UROP REPORT - HEARING WITH COCHLEAR IMPLANTS

Costanza Gulli
Department of Electrical and Electronic Engineering
Imperial College London
costanza.gulli17@imperial.ac.uk

July 10, 2020

ABSTRACT

This document is aimed at briefly describing the work experience I carried out during Summer 2019. As part of the Undergraduate Research Opportunities Programme (UROP), I joined the Next Generation Neural Interfaces (NGNI) Lab at the Centre for Bio-Inspired Technology (Imperial College London). Under the supervision of Dr. Timothy Constandinou and Dr. Francesca Troiani, the project consisted in simulating the performance of cochlear implants through digital signal processing on MATLAB and FPGA.

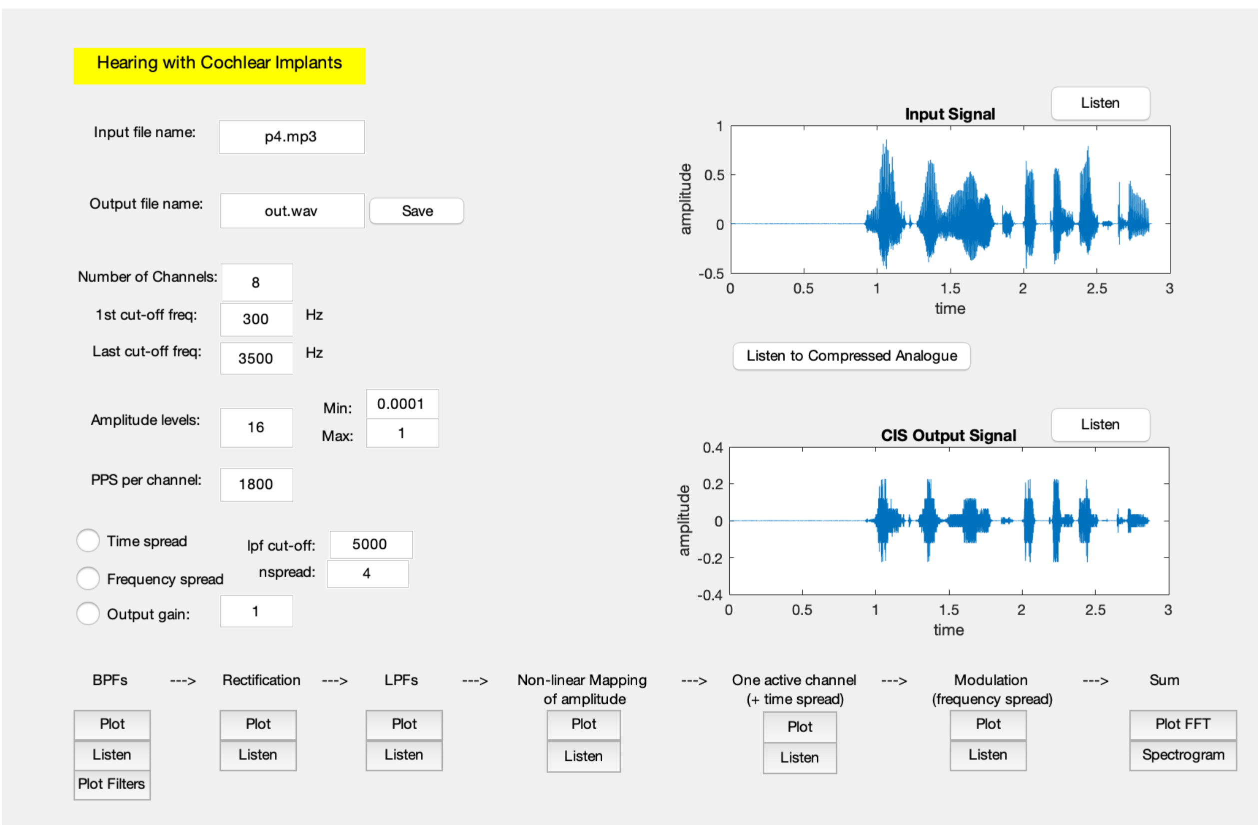


Figure 1: MATLAB Graphics User Interface

1 Introduction

The aim of the project was to develop a device to emulate what a person with a cochlear implant hears. Through a surgically implanted neuroprosthetic, such devices provide the sense of sound to patients with a pronounced hearing disability. These are already used but their current performance in not nearly as good as the natural human hearing.

