THE ELECTROACOUSTIC DESIGN OF A HANDHELD HEARING DEVICE

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1 INTRODUCTION

1.1 Modern Hearing Aids

Sensorineural hearing loss occurs when hair cells within the cochlea of the inner ear become damaged (NHS, 2015). The damage to the hair cells causes the person to be less sensitive to sound and so their hearing threshold is increased. Often this means that people that are less sensitive to quiet sound still find loud sounds uncomfortable or painful. Their sensitivity to sound is also frequency dependent, meaning that they can hear some frequencies normally but not others. Sensorineural hearing loss may also be due to the auditory nerve becoming damaged (NHS, 2015). Neither modern hearing aids nor a handheld can rectify this kind of hearing loss. Conductive hearing loss is when sound cannot travel from the outer ear to the inner ear and can be due to a blockage of the ear canal, ruptured ear drum or damage to the bones in the middle ear. This kind of hearing loss cannot be rectified by hearing aids using loudspeakers but can be by using bone-conduction to transfer the sound directly into the middle ear (NHS, 2015).

Modern hearing aids are often tailored to each individual user with sophisticated digital signal processing (DSP) to make up for the frequencies the patient is insensitive to. They achieve this by making a specific equalization for each patient with their specific audiogram (Tran, et al., 2015). Filters such as notch filters, compression and band pass filters are also included and are specifically designed to reduce the chances of the hearing aid becoming stuck in a positive feedback loop (see 1.2).

Using DSP rather than analogue filters greatly saves space in hearing aids as many filters can fit onto a single chip. The sound must be converted to a digital signal and then back into analogue for the person to hear. The sound is converted to digital values using an analogue to digital converter (ADC), the digital signal can then have filters applied, then a digital to analogue converter (DAC). However, this technique is relatively expensive when compared to analogue solutions.

The handheld hearing device (HHD) is very loosely based on a hearing aid. As the HHD needs to be used on multiple patients the specific equalization is not a possibility. The size of the device was less of an issue so mechanical methods were explored rather than DSP for feedback control. This in turn made the HHD less expensive to produce.

1.2 Speech Intelligibility

The primary function of a handheld hearing device is to deliver clear, amplified speech from one person to another. Therefore, the microphone and speaker used needed to be able to amplify the frequencies of human speech so that it is intelligible. This is done by choosing the components with suitable frequency responses.

The standard bandwidth for telecommunications is from 300 Hz to 3.4 kHz (Munir, 2012). The fundamental frequencies of human speech, however, are between 85 and 120 Hz for adult males and twice that for females (Eulenberg, Farhad, 2011) (Brixen, 2016) (Other sources state slightly higher, or lower frequencies but for the purposes of this report the fundamental frequency will be taken at 120 Hz.). The full range of speech is between the fundamental frequency and 4 kHz (Brixen, 2016). This demonstrates that the fundamental frequency is not important for speech intelligibility but the harmonics of this frequency are. This is because most of the energy of speech is held within the harmonics. Harmonics are multiples of the fundamental frequency which give the speech its tone.

Vowels are the tonal part of speech and heavily rely on the harmonics of the fundamental frequency whereas consonants are higher in frequency, of 500 Hz or higher. The main frequency band for speech intelligibility is around 2 kHz (Brixen, 2016). This means for a successful HHD, this frequency band must be adequately amplified.

However, just because the speech is intelligible does not mean that it will sound natural. When speaking on a telephone the received speech sounds metallic, this is because the low frequencies are cut from the speech. On the HHD it could be important to include some of the fundamental frequency to make the speech sound natural and so the listeners are not disconcerted by the amplified speech.

1.3 Positive feedback loops

When the amplified sound from the speaker of a hearing aid or the HHD is picked up again by the microphone, it is known as feedback. This is not necessarily a problem as if the amplitude of the feedback is less than the initial sound then the subsequent feedback loops to the microphone will be quieter still and tend to zero. In large systems like public address, this kind of feedback loop may sound like echoes but in the case of the HHD and hearing aids, the distance between the microphone and speaker are so low that this kind of feedback will be hardly, if not at all noticeable to the listener.

Positive feedback occurs when the sound from the speaker returning to the microphone is louder than that of the incident sound. This sound is then amplified and so on until the noise from the speaker is sustained at a maximum. If a hearing aid or HHD shows this kind of feedback it could be painful to the listener and damage their hearing even further.

