A study on the Dik Universe

Development of a bigger dik

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Bachelor Project

STUDENT REPORT
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Here you can write something about which tools and software you have used for typesetting the document, running simulations and creating figures. If you do not know what to write, either leave this page blank or have a look at the colophon in some of your books.

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STUDENT REPORT

Electronics and IT
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Title:
Project Title

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Digital Filtering

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Project Group:
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Abstract:

Here is the abstract

Contents

Todo list

Preface

Here is the preface. You should put your signatures at the end of the preface.

Aalborg University, November 21, 2018

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Chapter 1

Introduction

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1.1 Examples

You can also have examples in your document such as in example 1.1.

Example 1.1 (An Example of an Example)
Here is an example with some math

$$0 = \exp(i\pi) + 1 \ . \tag{1.1}$$

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1.2 How Does Sections, Subsections, and Subsections Look?

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1.2.1 This is a Subsection

and this

This is a Subsubsection

and this.

A Paragraph You can also use paragraph titles which look like this.

A Subparagraph Moreover, you can also use subparagraph titles which look like this. They have a small indentation as opposed to the paragraph titles.

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Chapter 2

Problem Analysis

2.1 Problem Description

2.1.1 State of hearing aid

Hearing aid technologies have never been improving as fast as they do now. Ever since major smartphone manufacturer companies started investing into wireless wearable technology research, price of PSAP (personal sound amplification products) has decreased while the amount of features has increased. Throughout recent years wireless earphones became predominant in the global market due to new generation of Bluetooth technology. This improvement even further reduces power consumption of wireless technology. The only three differences between hearing aids and PSAPs were:

- The battery life. Due to a far higher number of features and consistent audio stream PSAPs consume a much higher amount of power.

- Hardware design differences. PSAPs are not oriented around sound localization to inform the user about where the sound is coming from regarding natural sources. PSAPs are often not oriented to be invisible to others, they are more often purposely made to stand out and be recognized among its competitors. Fit customization is also often minimal on PSAPs.

- Regulation requirements to produce the hearing aid and license requirements to sell it.

Other differences lay in software and could be eliminated through a software update.

It is believed that legislative issues could be solved if manufacturers would put effort to reach for an agreement with legislators although it would require a lot of changes since current hearing aid selling process consists of far more than just taking the product off the shelf and swiping it through the register - it is normally performed at hearing clinics, hearing aid is thoroughly adjusted to fit the consumer's ear for long periods of time, warranty for these devices also is taken in a far more serious manner: it comes with included follow-up office visits, checks and cleaning procedures to maintain the highest level of performance. Some companies do express interest to merge the two markets. According to "The State of Hearing Healthcare 2017" by Lindsey Banks, "If Apple Air Pods or Samsung Gear IconX could add in hearing aid functions, that's instant access to over half of the U.S. over night."

`https://www.everydayhearing.com/hearing-loss/articles/state-of-hearing-healthcare-2017/ ((puthashhere))tech`

2.2 Problem Delimitation

2.3 Initiating Problem

Chapter 3

Mathematical Analysis

3.1 Geometric Structure

Chapter 4

Development

4.1 Component List

4.1.1 Microphones characteristics

4.1.2 Measurements scenarios

4.1.3 Setup

4.2 Analog to digital conversion

Chapter 5

Noise Filtering

5.1 Research

5.1.1 Voice frequency analysis

Voice frequency ranges vary heavily depending on whether it sources from a male of a female. Fundamental voice frequency varies from 100Hz to 900 Hz for men and 350Hz to 3KHz for women. Including peaks to conserve natural sounding voice, a wider frequency range has to be considered. It rises to 8 KHz for males and 17KHz for females [**Seaindia**]. Yet different researches often come up with different results. For example, in phone communications it is accepted to transmit frequency range between 400Hz and 3400Hz. This is the reason some peoples' voices transit poorly over the phone yet for most cases it work fine. This example allows to conclude that smaller frequency ranges could be acceptable. To conserve all of the properties of the human voice, filter boundaries should be around 100Hz to 17KHz but this range would filter out any noise as it takes up almost an entire frequency range of human hearing (approximately 20Hz to 20KHz).

