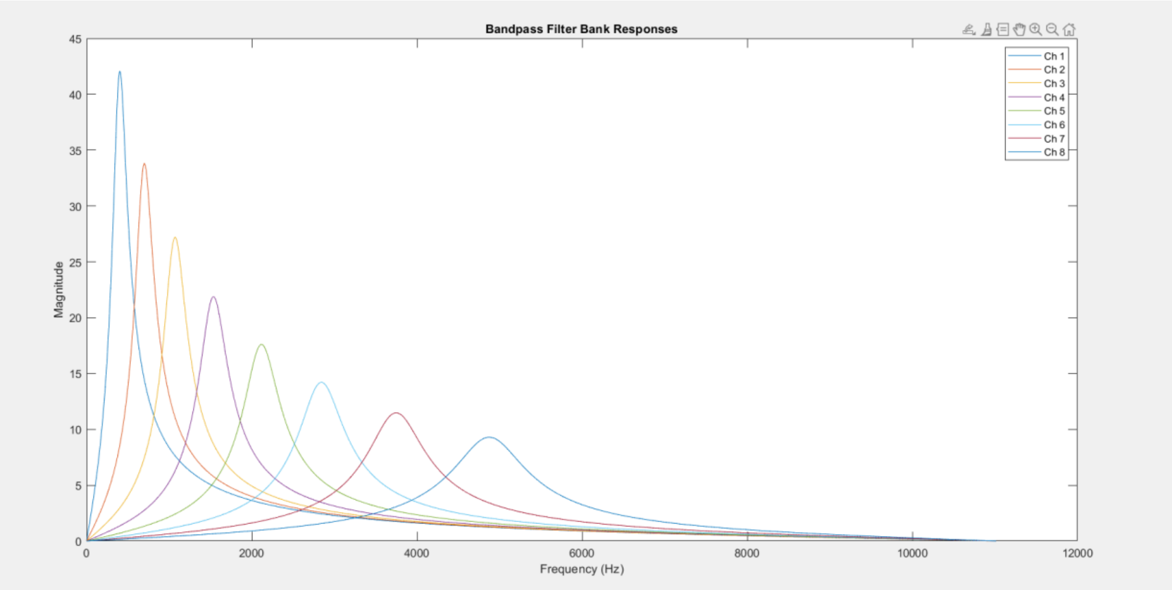

Figure 1 Frequency responses of the 8-channel bandpass filter bank. Each filter is centered at a distinct frequency, covering the speech bandwidth. The gain variations reflect the selectivity and bandwidth of each channel.

3. Envelope Extraction

The instantaneous magnitude of the filtered signals was calculated by full-wave rectification. After the calculation was done, a third-order Butterworth filter was applied to smooth the signals, while the cutoff frequency was adjusted according to the channel bandwidth. A DC-notch filter is created as follows:

$$H(z) = \frac{0.5(1+a)(1-z^{-1})}{1-az^{-1}} \quad where\ a = 0.995$$

The DC offset was wiped out and the signal only included important modulations.

4. Envelope Modulation and Reconstruction

A sinusoidal carrier was used to multiply every extracted envelope at the center frequency of that envelope. All these amplitude-modulated signals from all channels were combined to generate the final reconstruction. It follows the same process as linking information from envelopes to the electrode pulses used in real cochlear implants.

5. Playback and Visualization

The reconstructed data was made normal, recorded as a WAV file and checked with audio equipment. The visual difference between both the source and the recovered signals was also analyzed using time-domain and spectrogram graphs. The spectral features visible in the spectrograms showed which parts of the signal were in tact and which were modified.

RESULTS AND DISCUSSION

The new signal could be recognized by most of the participants as voices, but it also sounded robotic and metallic like descriptions by cochlear implant users. Main syllables and the way they were pronounced were present, yet their natural pitch and voice tone were mostly gone.