2 DESIGN PROCESS

2.1 Design Introduction

The design of the handheld hearing device was heavily influenced by its purpose. The HHD needed to be compact so that it could fit into a pocket and be used by one hand. It also had to be effective enough to increase the volume of speech without causing the system to become unstable and feedback. The HHD was designed to include a cardioid type microphone, small loudspeaker, microphone preamp, power amp, AA batteries and a housing. A volume control, power switch, and power indicator were also included in the design. The output of the HHD needed to have as equal gain and as much gain across all the speech frequencies (described in section 1.2) as possible. This is because the HHD was to be used on multiple patients all with different gain needs.

2.2 Component Housing

Most importantly the case needed to be large enough to fit all the components but small enough to be comfortable to hold and use with one hand, and be able to be fit into a pocket or be wearable in some way. The spacing between the microphone and speaker dictated the level of gain possible before the system became unstable with feedback. This spacing also had a direct effect on the comb filtering the device introduces.

2.2.1 CAD Technique

All the housings were designed to be manufactured using a fused deposition modelling (FDM) 3D printer. This type of 3D printer uses rolls of plastic filament, melted through a nozzle and deposits the plastic layer by layer to build up the component. The main advantage of 3D printing is that it is relatively quick to produce working prototype housings with very little post processing. Designing for FDM printing does have some limitations. For example, steep angles cannot be printed without support material. This is wasteful, time consuming, and requires more clean up, post print. In all the design concepts below, care was taken so that they could be easily printed. If the concept ever needed to be mass produced, they would need to be edited for appropriate manufacturing techniques.

2.2.2 Design 1: “The Trumpet”

The first concept was arguably the most basic. The main part of this design was to keep the microphone and speaker as far apart as possible to maximize the possible gain of the device before a constructive feedback loop occurred. The housing was rounded to be comfortable to hold and wide enough to allow space for batteries and electronics. For the purposes in the first prototype, the housing was designed in two halves to be snap fitted together with pegs. Future prototypes would have needed a more elegant way to change the batteries.

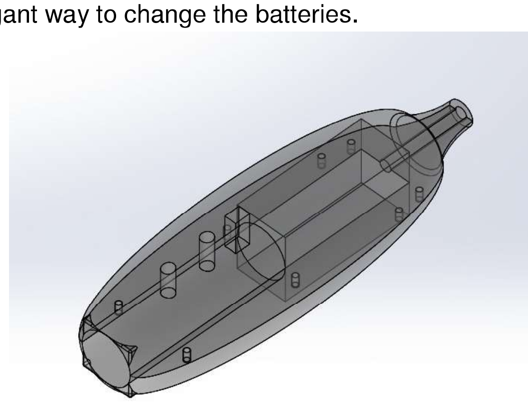

Figure 2.2.2.1: *The housing shape, transparent view of the trumpet style design. The speaker mounts in the front opening, the microphone at the back. The holes on top are spaces for a power switch, volume control and power indicator. The two halves of the housing are snap fitted together with pegs.*

This design was modified to have a moveable mount for the microphone. This would allow the user to position the microphone towards them whilst holding the HHD comfortably. This design therefore allows for the maximum response from the cardioid microphone by positioning further increasing the maximum usable gain before feedback. The speaker was also changed to be at 45 degrees to the microphone for a more comfortable user experience.

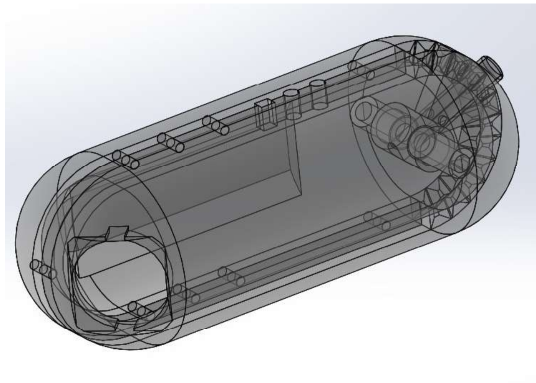

Figure 2.2.2.2: *Transparent CAD view. Like the trumpet design the two main halves of the housing are snap fit together with pegs. The moving microphone part is held by pegs and can move vertically by the spacing of the teeth. On the left image, the flat inside of the housing can be seen. This space is for mounting the electronics.*

2.2.3 Design 3: “The Tube”

This design further increases the perceived distance between the microphone and speaker without increasing the overall length of the device. This is done by first isolating the speaker from the microphone. In this design that is achieved by containing the speaker in a part separate to the main body of the housing. The microphone is placed at the end of a long tube. Assuming the seal between the speaker part and main housing is perfect, for the sound from the speaker to the microphone needs to travel around the outside of the case, then back down the tube, almost double the distance of the first two designs. The sound travelling down the tube will be more planar in nature so will not attenuate as much with distance as an open microphone but less of the sound from the speaker will be directed into the tube. Therefore, a greater level of gain can be achieved before feedback, assuming there is no interference from structure-borne vibrations. The shorter physical distance between the microphone and speaker should also reduce the effect of comb filtering.

