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Good morning ladies and gentlemen, my name \_\_\_\_\_\_. Today I will be presenting the findings from a feasibility study toward the design of a low cost, adaptive hearing aid.

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The objectives and specifications of the device will be explained followed by an overview of the system using a high level block diagram. Thereafter, a comparison between the hardware and software implementations of the device will be given and the functionality of the device will be highlighted. The results acquired from testing the device are discussed and results of an error analysis is included. Finally, future recommendations of work required to improve the device are highlighted.

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The main objective of the study was to develop a hearing aid device that has the functionality of the hearing aids in existence, but at a fraction of the cost. The device must have the ability to apply compensatory amplification to match an individuals audiogram as well user tuneable directionality. The approach of the project was such that the adaptive hearing aid was implemented fully as a software simulation demonstrating the features of compensatory amplification and directionality. The hardware solution was created with the sole purpose of proving the concepts demonstrated in the software. The project would be considered a success if these criteria were met.

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This is the high-level system block diagram of the designed hardware hearing aid proof-of-concept. The developed system starts with a sound signal being acquired in real time by a microphone array. The array consists of four omni-directional microphones.

The analog output of each microphone is passed through an analogue filter bank consisting of two bandpass filters. The output voltages from each filter bank are fed into eight ADC channels of an Arduino Due. If the device is set to omni-directional mode, the two sampled signals originating from one microphone are independently amplified by specified gain values and are then summed to form one signal.

Alternatively, if the device is set to directional mode, the two frequency bands from each of the four microphones are amplified and summed, resulting in four separate signals. These signals are delayed with respect to one another according to the direction in which the user wants to hear and then summed to form one signal.

In either mode, the DAC converts the resultant digital signal back to analog where it is processed by a cascaded low-pass and high-pass filter for noise reduction, and then an audio amplifier.

The same process occurs in the software simulation, except that pre-recorded audio is used, thus implying non real time processing. The filter bank implemented in the software also makes use of 16 filters, rather than 2.

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The following table provides a comparison between the software solution and the hardware proof of concept. The software solution processes the full frequency range of speech, while the hardware solution only processes two frequency bands within this range. This implies that output audio from the hardware solution is severely distorted.

The software solution makes use of 10 microphones to demonstrate the effectiveness of using an increased number of microphones to attenuate sounds coming from directions other than the one you would like to hear in. 4 microphones are used to prove the concept of directionality in hardware because this is the minimum number of microphones needed to steer an acoustic beam.

14th order filters were selected in the simulation to ensure a decreased interaction in the stop-bands adjacent filter bands. The hardware solution made use of 2nd order band pass filters. The low order was selected to reduce the cost involved in construction of the filters. In order to decrease the interactions between filter stop-bands, frequency bands that were sufficiently separated in the frequency spectrum were selected.

The simulation had 19 steerable angles, in 10 degree increments. These increments were selected as they represented the maximum resolution with which differences in the resultant audio at different steerable angles could be distinguished.

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The feature of compensatory amplification is dependent on the users audiogram, and as such it is unique for each person. In order for compensatory amplification to be matched to the audiogram, individual frequency bands require different amplifications. In order to separate signals so that different gains can be applied to different frequency bands, a filter bank consisting of band pass filters is required. The image shown on the slide illustrates the individual band responses and overall response of the 16 band 1/3 octave filter bank implemented in the software solution. In this case, a gain of 0dB is applied to each frequency band. 14th order filters were used as the use thereof ensured that each band had a maximum ripple of 1dB and the overall filter response had a maximum ripple of 2.4dB.

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Directionality is necessary to attenuate sounds coming from directions other than the one specified by the user. It was implemented through the use of a delay and sum beamformer. The diagram illustrates the mechanics of the beamformer.

When sound is incident on a microphone array, there is an inherent time delay between when the sound hits each microphone. This delay is dependent on the location of the sound source. This is seen at this stage \*\*\*point\*\*\*, where the time delays between output signals from the microphones are evident. In order to achieve constructive interference when summing the signals, further time delays must be applied to each signal so that each signal is in phase with the microphone signal furthest from the source. As such, sound coming from other directions will be out of phase and will destructively interfere to attenuate the signal.

By changing the applied time delays, sound coming from different directions can be amplified.

The user selects the desired direction for amplification by changing the angle pointed to by a dial.

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In order to test the device, the compensatory gain feature and the directionality feature required testing. The images on this slide illustrate the testing procedures used for testing the hardware hearing aid.