More specifically, the simulation involves getting some sound or voice through a microphone, processing it in real-time and making the modified version hearable through headphones to another user. The final aim is for the research team to use it in its outreach events. This is useful to demonstrate the significance behind the use of cochlear implants, as well as showing how important it is for research to make further developments in the technology, that at present is not even close to being optimal.

2 Theoretical background

The human ear translates sound waves (20 - 20kHz) into electrical stimuli that the brain perceives as sound. Sound waves are first transformed into mechanical vibrations by the outer and middle ear, then the inner ear translates the mechanical movements into electrical signals that are transmitted to the brain by the auditory nerve. This structure is shown in Figure 2[1].

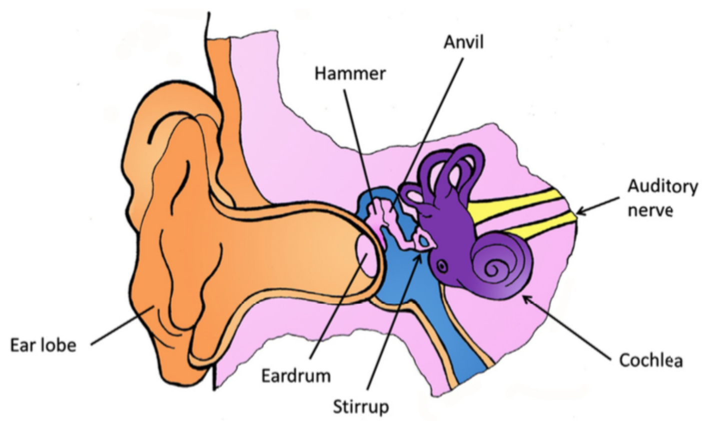


Figure 2: Anatomical structure of the parts of the human ear involved in hearing

Due to the anatomical structure of the cochlea, different parts of the spiral resonate at different frequencies. Frequencies are logarithmically distributed in such a way that low frequencies are placed towards the apical end and high frequencies are placed at the base. This is because for good sound recognition, identifying low frequencies precisely is more relevant than distinguishing between high frequencies that are few Hz apart. This logarithmic relation between physical stimulation on the cochlea and pitch recognition is called tonotopy. The tonotopic arrangement of the ear is illustrated in Figure 3.

Hearing losses can be conductive or sensoneurial. The first occur when sound transmission is impeded between the outer and inner ear, while the latter are commonly due to sensory hair cell or auditory nerve damage. In case of profound or total deafness, cochlear implants can be used to re-establish the hearing partly. In fact when the auditory nerve is in good conditions, it can be directly stimulated by placing electrodes in the cochlea that act on the nerve itself.

Cochlear implant systems are surgically implanted to tackle severe hearing losses. They consist of the following main components:

- An external microphone to receive the sound from the environment;
- A processor;
- A radio transmitter and receiver to transmit the signal through the skull;
- An electrode array placed inside the cochlea that directly stimulates the auditory nerve.

Placed on the outside of the body are the microphone, processor and radio transmitter. Surgically implanted inside the skull are the radio receiver, which is placed on the inner part of the skull in correspondence of the transmitter in order to minimise the distance that the signal has to travel, and the electrode array. It can be noticed that a cochlear implant is quite invasive as a treatment, and is in fact only used in cases when no other solution is feasible.

[1]Figures 1-3 are from Implantable Electronic Medical Devices. DOI: http://dx.doi.org/10.1016/B978-0-12-416556-4.00005-X © 2015 Elsevier Ltd.

