Cochlear Implant Processing MATLAB Simulation Program - Phase-III - Final Product & Wrap-up

A Report Submitted in Partial Fulfillment of the Requirements for BME 252 Project

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1. Summary of Phase I & II

One of the major components of a cochlear implant is a signal/speech processor in which there is a transformation of collected sound signals to electrical signals. It is a system to transmit electrical signals to implanted electrodes and to the brain for interpretation. Phase I of this project involved importing audio files (.WAV format), summing stereo signals to mono, and downsampling the audio file to 16 kHz. In Phase II, the downsampled sound signal was divided into different frequencies by eight bandpass filters (6th-order Butterworth) between 100 Hz - 8 kHz. An increment of 987.5 Hz was obtained by dividing the entire range of 7900 Hz into eight channels. The cutoff frequencies for bandwidths between said range with increments of 987.5 Hz were obtained by implementing a for loop in MATLAB. The physical frequency was normalized by dividing it by the Nyquist rate, which is half the sampling rate (π rad/sample). Next, the waveform was transformed into an envelope signal using rectification (abs()' command) and low-pass filtering ('filtfilt()' command) with a cutoff frequency of 400 Hz.

2. Evaluation Method

A wealth of information can be obtained from a sound signal, regarding the content of the message. The betterment of sound signals depend on many variables such as context, tone, background noise, distance from source, etc. In order to determine the performance accuracy of the cochlear implant processing simulation, the following criteria will be tested.

2.1 Clarity (Bonus)

An extra evaluation index that quantitatively assesses the design is clarity. In general, clarity depends on the range of pitches of sounds, with great emphasis on particular frequencies. In Figure 1 below, the orange coloured signal is the original signal, while the blue coloured signal is the filtered noisy sound signal using the envelope discussed in the testing section of this report. From Figure 1, it is evident that an appropriate evaluation method could be the use of signal-to-noise ratio to assess the signal and its clarity quantitatively. This ratio helps measure the noise that is being detected in the signal (in other words, the noise power) to the desired signal outcome (in other words, the signal power) using quantitative measurements in decibels [1]. A ratio that is greater than a 1:1 ratio would indicate that there is more signal than noise, and by the amount of decibels [1].

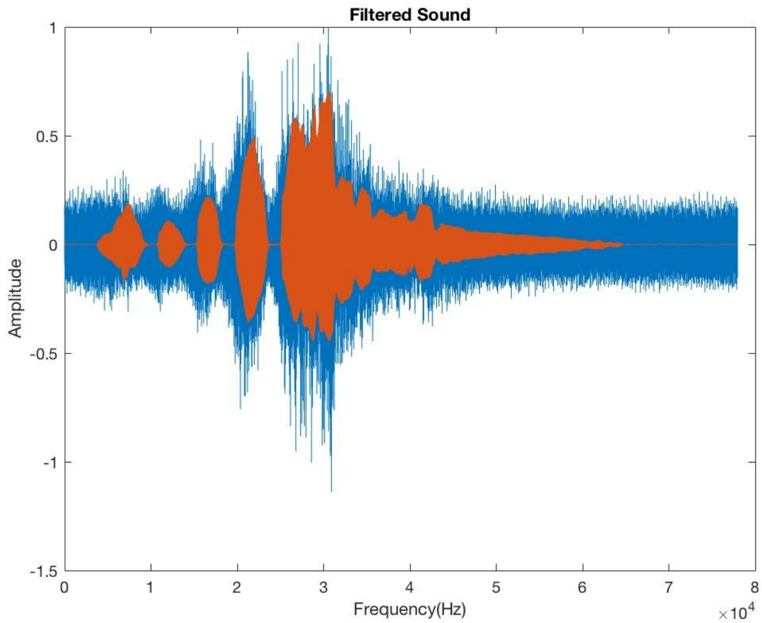

Figure 1 : Noisy Compared to Original Signal

2.2 Intonation

Ideally, a cochlear implant output should be able to provide voice tone information. This is critical for listeners' recognition of prosodic contrasts of speech (e.g., pitch, intonation, timbre, length of sound, loudness, etc). Prosody reflects many aspects of a speaker's emotional state or form of occurrence (question, command, advice, etc.). In this evaluation technique, one aspect of prosodic speech - intonation will be studied. Multiple sound files containing sentences with varying tones can be used for testing. The sentences are to be presented to the subject, who will then be encouraged to determine the tone of the sentence. If the hypothesized tone matches the tone of the original sound, the design passes the evaluation.