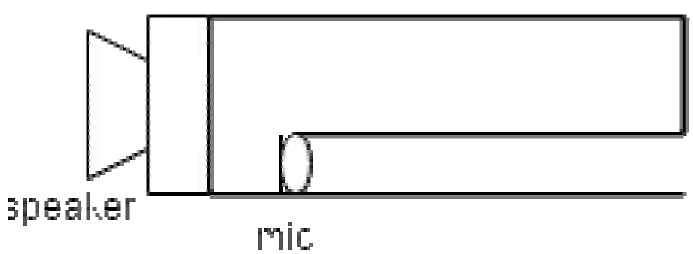

Figure 2.2.4.1: *A diagram showing how the isolation of the speaker from the microphone at the end of a tube increases the apparent distance between the two.*

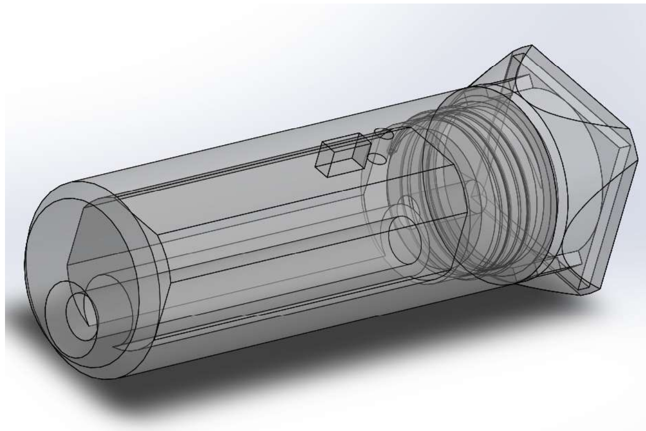

Figure 2.5.4.2: *A transparent view of the tube style housing. The inner walls of the casing are kept flat for easier mounting of the electronics within.*

However, from a practical point of view this design has some draw backs. The tube for the microphone takes up a lot of internal space that previous designs had for electronics. To accommodate this, the microphone was moved to the side of the device and the diameter of the device was slightly increased from other designs.

This design was modified to include a mechanical technique to further reduce comb filtering. Assuming the sound waves traveling down the tube are planar, curving the tube increases the path length of the sound. The tube length can be made to be equal to the length of the entire device. This means that speech will reach both the microphone and the speaker at the same time. Therefore, the amplified sound of the speaker will be in phase with the speech so no comb filtering should occur.

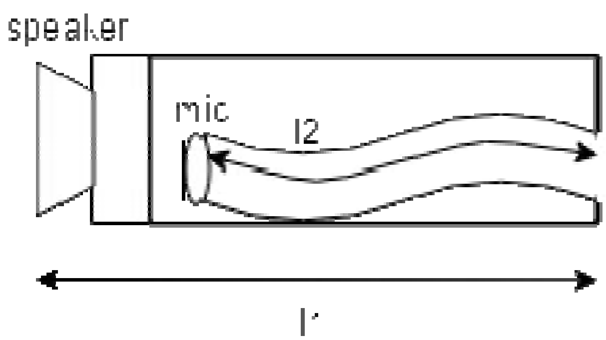

Figure 2.5.5.1: *A diagram showing how the curved microphone tube should work. If l2 is equal to l1 then the sound from speech will arrive at the microphone and the front of the speaker at the same time so the amplified speech will be in phase with the initial speech so no comb-filtering will occur. Again, the inclusion of the microphone tube, internal space is reduced, but more so. It also means the internal cavity is oddly shaped.*