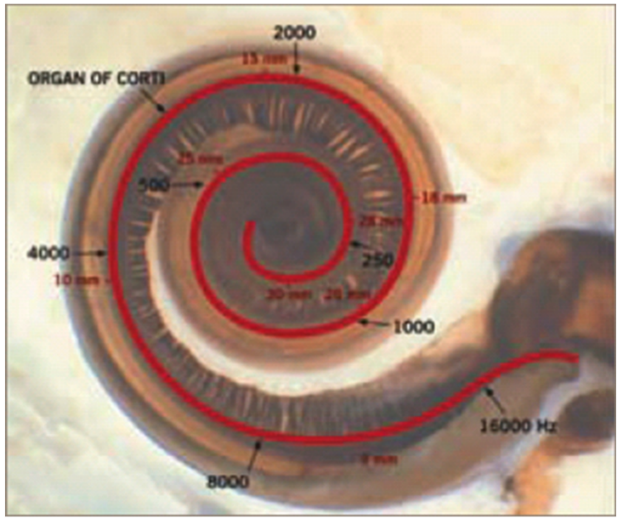


Figure 3: Tonotopic arrangement of the cochlea

3 Speech processing

Speech processing is aimed at simulating what a cochlear implant user hears. The stages can be divided into two main categories:

- some correspond to the processing that happens in the real devices (stages 1-5);
- some are used to simulate physical phenomena that happen in the human ear. These are performed in such a way that they are as close as possible to reality, but are intrinsically an approximation. This is because the physical phenomena involved in hearing are still not completely known. Moreover, the brain has been proven to adapt to the new type of stimulation that occurs with a cochlear implant, but such process is hard to predict mathematically (stages 6-7).

Various processing techniques are used for speech processing in cochlear implants, but the most widely employed is Continuous Interleaved Sampling (CIS). This has been simulated with MATLAB and FPGA simulations. Figure 4 is a summary of the processing stages.

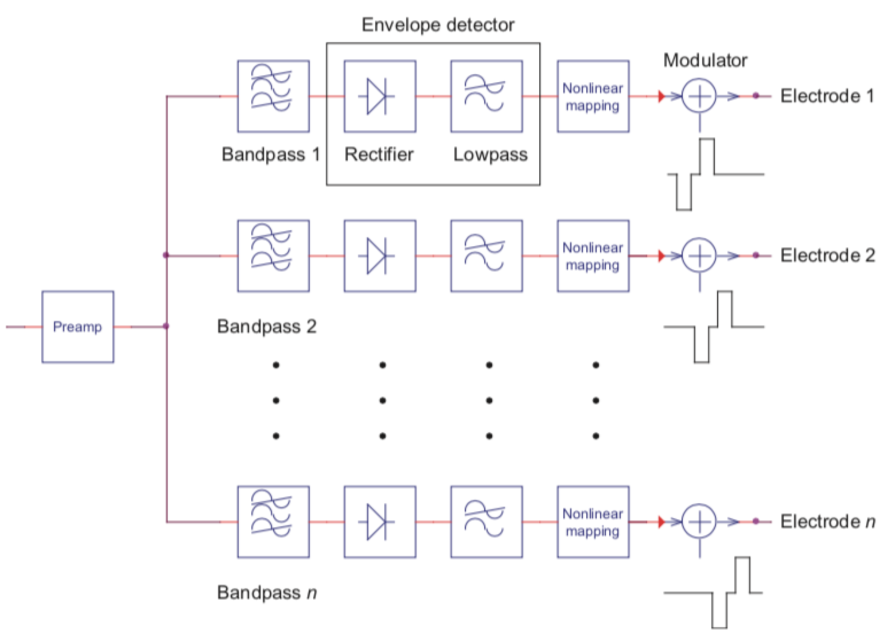


Figure 4: Continuous Interleaved Sampling processing stages

1. **Preamplifier:** Microphone circuits are built so that they have higher gain at higher frequencies. This is because high frequency components have low intensities but provide intelligibility for speech.

