1

Good morning ladies and gentlemen, my name is \_\_\_\_\_\_\_\_ and this is my partner \_\_\_\_\_\_\_\_\_. Today we will be presenting our findings from an investigational study into the design of a low cost, adaptive hearing aid.

2

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 conducted and its included functionality will be highlighted. A cost analysis of the device is included. The results acquired from testing the relevant functionality are discussed and results of an error analysis is included. Finally, future recommendations of work required to improve the device are discussed and the presentation will be concluded.

3

The main objective of the project is to develop a hearing aid device that has the functionality of the hearing aids in existence, but at a fraction of the cost. In addition to this functionality, the device must have the ability to apply compensatory amplification in accordance with an individuals audiogram as well directionality in a direction which the user would like to hear best. 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 would have the sole purpose of proving the concepts demonstrated in the software. The project would be considered a success if these criteria were met.

4

This is the high-level system block diagram of the designed hardware proof-of-concept hearing aid system. The system starts with a sound signal being acquired by the four omni-directional microphone array in real-time. The analog output of each microphone is passed through an analogue filter bank consisting of two bandpass filters. The output voltages of each filter bank are fed into eight separate ADC channels of the Arduino Due. If the device is toggled in omni-directional mode, the two sampled signals from one microphone are amplified with specified gain independently and are summed to form one signal. Alternatively, if the device is toggled in directional mode the two frequency bands from each microphone 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 summed to form one signal. In either mode, the DAC converts the single channel digital signal back to analog whereby it is processed by cascaded low-pass and high-pass filters and an audio amplifier. The block diagram of the full software solution only differs to this one in that its filter bank has 16 different filters and deals with pre-recorded audio which implies non real-time processing.

5

The following table summarises a comparison between the fully implemented software solution and the hardware proof of concept. Evidently, there are differences between each solution and since the hardware was utilized to merely proved concepts, its overall functionality was somewhat limited.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. \*\*\*\*\*\*basically chat from the table\*\*\*\*\*\*\*\*\*\*\*\*\*

6

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 to others. 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. Notice how by implementing a bandpass filters with an order of 14 results in a decreased interaction in the stop-bands adjacent filter bands. The hardware solution made use of 2nd order band pass filters. In order to decrease the interactions between filter stop-bands, frequency bands that were sufficiently separated in the frequency spectrum were selected. \*\*point to the chosen frequency band in the picture\*\*\*\*

7

8

The overall cost of the device was R1460. While this cost was over the formal budget for the project, it is still a fraction of the cost of a commercially available hearing aid, which costs R30 000 on average. The most expensive element of the hearing aid was the Arduino Due. For future development of the device, the Due should be replaced with a most cost effective microcontroller to reduce the cost of the device.

9

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

When testing the device, a speaker was placed half a meter from the hearing aid, and pure tones of various frequencies were played. When testing the compensatory gain, the hearing aid was programmed with various amplifications on the two frequency bands and the output signals were recorded with various input signals.

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, sinusoidal signals with frequencies of 3.34kHz and 6.00kHz were played from a set direction 50 cm from the centre of the device and the output signals from the hearing aid were recorded. This procedure was repeated for each tunable angle.

10

This slides 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. On average, there is a 1.4% error between the expected magnitude and the measured magnitude. This error is due to the ripple present on the filters.

11

12

The graph indicate an FFT of the output signal from the hearing aid with various 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 inbetween 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.

13