5.1.2 Filter characteristics

Firstly it was defined that the input response of the signal is infinite and therefore an IIR filter has to be applied. Considering that computing power limitation is out of scope for this project, it was decided to use Butterworth filter as around higher orders of the filter approaches rectangular form.

5.2 Results

Making a field research to find the frequency range that would fit the needs of this project was out scope, therefore to test the filters it was decided to take trial-error approach. A few samples were made outside during a windy day. This was considered a good idea as it has recreated one of the most common conversation scenarios.

At first it was attempted to conserve the entire frequency range that humans can produce. This has resulted in a filter that seemed to filter out a big part of the noise but if it was listened to, all of the previously recorded noise was still there.It was hard to tell the difference between filtered and not filtered sound samples.

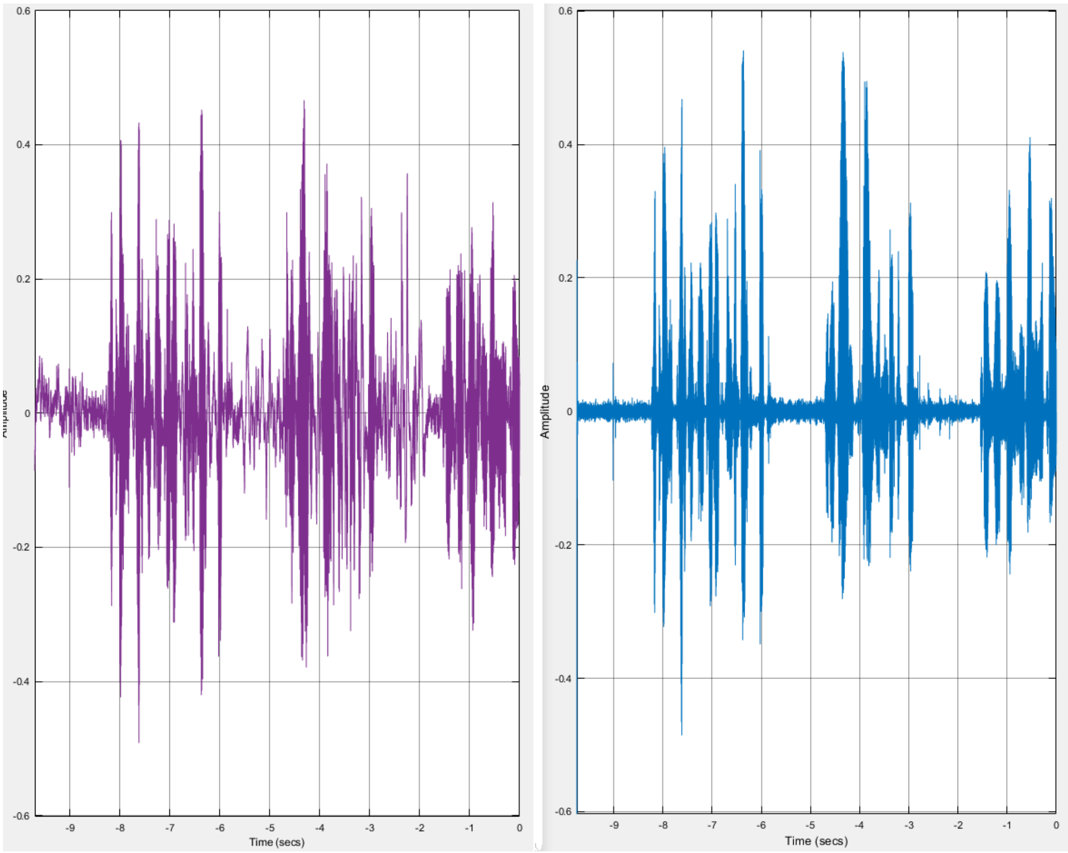

Figure 5.1: Recorded sample on the right and filtered result on the left

As the first approach did not seem to output a satisfying result, it was attempted to the signals by leaving only fundamental frequency radius (100Hz to 3KHz. The result seemed to be far more satisfying as most of the noise was eliminated. However, output clarity was poorer by far in comparison to original. Result of this test could be considered a solution for noise canceling although it would be a very poor one.