2. **Band Pass Filters:** In order to map specific electrodes of the electrode array to specific frequencies, the external speech signal is separated into frequency bands using a filter bank. Figure 5 illustrates the logarithmically-spaced components in which speech is separated. Commonly, bands are 8-12.

   The range of human hearing is 20 to 20k Hz. However, telephones use the range 300 to 3500 Hz because most of the speech components are concentrated in this section of the spectrum. This is sufficient for good sound recognition, therefore only these frequencies are used in cochlear systems and in this simulation. Very low frequencies are also not stimulated because this would require the implant to be surgically placed very deep inside the spiral of the cochlea.

   The filters are 2nd order Bandpass Butterworth Infinite Impulse Response filters. Higher order filters were discarded because they would have added worthless complexity to the program and they exhibited overshooting behaviour at narrow bandwidths.

   From this stage onward, all processing is done on separate channels. These are summed only at the end.

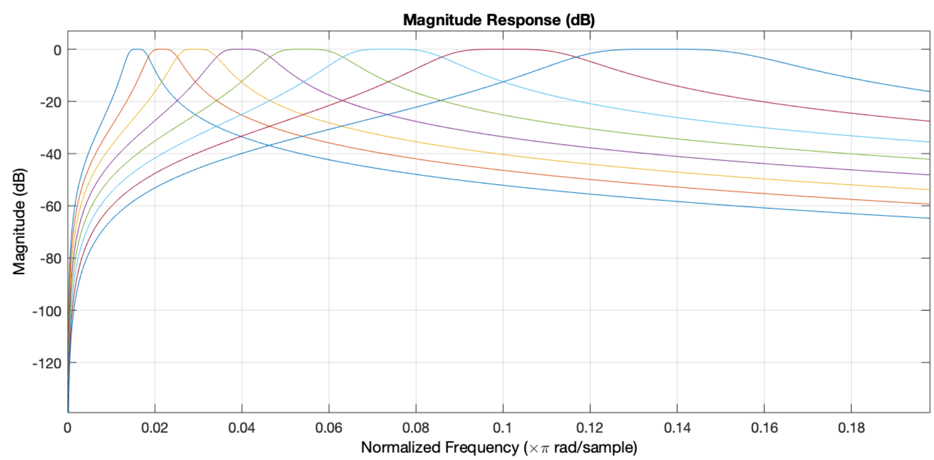


Figure 5: Magnitude response of the filter bank that divides the range 300-3500Hz into 8 bands

3. **Envelope detector = Rectification and Low Pass Filters:** Envelope detection is performed by rectifying the input and passing the resulting signal through a set of low pass filters with corner frequency of 200Hz. The filters are 2nd order Infinite Impulse Response Butterworth Lowpass.

4. **Non-linear mapping:** Amplitudes are quantised. This is because processing is digital and therefore only a finite set of levels is allowed. Mapping of the analogue amplitudes to digital levels is logarithmic since it is more important to distinguish small changes in small amplitudes than in higher ones. Moreover, the range is mapped to the dynamic range of the patient, which may differ from that of a healthy individual. The non-linear compression maps levels from a barely audible level to a very loud comfortable volume.

5. **One active channel:** If two adjacent electrodes are stimulated, the interaction between them will be higher than the separate stimulation of the different sections in the cochlea. Therefore, the perceived pitch will lie in between the two stimulated frequencies, creating a virtual channel. In CIS processing this problem is dealt with by using non-overlapping pulses, therefore stimulating each electrode in turn. In this way, the interaction between electrodes is minimised. Figure 8 illustrates the alternating stimulation of electrodes over time.

   In the simulation, no modelling of negative pulse was done. This is because it effectively stimulates the neurons in the same way and is just used in the practical application to maintain charge balance inside the brain.