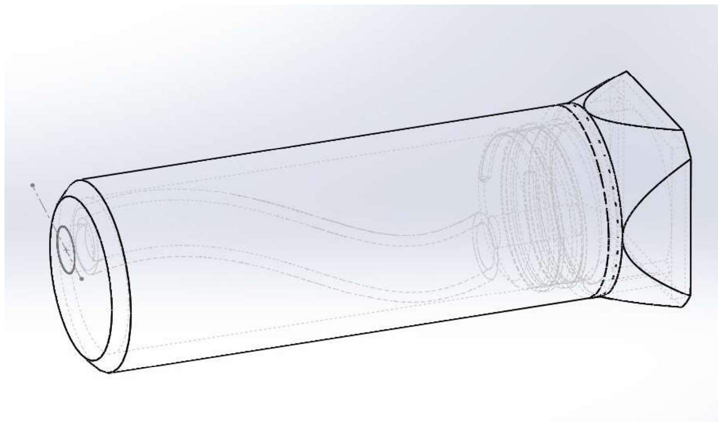

Figure 2.5.5.2: *The adapted "Tube" CAD model to include a spiral tube instead of a straight microphone tube. As mentioned, the spiral takes up much of the internal space for batteries and electronics. In future iterations, an extra threaded part should be made at the bottom end of the HHD as well as where the speaker is mounted for ease of electronics placement.*

2.3 Summary

The housing iterations were designed for their mechanical signal processing and for user comfort. The trumpet design is simple but has comb filtering issues and is potentially uncomfortable to use. The adjustable design is more comfortable but still has comb filtering issues. The third and fourth designs compromise by potentially cutting down the comb-filtering issues but with more awkward component and battery placement.

3 EXPERIMENTAL METHODOLOGY

Prototypes of the designs were build and tested objectively for their maximum gains before feedback. The set-up of the testing apparatus is shown below:

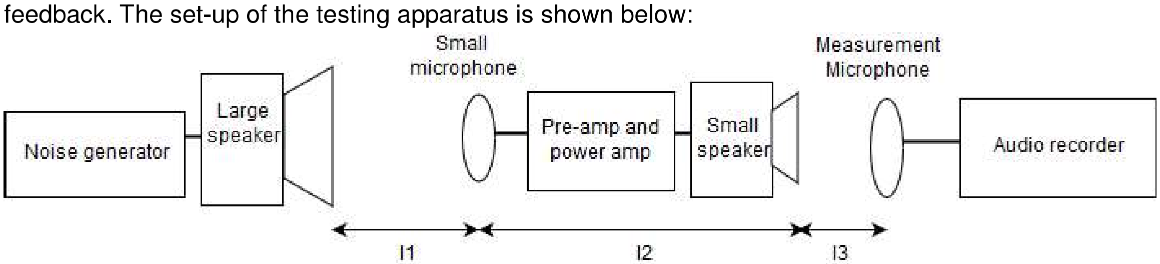

Figure 3.1: *A block diagram showing the layout of apparatus for testing the HHD. l1 represents the distance from the talker to the device, l2 the length of the device, and l3 the distance between the device and the listener. These lengths were changed to simulate different device sizes and distances between user and listener. The sum of all the lengths was always equal to 0.3 m.*

Firstly, a baseline was measured by removing the HHD, and recording 30 seconds of white noise. The HHD was re-inserted, its gain was adjusted to just below feedback, and the white noise was recorded again. Using software, the amplitudes from 80 Hz to 22 kHz was recorded. The baseline was then subtracted from the HHD recordings to give the gains over all the frequencies.
The practical method was compared to a Python model simulating the distances between speaker and microphone without the plastic housings present.

4 RESULTS

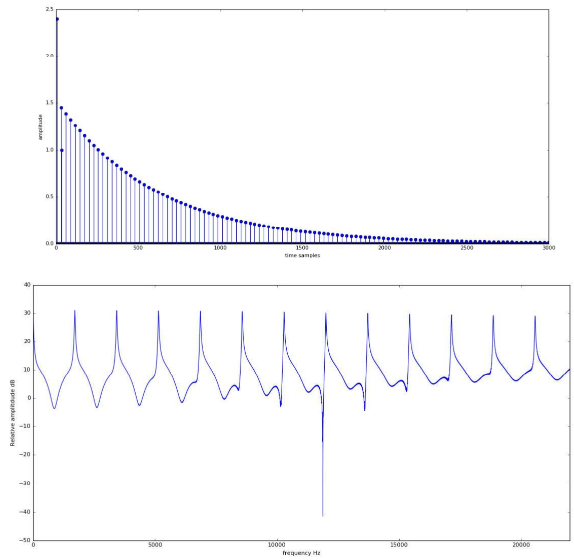

Figure 4.1: *The predicted time and frequency response of a HHD of length 20cm. A sample rate of 48000 Hz was used. Clear comb filtering is shown of period 1.6 kHz.*