2.3 Sharpness

Figure 2 below has an orange signal which is the original sound signal, and the blue represents the sharp signal after filtering. Sharpness of the amplitude response or intelligibility is an evaluation method that can be used to quantitatively assess the design. For the purpose of this experiment and project, sharpness will be evaluated qualitatively based on visual observation of the signal, as shown clearly in Figure 2. The sharpness of a signal can very easily be detected by the blue peaks in the Figure and usually results in an abnormally robotic sound. Thus, if it is evaluated to be a "1", it indicates that the signal is not sharp, and thus, ideal. This can also be measured quantitatively using the bandpass filter value "Q", which is the ratio of the centre frequency to the -3 decibel bandwidth or frequencies [3]. The equation for Q = Fc/(Fh - Fl) or in other words Q is center frequency divided by higher frequency - lower frequency or bandwidth [3]. This gives the performance of the filter in terms of sharpness, which also gives the attenuation at any particular instance of the centre frequency [3]. The intelligibility of the signal can also be measured qualitatively using the comprehensibility of a certain number of words or consonants. If certain sounds of the sound signal and consonants such as "s" or "f" can be heard clearly/not robotically, and if words can be heard correctly, the qualitative measure of clarity and intelligibility could help determine and evaluate the signal [5].

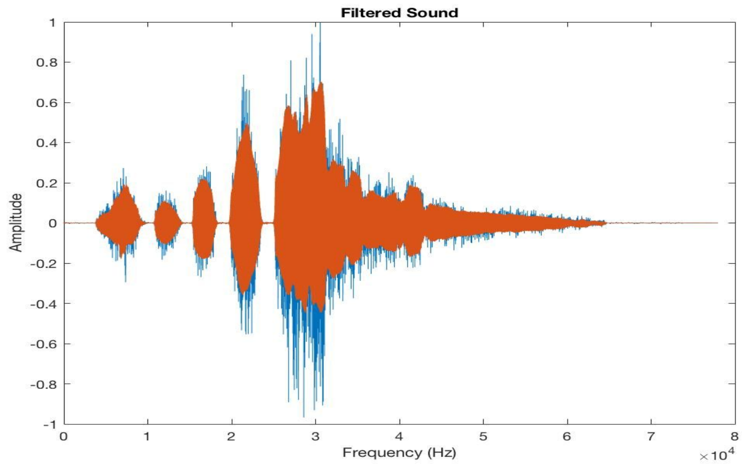

Figure 2: Sharp Signal Compared to Original Signal

3. Tests & Results

To best determine the criteria at which the functionality of the cochlear implant was optimal, multiple tests were performed that involved varying multiple variables in a trial and error method. This helped in determining how varying each component of the implant would affect the overall output sound.

3.1 Bandpass Filters

Initially, we opted for built in matlab filters for simplicity. However, it was essential to explore how different filtering methods would affect the overall output sound. With the combination of the butterworth filter and the “filter()” command, it was noticed that there was a delay in time between the original signal and the output signal. While real-time processing is not within the scope of this project, we wanted to avoid a delay in the beginning steps of the filtering process. Through research, it was soon discovered that the “filtfilt()” command is a better alternative because it provides the same functionality as the “filter()” command without delaying the signal in time. Also, the “filter()” command only works in instances of a low or high pass filter but is not compatible with bandpass or notch filters where two cut-off frequencies are selected. Thus, we opted for using the built-in “filtfilt()” filter.

