Cochlear Implant Project

Phase 3

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Tasks 10 and 11

The signal was generated for each of the 22 channels by creating cosine functions that oscillate at frequencies matching the central frequency of the associated bandpass filter channels. The 22 cosine functions were created using a *for* loop that iterated through the channels, and then stored in a single variable ‘x’. Each of the 22 waveforms were then multiplied by the same rectified low pass filter from task 8, resulting in various amplitude values for each cosine function.

Task 14: Evaluation

The evaluation method proposed in Phase 1 was an evaluation matrix, in which various metrics were scored and then compared with each other. A version of this matrix is shown in Table 1 below, in which the non-iterated designs are scored for each of the metrics. Some of the metrics in the matrix were changed slightly from those initially proposed in Phase 1, due to a need for increased amount of distinction between the various processor designs. The original matrix from Phase 1 did not have enough individual metrics to guarantee a distinct difference between overall scores.

It should be noted that previously, in Phase 2, only IIR filters were selected to be compared, since IIR processors are known to have faster processing times than FIR filters and it could be seen from our tests that the phase instability that is usually seen in IIR filters was qualitatively insignificant. It was also decided early on in the process that, in the context of creating a signal processor for a cochlear implant, the speed of processing was a priority, so therefore FIR filters were not considered. However, for Phase 3 it was determined that in order to accurately evaluate all options, at least one or two FIR filters should be created and compared. This would allow for the verification that the computational time of an FIR filter is actually large enough to validate our decision to eliminate it in Phase 2. It was decided that both high and low order FIR filters would be created to evaluate alongside the 4 IIR filters from Phase 2; the reasoning for this was that in order to produce satisfactory cutoffs and therefore ensure frequencies outside of the bandpass frequency range were rejected, a higher order may be necessary. 2 FIR filters were created of type Least-Squares, with the first having an order of 8 and the second having a much higher order of 200.

The 6 processors being evaluated in this matrix are therefore the Butterworth, Chebyshev, Chebyshev 2, Elliptic, 8th order Least-Squares, and 200th order LeastSquares filters, respectively. The processors were scored out of ten for each metric, and then converted to a percentage value and multiplied by the associated weight. The "quality of sound" metric was still weighted the highest as the overall subjective score was considered the most important metric from a consumer perspective. The explanation of the individual scores displayed in Table 1 and why they were awarded is as follows:

For the quality of sound metric, the output sound file of the processor was played for three test subjects. They were asked to rank the overall quality out of ten, and the results were averaged. Therefore, this score is a purely subjective score. For the second metric, a similar method was employed; various sounds were played through the different processors and the participants were asked to identify the type of sound, such as a lion roar or an alarm clock beeping. This is a more objective method of determining the quality of the output.

For the third metric, an initial sound was played for the test subjects and then the same sound file was played via the various processor designs. The test subjects were asked to determine which processor produced an output closest to the original file in terms of loudness, making this a subjective score. For the fourth and fifth metrics, another similar testing method was employed; test subjects were asked to determine when a pitch change occurred and when a type of sound changed. The number of changes they correctly identified (out of 10) were recorded, making this an objective score. Again, the results were averaged across all three participants.

Cross correlation was evaluated by using the built in MATLAB function in order to visualize the lag between the input and output at each specific sample. Each cross-correlation graph can be produced with the provided MATLAB code. The graph visualizes the correlation coefficient and the lag between each individual point. To relate to the course, the cross-correlation function is analogous to convolution except in cross-correlation the system response does not need to be time-inverted. The value of the cross-correlation graph at lag = 0 shows how similar both respective signals are when there is no lag or delay between them. This value at $x = 0$ was normalized to put it along a scale of 1-10, and then multiplied by the weighting to help determine the final score value. The cross-correlation score for the FIR Least-Squares 200th order filter was simply assigned a perfect score of ten, as it was a very large outlier due to the incredibly high order.

Table 1: Evaluation Matrix for Initial Signal Processor Designs

Metric	Weight	BW	ChB	ChB 2	Elliptic	LS 8th	LS 200th
Quality of sound	0.2	4	6	4	8	8	7
Determination of type of sound	0.18	4	5	2	7	9	6
Distinction between loudness	0.16	5	3.67	3.33	6	7	5
Detection of sound changes	0.16	5	4	4	9	9	7
Distinction between higher and lower pitches	0.18	7	6	2.67	8	8	5.67
Cross correlation	0.12	2.62	2.223	1.97	4.147	4.959	10
Total (Max 10)		4.69	5.27	3.04	7.20	7.82	6.62

The numerical results of the evaluation matrix in Table 1 show that the design containing the 8th order Least-Squares filter received the highest overall score, therefore making it the best design of the 6. From deeper analysis of the results of the evaluation process in Table 1, as well as more in depth conversation with the testing participants, various trends and observations could be concluded about the different filters. For example, it was determined that the Chebyshev filter had a much duller tone compared to the Butterworth and Elliptic filters, and the Chebyshev 2 filter produced an almost inaudible output with most of the higher frequencies in the input files being lost. Additionally, the Elliptic filter produced a fair amount of reverb in the lower register, which, although not ideal, was determined to be preferable compared to the higher frequency reverb that was heard in the output of the Butterworth filter processor. Finally, the 8th order FIR filter produced decent quality output, the best of all 6, but the higher order FIR filter was much quieter and more muffled than the 8th order one, which was unexpected. Although the 8th order Least Squares filter is concluded to be the best design based on the evaluation above, it is also known that FIR filters are computationally slower than IIR filters and so it was decided that the second highest score, the Elliptic filter, would also be iterated in order to further compare the two. The iteration process was undertaken with the goal of improving the two processors and then making an informed decision about the processing performance vs. computation time. If the Elliptic filter ended up scoring better after iteration, then it might be a better recommendation to use it instead of the FIR filter simply due to the improved computation time.

