

Development of a bigger dik

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Bachelor Project







Electronics and IT Aalborg University http://www.aau.dk

STUDENT REPORT

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Project Title

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Digital Filtering

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Abstract:

Here is the abstract

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Preface

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		Aalborg	University, N	November 20, 2	2018
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Introduction

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1.1 Examples

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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?

Well, like this

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.

Is it possible to add a subsubparagraph?

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

I think that a summary of this exciting chapter should be added.

Problem Analysis

- 2.1 Problem Description
- 2.2 Problem Delimitation
- 2.3 Initiating Problem

Mathematical Analysis

3.1 Geometric Structure

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

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 Filtering characteristics

5.1.3 Results

Making a field researh 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

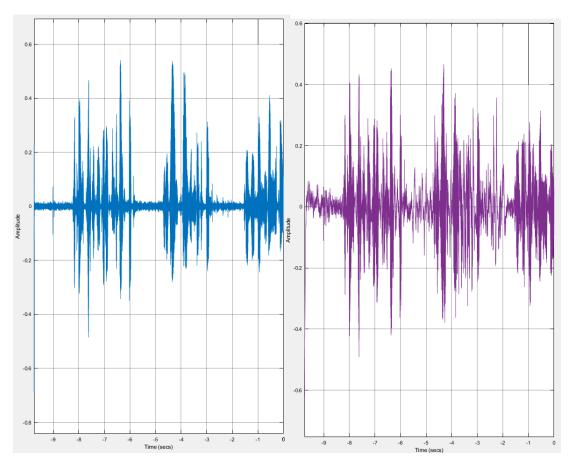


Figure 5.1: Filtered result on the left and recorded sample on the right

samples.

Directional Noise Elimination

Neural Network Speech Isolation

Comparison

- 8.1 Synthetic Samples
- 8.2 Recorded Samples
- 8.3 Live events
- 8.4 Comparing all three methods and combining them

Results

Future Work

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