Then, we compared the use of IIR (infinite impulse response) filters and FIR (finite impulse response). For the IIR filter, the butterworth filter was chosen as opposed to the chebyshev due to the flattening property that comes with the butterworth filter which eliminates any bandpass ripples. The “butter()” command in collaboration with the “filtfilt()” command was used to produce each filtered channel. The FIR filter used the command “fir1()” with the “filtfilt()” function. Plots of the varying filter types can be seen below.

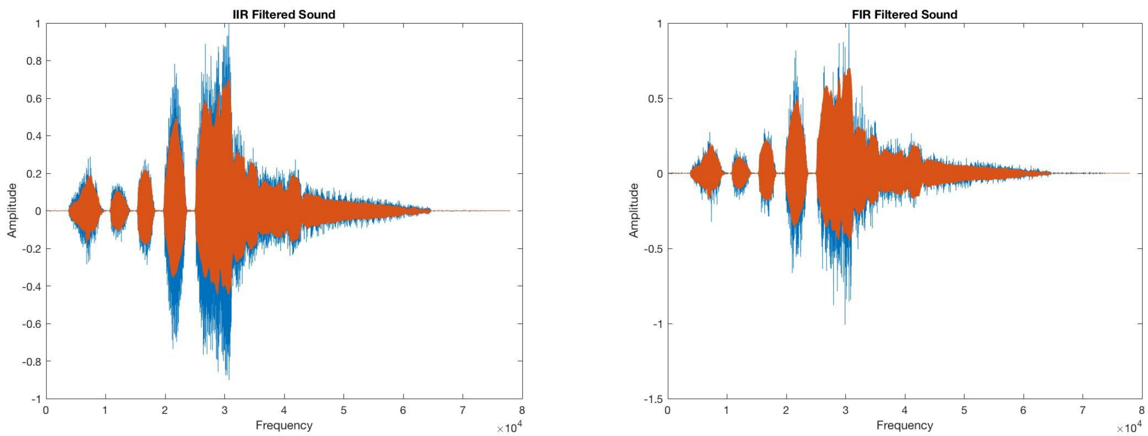

Figure 3: IIR Filtering versus FIR Filtering where the Orange is the Original Sound and the Blue is the Filtered Sound

In order to determine which type of filter was better, we referred to our evaluation methods. First method was the clarity or the noise amount of the signal. It is important to note that for the sake of consistency the test was performed at on order of 6 for both types of filters. The filters in comparison to our evaluation method can be seen in Table 1. The scoring was developed on a 0 to 1 scale where 0 indicates that the criteria was not achieved and 1 indicates that it was.

Table 1: Comparison of the Filter Types with respect to the Evaluation Methods

Filter Type	Clarity	Tone	Sharpness	Total
IIR	0	1	0	1
FIR	1	0	0	1

Looking at Figure 3, it can be observed that with the IIR filter, the output signal (blue) was noisier than its FIR equivalent as it contained more scattered peaks that do not follow the overall sequence of the original noise. Furthermore, both filters failed the sharpness criteria because they resulted in sharper peaks than the original sound which would have caused a more high pitched and robotic final sound. However, the IIR filter had a better tone that closely resembled that of the input sound as opposed to the FIR filter. For the sake of simplicity, and because our group had a better understanding of IIR filters, we opted to use these instead.