   In CIS processing the typical rate is 1500-1800 pulses per second per channel. Total number of pulses is (rate)*(channels). For example, with 8 channels the pulse duration will be 69 µs.

6. **Modulation:** stimulating a certain point on the cochlea with an electric pulses will result in perceiving a sound where such frequency component is present. In order to reproduce this process, the signal is modulated with a sinusoidal at a frequency in the mid-band.

7. **Non-idealities I = time spread:** the stimulation is a spike, but in reality there will be a rise and fall time in the stimulus. Two options were considered to simulate this process (Figure 6).

(a) Simulate the rise time through a linear increase and the fall time with a decaying exponential;
(b) Pass the signal through a low-pass filter with corner frequency of 5000Hz.

Although more realistic, the first option was discarded because it added useless extra complexity to the program, considering that the second option is already a very good approximation and easier to implement.

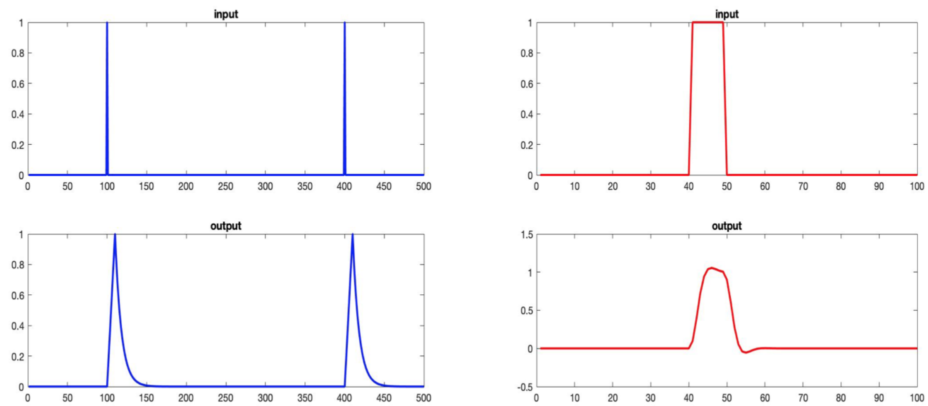
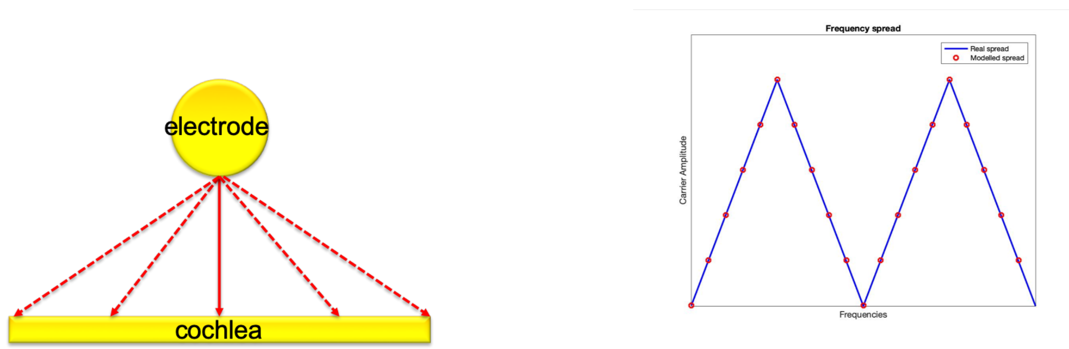


Figure 6: Simulation of spread of the stimulus in time. Option (a) on the left plot, option (b) on the right

8. **Non-idealities II = frequency spread:** When the electrode is active it does not only stimulate the desired frequency, but also a set of frequencies placed around it, as illustrated in Figure 7a. This is simulated by modulating the envelope also at frequencies around the desired one. Such frequencies are logarithmically spaced because of the anatomical structure of the cochlea (notice that the x-axis of the plot in Figure 7b is logarithmic).