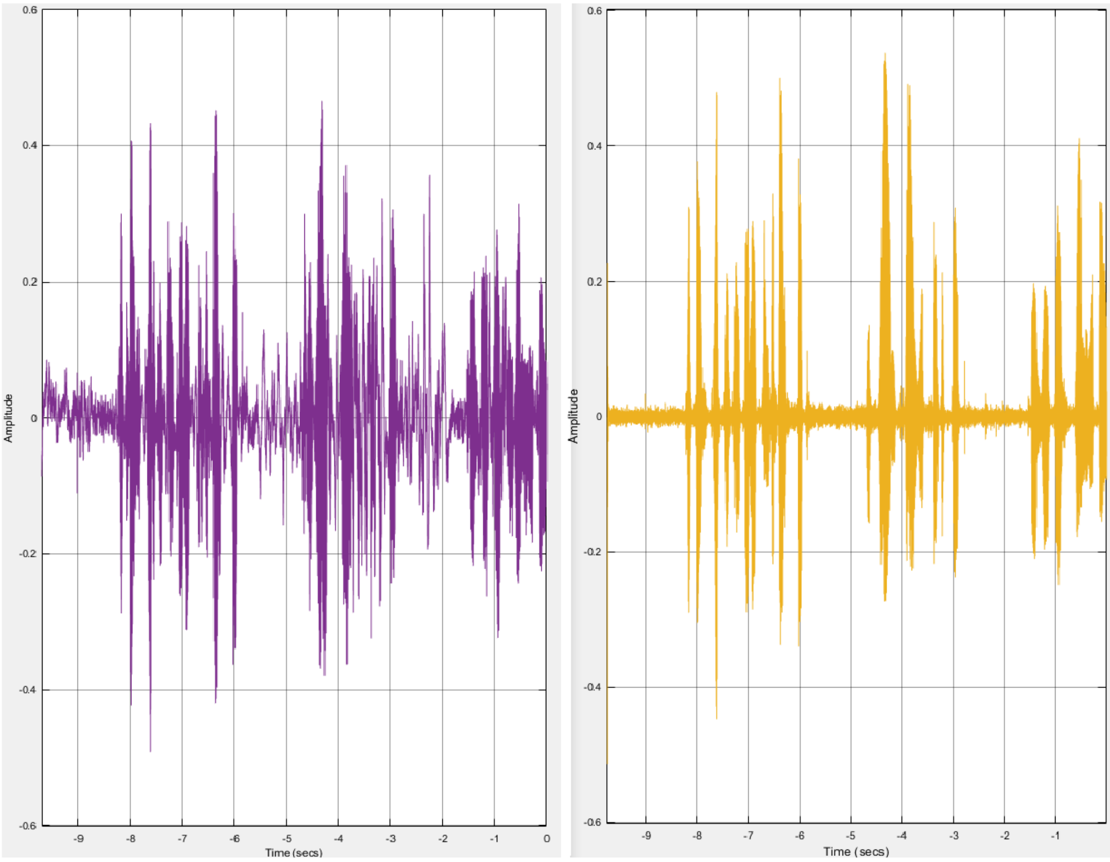

Figure 5.2: Recorded sample on the right and filtered result on the left

Finally the frequency range picked for communication through phone (400Hz to 3400Hz) was taken to trial. Output of the filter was barely different when compared to one the previous one. It had filtered out most of the noise yet the sound clarity was still very poor.