Task 15: Iteration

In order to iterate the design after hearing the final sound output, there were various considerations that had to be made depending on the iteration. The filter types had already been tested in Task 14, and it had already been concluded that the Elliptic and 8th order Least Squares filters were the best of the 6 options tested. Additional possible iteration methods therefore included altering the spacing between the sub bands, overlapping the sub bands, and altering the cutoff frequency of the low pass filter. All three of these methods were explored and tested, and the results are explained below.

In phase 2, the purpose of filtering the envelope detector was to smooth the final result of the rectified signal. As can be seen from the results in Phase 2, the filter was not perfect and the output still exhibited rippling, particularly in the lower frequency inputs. Reducing the cutoff frequency of the envelope detector low pass filter would result in a smoother output, but would also decrease the high frequency response. Conversely, increasing the cutoff frequency would increase the high frequency response, but also cause more rippling or fuzziness in the output. In phase 2, the lowpass filter was created using a cutoff frequency of 400Hz as required. In phase 3 testing, it was found that in general, increasing the cutoff frequency of the low-pass filter caused the output to become louder, but increasing it too far started to introduce high frequency noise. Therefore, it was determined that a cutoff frequency of 550 Hz produced the best result, producing the least amount of ripple while still retaining an acceptable high frequency response.

It had been determined in earlier phases, through both research and experimentation, that the asymmetrical sub band spacing already implemented was the best arrangement in terms of producing a coherent output signal once the bands were passed through the bandpass filter and summed together again. However, it was decided that the spacing would be altered again in phase 3 and tested to determine if this was still true. It was found that most combinations of asymmetrical bands did not produce as good of an output as the original, but then when the bands were returned to being completely equidistant rather than asymmetrical, the output quality slightly increased. Additionally, when the bands were kept equidistant and then also overlapped with each other by 300Hz per band, the quality of the output was even more improved; although it was slightly quieter, it was smoother, and the high pitched “chime” sound that was heard previously was now reduced. Also, when additional factors were changed, such as the lowpass cutoff frequency and the different types of filters, this overlapped sub band spacing arrangement was the only version that produced a consistent quality, and was therefore determined to be the best option.

After performing these iterations, the best two filter options from the original evaluation process in Task 14 were once again evaluated, with the iterations included. The purpose of this second evaluation was threefold: to ensure that the iterations did indeed improve the output, to quantify how much it was improved, and to determine a final recommended design.

Table 2: Evaluation Matrix for Iterated Signal Processor Designs

Evaluation metric	**Weight**	**Elliptic**	**LS 8th**
Quality of sound	0.2	8	7
Determination of type of sound	0.18	8.67	7
Distinction between loudness	0.16	9	8.67
Detection of sound changes	0.16	10	10
Distinction between higher and lower pitches	0.18	10	10
Cross correlation	0.12	4.005	9.619
Total (Max 10)	1.0	8.48	8.60

As can be seen in Table 2 above, the results of the same evaluation matrix on the two iterated filters, one IIR and one FIR, show that both processor designs improved, and the LS 8th order filter has a slightly higher total score. Therefore, despite the marginally quicker computation time of the Elliptic filter, the 8th order Least-Squares filter is concluded to be the better processor after all.

Task 16: Effect of the number of channels

By evaluating the output of the 8th order Least-Squares filter processor with different numbers of channels, the optimum number of channels was decided to be 21. It could be seen in testing that the higher the number of channels, the more precise the processing became and therefore the output increased in quality. However, it is important to consider the cost of manufacturing a device, as well as the overall size and usability, so fewer channels would allow for a smaller and less expensive device. The lowest number of channels that produced an acceptable level of quality (as determined by testing with various volunteer test subjects) was 18, but this quality rating was not consistent across all test subjects until 21 channels was reached. 21 channels is around the average for signal processors in current market cochlear implants, which therefore means that this number of channels is likely not so high that the cost becomes astronomical or the physical design becomes too bulky or heavy.

Final Design:

After evaluating and iterating various aspects of the signal processor, the best and final design was considered to be the processor containing an 8th order Least-Squares FIR filter, with equidistant but overlapping frequency bands divided into 21 channels, and with an envelope created by the application of a lowpass filter with a cutoff frequency of 550 Hz. The various input sound files used and an example of an output sound file passed through this final processor design is included in the attached .wav files.

Bonus Task: Evaluation Based on Root Mean Squared Error (RMSE)

After having learned more about signals and signal processing, and after extensive research on the subject, it is decided that another useful method of evaluating the signal processor's performance would be by analyzing the root mean squared error (RMSE) between the input and output signals. The RMSE is another method for qualitatively determining the difference between the input and output signals. The RMSE formula does exactly as its name states; it calculates the root mean squared error between the input and output signals for each respective sample from the input and output files. Comparing one of our poorer performing filters, the Chebyshev filter, with our best performing filter, the 8th order Least Squares filter, these filters produced RMSE values of 12.59% and 7.44%, respectively. This therefore shows that the RMSE value is a valid method of evaluating processor performance.