(a) Frequency spread of the stimulus

(b) Simulation of the spread
X-axis is logarithmic

Figure 7

The output of the processing stages is shown in Figures 9 (Spectrogram) and 10 (Fourier Transform). From the comparison of the input and output, it can be noticed that the distribution of power at different frequencies is very similar, but the input presents a continuous spectrum while only certain specific frequencies are present in the output. When all channels are active, side-bands are present.

4 Further research and Other technologies

A great deal of research was put into understanding the physical phenomena that occur in the brain in order to simulate the perception of sound at various stages. Although improvable, a good result was obtained in approximating the response of the cochlea to the electrical stimulus. Further work will consist in simulating the adaptation process that happens over the years in the brain, which is one of the main reasons why cochlear systems are effective at all.

Other technologies that are commonly employed were considered. If more time was available, it would be interesting to implement them and compare their performance. The most relevant are:

- FS4 Finite Structure Processing: for low frequencies, not only the information of the envelope is used, but also of the fine structure of the signal. A zero-crossing detector is used to detect the times at which the signal changes sign;

- HiRes120: the signal is simultaneously analysed in the time and frequency domain in order to stimulate the spectral peak in each band instead of the mid-point as happens in the CIS technique;
- SPEAK, which is the same as HiRes120 bus uses the overall spectral peaks instead of the peaks within each band;
- n-of-m: this is similar to the CIS technique. However, the number of bands in which the signal is initially divided is much higher ("m", normally around 20). Then, only the "n" spectral maxima are stimulated (normally 6-10).

5 MATLAB Simulation

The processing was implemented in MATLAB, where a Graphics User Interface (GUI) allows to control all the relevant parameters. The interface is shown in Figure 1. Moreover, additional plots can be produced and the user can listen to the sound at the different stages of the processing. In this simulation the input file is a .mp3 or .mp4 file, inputted in the GUI itself. The output file can be saved as a .wav file.

6 FPGA Simulation

The same processing is to be implemented on a hardware board in order to make a physical device to be used in demonstrations.

A first step involved choosing between a Digital Signal Processor and an FPGA for the implementation of the processing stages. Table 1 shows the main criteria behind the choice. The final decision was to use an iCE40 Ultra Breakout Board with iCE5LP4K FPGA. This is a low-power FPGA that compensates for the major drawbacks of the FPGAs (high power consumption) while maintaining its flexibility advantages.

DSP	**FPGA**
Advantages	***Advantages***
Specific for the purpose	Parallelism →faster
Can find libraries and built-in functions	More flexible
Floating point calculations →more precision	FIR filter design in Quartus Prime
Easier to implement filters	
Disadvantages	***Disadvantages***
Sequential operations →slower	High power consumption
Never used it, but similar to microcontroller	Fixed point arithmetic
Don't have it	Harder to implement filters

Table 1: Comparison of DSP and FPGA performance

For real-time processing of the incoming sound, an ADC is needed at the input and a DAC is needed at the output. A PWM DAC was implemented in Verilog and tested. As far as the ADC is concerned, different options are being considered:

- implementing a Sigma-Delta converter on the FPGA itself (see Figures 11 and 12 for a block diagram of the circuit);
- an external device with serial interface;
- an external device with parallel connection.

Testing was done for implementing the first option. This looked optimal because it did not involve additional hardware but was ultimately discarded because of useless complexity and limited performance. For the connection to the FPGA, the last option is considered optimal and will be implemented.

Further work will consist in implementing all the processing stages on the FPGA in order to have the same algorithm running on a hardware board.

7 Conclusion

The overall purpose of the project was to produce a device to explain the significance between invasive and non-invasive treatments on the brain. The simulation was completed in software and the results are very close to the initial expectations. With the parameters currently used in cochlear implants the output is barely recognisable when the input is speech in a quiet environment. Music and sound in noisy environments is impossible to distinguish. Further work will consist in completing the simulation on FPGA hardware and further developing the processing algorithm to simulate the natural behaviour better and to test other technologies.