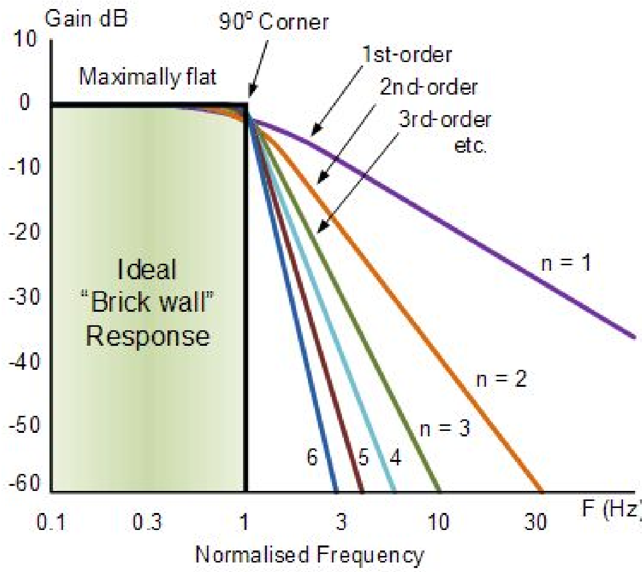

Figure 4: Normalized Frequency (Hz) vs. Gain (dB) and Order Effects

Next, we compared the order of the filters to visualize its effect on the output sound. Higher order typically gives more attenuation (reduction of the intensity of the signal) between passband and stopband [1]. It also provides less ripple which preserves the purity of the DC output of a rectifier [1]. Additionally, higher orders help create a narrower transition band increase roll off which will more accurately define the band and in general, frequency spectrum [1]. In summation, higher order filters are sophisticated and more closely replicated to the ideal/desired outcome when looking at the accuracy of signals, which can be visually observed through Figure 4 above [2]. Although high order filters are preferred, there also some fundamental downsides to the order. One downside of using a higher order is that it may result in unstable filter signals through the frequency response [2]. Higher orders are a series of cascaded second order circuits depending on the order number [2]. Thus, another downside is that it takes more processing per sample, making it slower and resulting in the processor potentially missing some samples [1]. With the IIR filter, any orders higher than the sixth order for the filter gave an error and the filtering did not work, as there was no display in signal which may have been due to missing samples due to processing as explained earlier.

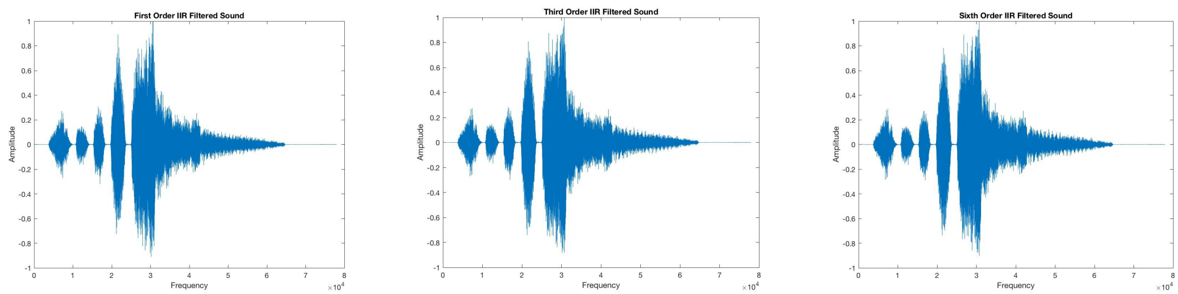

Figure 5: Output Sound with Different Order IIR Filters

Overall, changing the order of the IIR (butterworth) filter had little to no effect on the overall output signal and sound. Through testing, it was observed that FIR filters with an order that is less than 15 behaved similarly. An order higher than 15 going up to a 100 also had a similar behaviour. With an order of 10, for instance, it was observed that the output sound became very low-pitched and robotic. It was closely compared to a very loud noise rather than the sound of a person. With an order of 30, for instance, the noise became more high pitched and less robotic. It also closely followed the same tune as the original sound. Table 2 shows how the orders measured up to our evaluation methods.

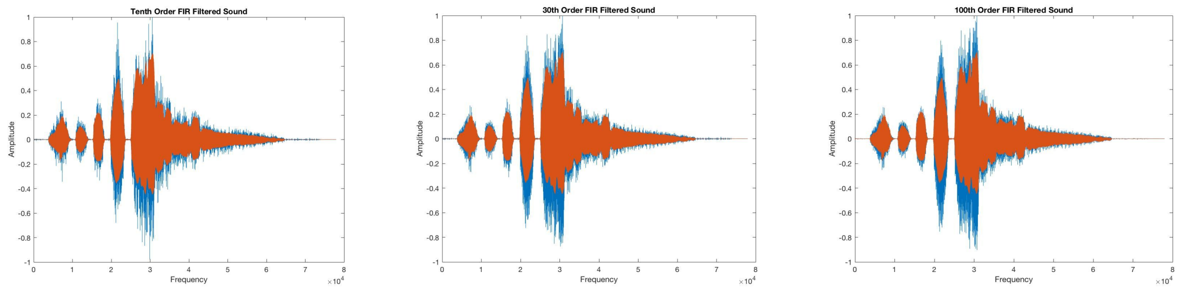

Figure 6: Different Order FIR Filter (blue) Compared to the Original Sound (orange)