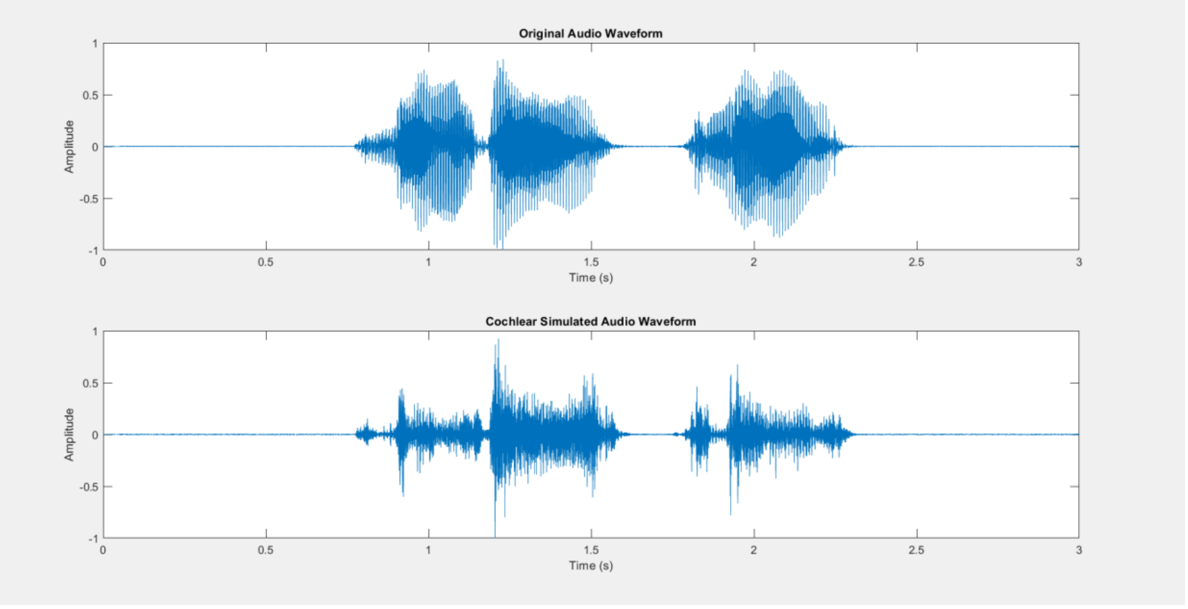

Figure 2 Checking the waveform signals at their moment-by-moment interaction. Figure A shows the original speech signal and Figure B shows how that signal is processed by the simulation cochlea. Despite missing its high notes, the piano keeps its outline.