When testing the device, a speaker was placed 50cm from the hearing aid, and pure tones with a frequency of 3.34kHz and 6.00kHz were played. When testing the compensatory gain, the hearing aid was programmed with specified amplifications on the two frequency bands and the output signals were recorded.

When testing the directionality, the hearing aid was placed on a rotating platform with a constant direction selected on the device. The device was rotated in 30 degree increments. At each angle, the two sinusoidal input signals were played from a set direction and the output signals from the hearing aid were recorded. This procedure was repeated for each tuneable angle.

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This slide provides a fast fourier transform of the input to and output from the hearing aid when a frame of audio is passed through the simulation. The grey lines indicate the -3dB cut-off points of the filters in the filter bank. In the simulation, a particular gain is applied to each band based on an audiogram. To calculate these gains, a sample audiogram was selected.

Since an audiogram only has seven readings in the frequency range of speech, the audiogram required interpolation to calculate the required amplifications for all the frequency bands. After the interpolation, the gain corresponding to each filter's centre frequency was selected as the gain for that band.

On average, there is a 1.4% error between the expected magnitude and the measured magnitude. This error is due to the large ripple on the overall response of the filter bank.

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Since the microphone array was designed to have a microphone separation distance of 5cm, it was found that the frequency at which the most precise beam steering was achieved was at 3.15 kHz. This slide shows the response of the hearing aid at three different frequencies within the frequency range of speech, at two steerable angles. At 60 degrees, there are peaks at 75 and 150 degree with equal magnitudes for the 6.3KHz response. This phenomenon is known as spatial aliasing. The spacing between microphones was selected such that this inequality was satisfied. Lambda min was chosen to correspond to the 3.4kHz is the upper bound of telephone frequency, which is the minimum frequency range required to process intelligible speech. For frequencies higher than this frequency range, spatial aliasing occurs.

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The cost to construct the hardware solution of the hearing aid was R1400. The graph indicates an FFT of the output signal from the hearing aid with specified amplifications applied to the two frequency bands. The red graph represents the raw output with no amplification applied. The black graph is when a gain of 6dB is applied to the 3kHz band and 0dB to the 6kHz band, and the blue graph is when a gain of 6dB is applied to the 6kHz band, and 0dB to the 3kHz band. In the graph, there is a clear peak here \*\*\*point to peak in between 2 bands\*\*\*

This peak is caused by the interaction of the stopbands of the filters. By using higher order filters, this peak would be reduced, as was seen in the software simulation.

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This slide shows the response of the hearing aid at two steerable angles with an input signal with a frequency of 3.34Khz.

In order to quantify the performance of the feature, a simulated polar plot was created for each tuneable angle and the simulation was compared to the measured results. This simulation is overlaid with the measured plot in the graphs.

From the graphs, it is clear that the smallest error in directionality is present at 90 degrees. This is because at 90 degrees, the signals from the microphones are not time delayed. At the other angles, the signals from the microphones are shifted by an integer number of samples, when in fact mathematically, they should have been shifted by fractional values. This introduces errors into the feature.

The most prominent errors in each response were seen where nulls were expected in the simulation, but not apparent in the measured results.

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The error analysis for the system is provided in the table. Error in compensatory gain was calculated by comparing the measured gain to the desired gain at different applied frequencies and gain values. From the table, it is evident that amplification of one band causes signification errors in the other band.

Error in directionality was calculated by comparing the mismatch between the measured response and the simulated response. As mentioned before, the smallest error is found at 90 degrees. There is an increasing error as the dial is pointed further away from 90 degrees. This is on account of the non-ideal omni-directional behaviour exhibited by the microphone.

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For future work, a number of improvements should be made. The first is replacing the microphones with higher quality omni-directional microphones. In doing so, the error in directional hearing will decrease, as will that of the omni-directional mode.

Additionally, an integrated circuit chip should be created to perform the pre-processing of the audio signals. This pre-processing includes all the necessary filtering and amplification of the audio signals. By creating a dedicated IC, the entire frequency range of speech could be processed by the hearing aid. It would also allow for the use of higher order filters to reduce the interaction of neighbouring frequency bands, which would improve the accuracy of the compensatory gain feature.

Finally, an essential improvement is to reduce the size of the device. This can be done by embedding all the necessary circuitry into the headband of a set of headphones. This would also make the device easier to use as the user would not be required to wear any extra hardware.

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In this project we have proven that a low cost hearing aid with the ability to apply individual specific compensatory amplification and user tuneable directionality. The proof of concept for the device was produced for under R1460. As such, the findings show that it is feasible to develop this device further.