Table 2: Comparison of the Order Number with Respect to the Evaluation Methods

Order Number	Clarity	Tone	Sharpness	Total
<15	1	0	0	1
>15	0	1	1	2

Through Table 2 and Figure 6, it is clear that orders above 15 produces the better output sound. However, since we had opted against choosing FIR filters for the aforementioned reasons, the results obtained from this test were used to determine how changing the order of the filter generally affects the output signal and sound.

When testing to determine whether linearly spaced cut-off frequencies were better than overlapped cut-off frequencies, it was found that there were little to no difference between the two instances. Through multiple iterations, spacing and overlapping of bandpass filters varied through each test, and our group discovered that changing the bandwidth of each cut-off frequency did not affect the overall signal. With each test, the number of bandpass filters was kept constant at 8 channels in order to ensure consistency across all tests. Figure 7 shows bandpass filters where the bandwidth of each one was set to be 500Hz (left) as opposed to equidistant bandwidth of 987.5Hz across the bandpass filters.

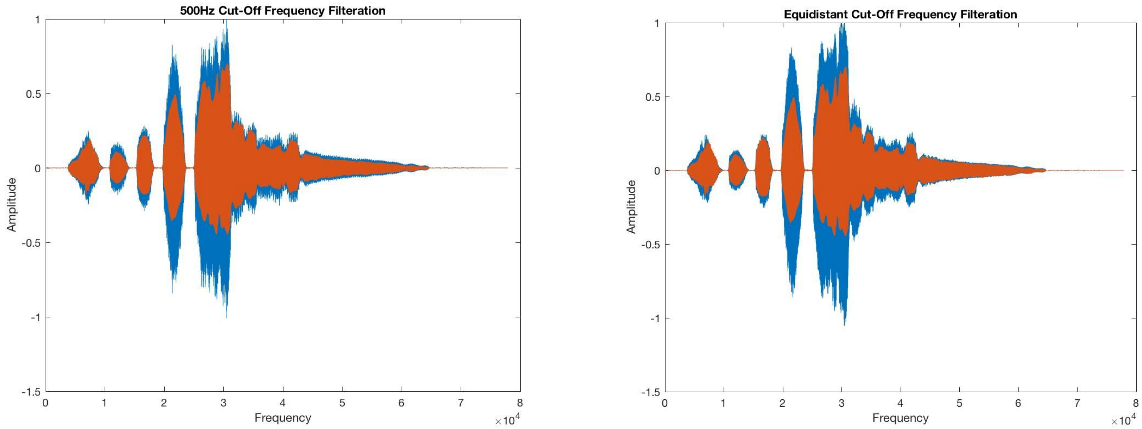

Figure 7: 500Hz Bandwidth Bandpass Filters vs. Equidistant Bandwidth Bandpass Filters

In Figure 7, the orange signal represents the original input sound, and the blue signal represents the output filtered sound. It can also be observed that there is little to no difference between each graphs. However, it was essential to compare the plots to our evaluation methods to make the final choice.

Table 3: Comparing Bandwidth Sizes of Bandpass Filter with Respect to the Evaluation Methods

Bandwidth	Clarity	Tone	Sharpness	Total
500 Hz	1	1	0	2
Equidistant	1	1	1	3

Table 3 shows that linearly equidistant bandwidth across the number of bandpass filters produced a better signal. Through Figure 7, it can be seen that equidistant linear bandwidth of the bandpass filters produced smoother signals which resulted in a more human-like output voice with a more natural pitch. In terms of clarity and tone, both figures scored the same because they both followed the overall pitch of the input sound and were relatively noise-free. Thus, it was determined that linearly equidistant bandwidth across all the bandpass filters resulted in a better signal.