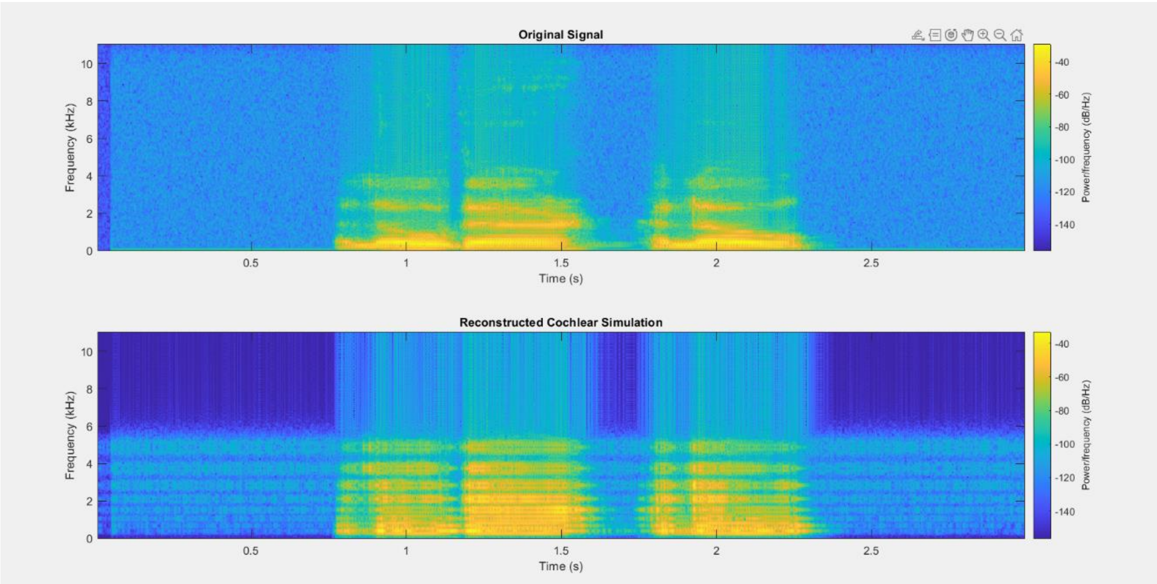

Figure 3 The spectra corresponding to the original signal are above and the spectra occurring only in the cochlear output are presented below. In the reconstruction, the sequence of events is kept, but no finer harmonic or pitch elements are present.

Spectrogram Comparison:

- We observed in the original signal that harmonics were clear, the signal included a large range of frequencies and there were many fine structure details.
- The output of the cochlear-based simulation expressed time envelope differences but lacked the higher order harmonic frequencies that were not included in the model.

<u>Observations:</u>

- The signal was still understandable with 8 channels.
- Bringing the channel to 16 helped a little with understanding; making it 4 gave me almost no idea of the video.
- Considering speech, this part of digital audio helped increase consonant clarity.
- Because DC-notch filters were not used, the energy in the baseline made the envelope move less.
- It was obvious from the waveform that envelope-shaped amplitude changes were present in each sinusoidal region.

Limitations:

- Since the pitch and harmonic information was lost, there was poor music processing.
- Ideal carriers of sinusoidal signals are used, but this ignores matters of phase and auditory blockage.
- The processing load of a network is linked to both the number of channels and the number of filter orders.

Even with these problems, the simulation allows us to see how cochlear implant processing affects sound perceived by the implanted user.

CONCLUSIONS

We successfully duplicated the central functions of a cochlear implant in this lab using MATLAB. Using a filter bank, along with rectification, smoothing and modulation, we were able to develop a working model for turning an acoustic signal into a surrogate for neural stimulation.

The simulation demonstrated that proper speed of pitch and timbre changes across the frequency bands support the communication of clear speech without the need for actual pitch or harmonic pitches. As a result of this project, our knowledge about auditory DSP, filter design and biomedical field increased. The developed system relates directly to modern hearing aids and underlines the importance of engineering in addressing medical problems.

To check the codes for this project: https://github.com/AbdullahRIHAWI/Cochlear_Matlab

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REFERENCES

1. **Loizou, P. C. (1999).** Introduction to cochlear implants. *IEEE Engineering in Medicine and Biology Magazine*, 18(1), 32–42. https://doi.org/10.1109/51.740962

   Here, you will find a detailed account of cochlear implant technology and the ways the devices process signals.

2. **Loizou, P. C. (1998).** Mimicking the human ear. *IEEE Signal Processing Magazine*, 15(5), 101–130. https://doi.org/10.1109/79.708543

   Several signal-processing techniques applied in cochlear prostheses are discussed in this paper, along with how they mimic the way humans hear.

3. **Bhatti, P. T., & McClellan, J. H. (2011).** A cochlear implant signal processing lab: Exploration of a problem-based learning exercise. *IEEE Transactions on Education*, 54(4), 628–636. https://doi.org/10.1109/TE.2010.2103317

   A laboratory simulation exercise is presented in this paper, teaching students how cochlear implant signal processing functions in a practical way.

4. **McClellan, J. H., Schafer, R. W., & Yoder, M. A. (2015).** *DSP First: A Multimedia Approach* (2nd ed.). Pearson Education. https://www.amazon.com/DSP-First-2nd-James-McClellan/dp/0136019250

   The book describes basic principles in DSP, providing a proper basis for learning about cochlear implant simulations.