8 Figures

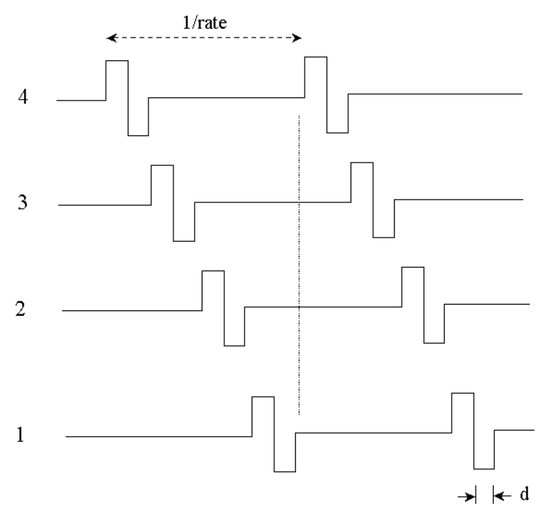


Figure 8: Stimulation of one channel at a time in CIS processing

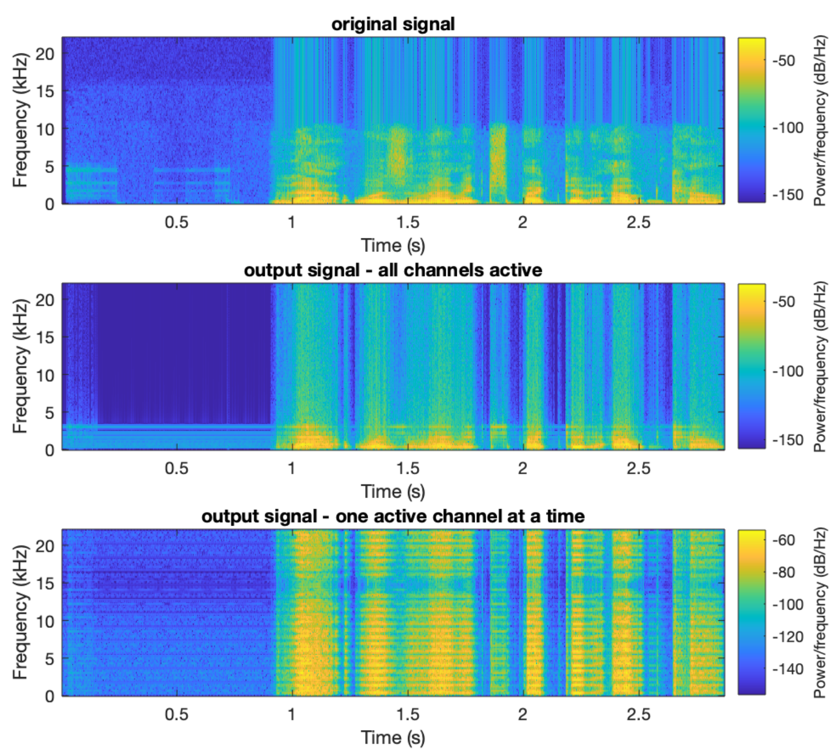


Figure 9: Spectrogram of the Input, Compressed Analogue Output and CIS Output

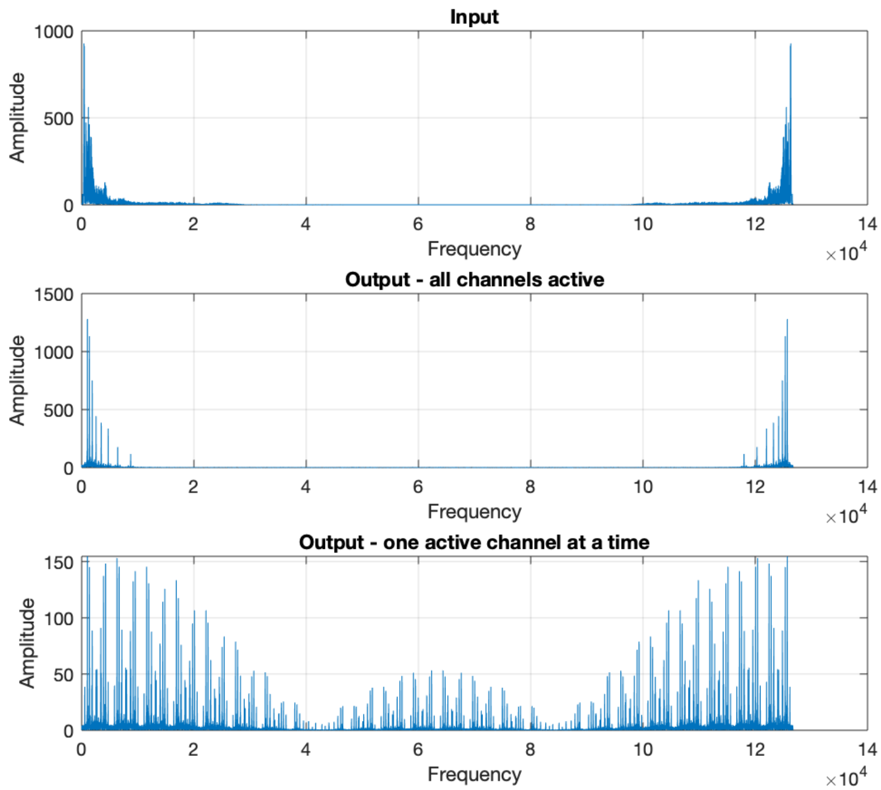


Figure 10: Fast Fourier Transform of the Input, Compressed Analogue Output and CIS Output

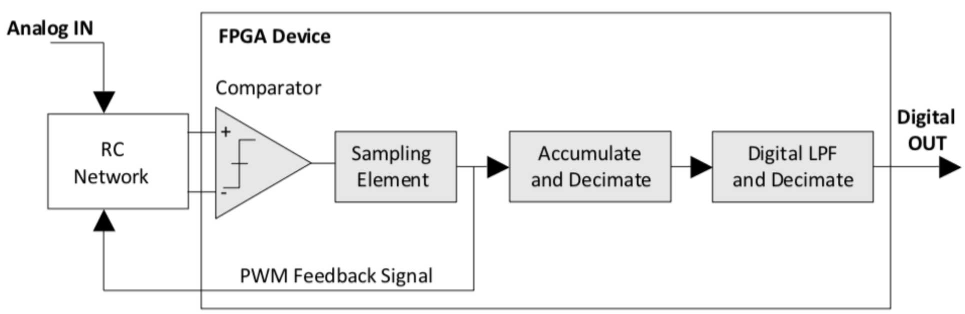


Figure 11: Block diagram of the Sigma Delta ADC

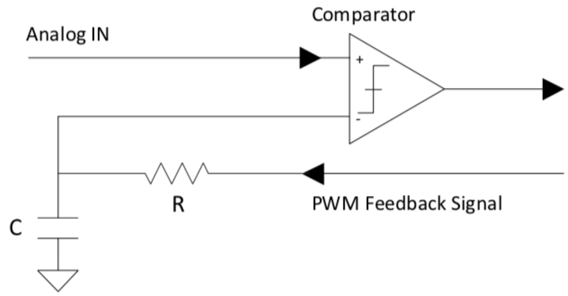


Figure 12: RC Network design of the Sigma Delta ADC

References

[1] Dennis Fitzpatrick. Implantable Electronic Medical Devices. *DOI: http://dx.doi.org/10.1016/B978-0-12-416556-4.00005-X © 2015 Elsevier Ltd.*

[2] Philipos C. Loizou. Introduction to cochlear implants Tutorial article on cochlear implants that appeared in the IEEE Signal Processing Magazine, pages 101-130, September 1998.

[3] Lattice Semiconductors. Simple Sigma-Delta ADC - Reference design. FPGA-RD-02047 Version 1.5, September 2018.