In order to determine the optimum number of channels further testing was completed. Now that all the other parameters were set, it was essential to determine how changing the number of channels affected the output sound. Multiple tests were performed where the initial one started at 8 channels and the final one had 50 channels. The overall trend noticed was that as we approached 30 channels the signal was smoother (less sharp) and as a result the output was more human-like and natural as opposed to robotic and very sharp. It was also noticed that as we got passed the 45 channels benchmark, the signal get very high pitched and sounded more robotic again. Thus, it was determined that the optimal range of channels would lie around the range of 30. However, it was essential to reach this conclusion with the help of our evaluation methods. Figure 8 shows the output sound at 8 channels, 30 channels, and 50 channels, respectively.

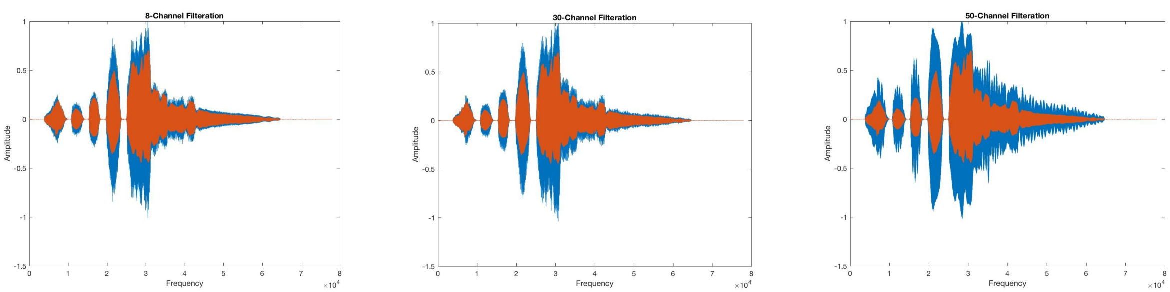

Figure 8: Different Number of Bandpass Filter and their Effect on the Output Sound

In Figure 8, the orange signal represents the input sound and the blue signal represents the output filtered sound. Each plot shows a varying number of bandpass filters as labelled by its title.

Table 4: Comparing Different Number of Bandpass Filters with respect to Evaluation Methods

Number of Bandpass Filters	**Clarity**	**Tone**	**Sharpness**	**Total**
8	0	0	0	0

30	1	1	0	2
50	0	0	1	1

The signal with 8 bandpass filters scored 0 across all categories because it was the worse signal compared to the other two. It contained the sharpest peaks and the second-most noise. Also, the output sound it produced was very sharp and unlike the input sound. It also sounded very robotic and thus would not be suitable for the implant. The signal with the 30 bandpass filters was the clearest of the three as it contained the least amount of noise. It also had the best tone as it closely resembled the pitch of the input sound and sounded very human-like. It resulted in a soft sound that would not cause any discomfort to the user. However, it did not contain the smoothest peaks but had relatively sharp ones. The last signal with 50 bandpass filters contained the most noise and was the least clear. It was very high pitched and did not match the original sound. It did, however, have the smoothest peaks but because it failed in other criteria it was not the optimum number of bandpass filters. Overall, our findings indicated that 30 bandpass filters provided the best sound. However, that does not necessarily mean its the most optimum.