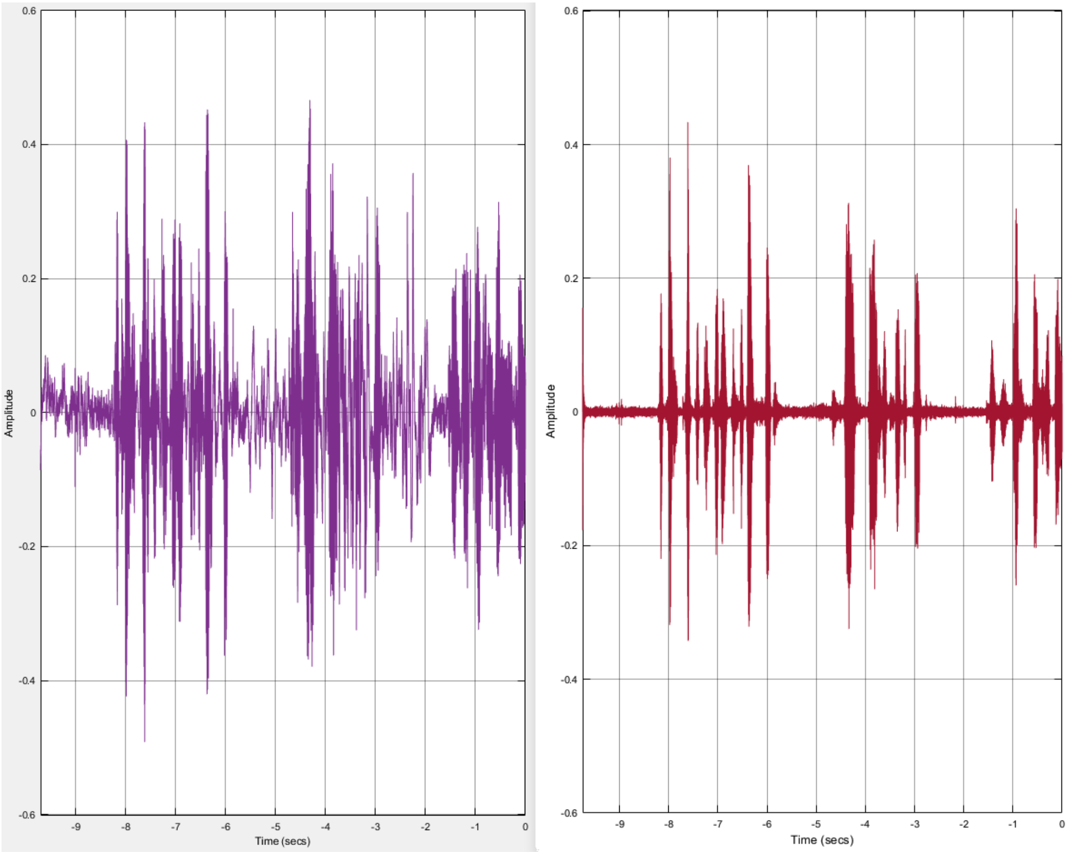

Figure 5.3: Recorded sample on the right and filtered result on the left

5.3 Conclusion

The approach that the group has taken seems to be correct - the filters worked as intended, however a question was raised whether this method is capable of producing a signal that would cancel out most of the sound without affecting sound quality. The filter could be more efficient if different frequency ranges were available to pick from to filter out noises that are common in different environments. This would require in-depth analysis of noises in these environments.

Chapter 6

Directional Noise Elimination

Chapter 7

Neural Network Speech Isolation

Chapter 8

Comparison

8.1 Synthetic Samples

8.2 Recorded Samples

8.3 Live events

8.4 Comparing all three methods and combining them

Chapter 9

Results

Chapter 10

Future Work

Chapter 11

Conclusion

In case you have questions, comments, suggestions or have found a bug, please do not hesitate to contact me. You can find my contact details below.

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Appendix A

Appendix A name

Here is the first appendix