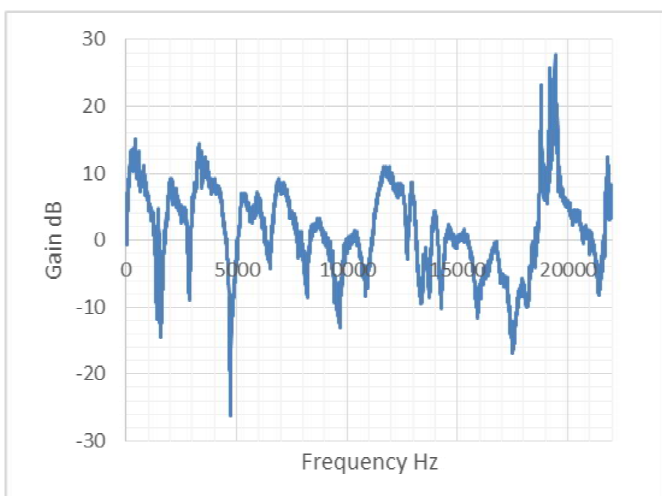

Figure 4.2: *HHD of length 20cm with no housing. A similar periodicity of comb filtering is shown but the maximum gain is lower than predicted.*

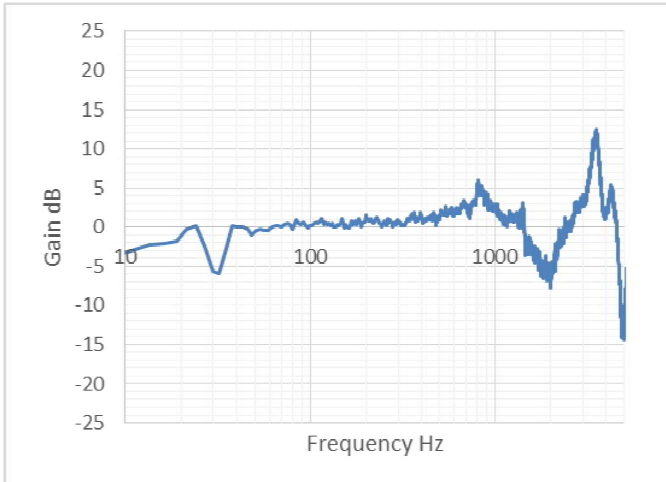

Figure 4.3: *The maximum gain before feedback of "The Trumpet style" housing of length 20 cm at human speech frequencies. The average gain for the telecommunication frequencies was 0.66 dB and for the wider speech bandwidth as 1.58 dB. It is shown that the amplitude of much of the low frequencies are reduced in amplitude otherwise the system becomes unstable. This is most likely due to structure-borne vibrations of the housing.*

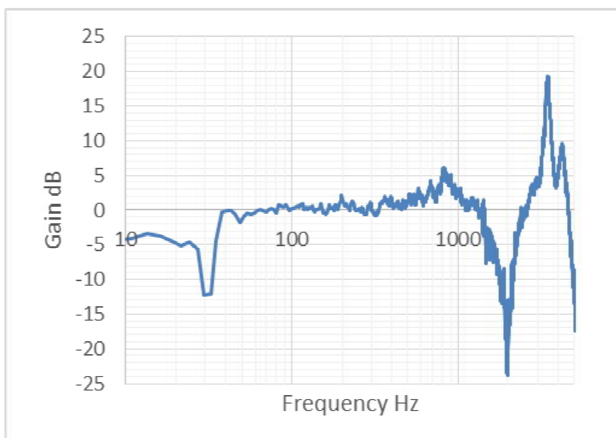

Figure 4.3: *A graph showing the maximum gain before positive feedback when the microphone and speaker are mounted in the adjustable microphone housing design, length 15 cm. Much of the frequencies of speech have very little gain or negative gain due to structure-borne vibrations causing the microphone to vibrate. The average gain for the telecommunications bandwidth was -0.54 dB and the wider bandwidth gain was 1.05 dB*

5 CONCLUSIONS

This project aimed to design and manufacture a handheld hearing device and assess its feasibility. The current iteration of the device is not a functional prototype, but that is not to say that it is feasible as a final product. With a few minor changes to the housing designs, the maximum gains of the device could be much closer to the gains seen without a housing. With the inclusion of analogue or digital filters, these gains may even be exceeded.

It has been found that low-cost miniature electronic components are affordable and effective. These components can fit comfortably into a housing small enough to be kept in a pocket or worn on an item of clothing. However, the current housing design is prone to structure borne vibrations so extra isolation needs to be included to the microphone mount of future iterations.

Subjective tests need to be performed on future iterations to assess the qualitive effectiveness, robustness, and usability in a real-world situation. In the future, the handheld hearing device may be a convenient and comfortable way for users to communicate with patients with hearing loss that are not wearing, or don't have access to, their own personal hearing aids.

6 REFERENCES

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