When choosing the most optimum number of bandpass filters, it was important to also consider other factors like cost, size of implant, and battery life. As number of channels increases, the output sound is better but the size of the implant and its cost also increases. Processing time of the sound would also increase worsening the battery life. Generally, most cochlear implants have around 12-24 channels [4]. Our findings showed that around the 30-channel mark provided the best output sound. But to match what is already in the market and to ensure that an optimum number of channels is selected such that it does not affect the size or battery life of the implant, we opted for performing further testing in the range given by market implants. Through this final phase of testing, we discovered that channel 22 provided the best output sound (compared to lower channels) that closely reselmebed the pitch and tone of the input sound within the given confines. Also, it provided the a much better processing time as opposed to the higher channels. Also through this phase, we discovered that lowering the order of the filters resulted in a better tone while simultaneously decreasing processing time. While we had initially mentioned that changing the order had little effect on the output sound, that was mainly because it was limited by the number of bandpass filters which were initially set to 8. As we explored different options for the number of channels, we observed that the lower the order the more similar the output sound was to the input sound, in terms of tone, and the less robotic the sound became. Also, the processing time decreased. Through multiple iterations, we found that order 3 was the most optimum in providing an output signal with similar tone to the input. It also provided the clearest and least sharp signal as compared with higher and lower orders.Thus, we concluded that the most optimum number of channels was 22 with an order of 3.

3.2 Low-Pass Filter

As the frequency increased above 400 Hz, the sound signal sounded very automated, and tart. As the frequency decreased below 400 Hz, the sound signal became more ideal, and closer to the desired frequency response. The lower frequencies (such as frequencies as low as 50 Hz) typically gave a better and more sharp response than the sound signal at 400 Hz. The plots below in Figure 9 display the signals with low frequency at 50 Hz, medium frequency at 400 Hz, and high frequency at 1000 Hz in low pass filtration. The original signals in orange were compared to the filtered signals in blue.

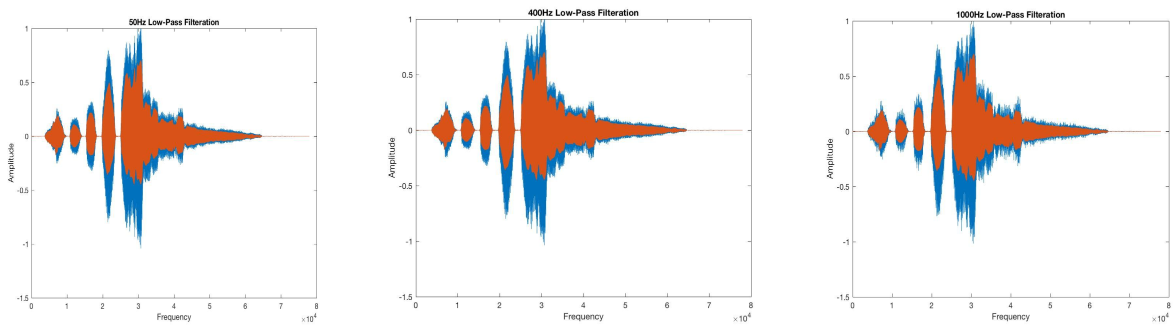

Figure 9: Various Frequency Alterations and their Effect on the Output Sound Signal

Table 5: Comparing Different Number of Bandpass Filters with respect to Evaluation Methods

Frequency (Hz) in Low Pass Filter	Clarity	Tone	Sharpness	Total
50	1	1	0	2
400	0	0	0	0
1000	0	0	0	0

Although the signals visually look quite similar, the auditory feedback presented otherwise as the sound changed with higher and lower frequencies as explained earlier. In addition, clarity could somewhat be seen to be the most apparent in the 50 Hz signal when carefully comparing the orange to blue signals in Figure 9 which is why it was ranked a 1 in Table 5. To add, the sharpness of all signals was quite poor as seen in Figure 9, and the tone when auditorily observed was significantly enhanced in the lowest frequency signal which is why it was ranked a 1 in Table 5.

3.3 Amplitude Modulation

The only variable in this part of the process that needed to be tested was whether the use of the envelope of the low-pass rectified signal was better than using the low-pass rectified signal itself. Figure 10 and Table 6 show the results of this test.

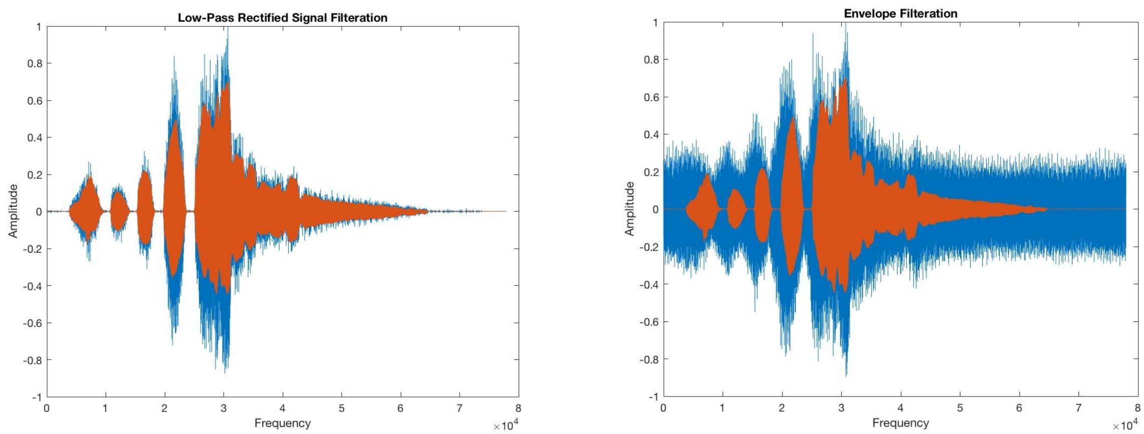

Figure 10: Different Signal Type Filtration (blue) Compared to the Original Sound (orange)

Table 6: Comparison of the Signal Type Used for Filtration with Respect to the Evaluation Methods

Signal Used	Clarity	Tone	Sharpness	Total
Low-Pass Rectified	1	1	1	3
Envelope	0	0	0	0

Both Figure 10 and Table 6 prove that the use of the low-pass rectified signal yielded a better output sound. It was much less sharper and noisier than its envelope. Also, the output sound sounded more natural and followed the same pitch as the original sound.

4. Conclusions

Through multiple iterations, we determined the overall final design to have the following parameters. The bandpass filters will have an order of 3 and will be numbered at 22. Their bandwidths will be equal and linear starting at 100 Hz and ending at 8000 Hz. IIR filters in combination with “filtfilt()” built-in MATLAB filter. The cut-off frequency of the low-pass filter will be set at 50 Hz and the rectified low-passed signal will be used to perform the amplitude modulation instead of the envelope of the signal. Figure 11 shows how our final filter design compares with the original sound. The original sound is shown in orange and the final sound is shown in blue.

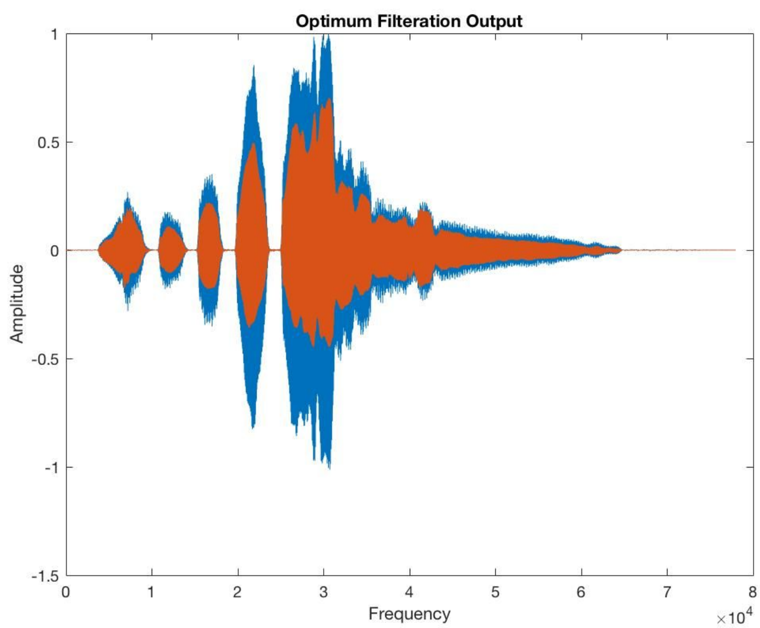

Figure 11: Final Output Sound Signal

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