
Speech Recognition

- With Information Retrieval & Command Prompt -

Project Report
ED5-1-E17

Aalborg University
Electronics and IT

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Electronics and IT

Aalborg University

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AALBORG UNIVERSITY

STUDENT REPORT

Title:

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Abstract:

This report goes into detail about the implementation of distributed speech recognition systems, information retrieval and machine learning. The main objective of the speech recognition system is to provide a handsfree human-computer interface experience, where the system is also able to handle noise.

The content of this report is freely available, but publication (with reference) may only be pursued due to agreement with the author.

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■ reference this part with <i>https : //www.allerin.com/blog/six – types – of – neural – networks</i>	8
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Preface

The project entitled Speech Recognition with Focus on information retrieval and command prompt was made by three students from the Electronics and Computer Engineering programme at Aalborg University Esbjerg, for the P5 project during the fifth semester.

Aalborg University, October 27, 2017

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Chapter 1

Introduction

Here is the introduction. The next chapter is chapter 1.
a new paragraph

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

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

You can adjust the colour and the line width in the `macros.tex` file [1].

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 [2].

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

Is it possible to add a subsubparagraph?

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

Chapter 2

Problem Description

2.1 Problem Description

List to write about here:

- Main Objective/Goal
- What will be required
- How are we going to solve the Objective?

2.2 Problem Delimitation

To ensure that the project meets the desired scope, specific success criteria are formed which will be evaluated during the final stages of the report. The following list of requirements are the desired success criteria, where the focus will be on reaching the given goals.

- Detect human speech
- Design a deep neural network
- Compare different regularization techniques (drop out, batch normalization...)
- Design a filter
- Create a network infrastructure for Client/Server Model (Distribution)
- Implement a speech recognition system with information retrieval

Brief summary on how are we going to obtain all of these bullet points.

Chapter 3

Speech Processing

Speech is an effortless and highly efficient form of communication. Continuous speech recognition software is readily available from local computing stores and comes complete with microphone, allowing the user to dictate text directly into a document. The current technology is not perfect but is getting better [3, p. 396].

Words are carried as sound waves, which are analogue signals. In order for the speech to be processed it needs to be run through a signal processor that selects the frequencies and amplitude of the signal. Further on, the signal is mapped to individual sound units called phones. These phones need to be identified and grouped in such a way to ensure that each word has a different phonetic structure, otherwise words would be impossible to distinguish from one another.

Phones can have different sounds depending on context. For example, the phone *th* in the word *three* has a different to *th* in *then*. To overcome this different variations of the same phones, it is better to abstract the phones into a generalized grouping called a **phoneme**. Phonemes are written inside forward slashes (*/th/*) and they will have a specific pronunciation for each sound, depending on the context. These phonemes are used as a transitional layer when trying to convert speech to text and vice versa when speech needs to be synthesised.

Table 3.1: Phones

[ay]	iris
[b]	bin
[er]	bird
[l]	lip
[p]	pin
[th]	thin

3.1 Signal processing

Sound waves that carry human speech are variations in air pressure. The key components of a sound wave are its **amplitude**, which measure the intensity of the sound and the **frequency**, which describes the rate at which the amplitude varies over time. When speaking in a microphone, the change in air pressure causes the diaphragm to oscillate. The size of the oscillations are directly proportional to the amplitude of the signal, while the rate at which the diaphragm oscillates gives represents the the rate at which the air pressure changes. At specific time intervals the signal can be sampled and the data can be used in a wide array of digital signal processing tools. For example, the digital signal can be plotted as an x-y plot, where the y-axis defines the amplitude and the x-axis shows the passage of time. The frequency can be easily be determined from the plot as we find the number of cycles that the signal does per second.

Although a visual difference can be seen when plotting vowels and consonants, a visual inspection of the waveform will not reveal any critical information, making it impossible to see all the phonemes.

Fourier transform

To analyse a discrete signal we use the Fourier transform. This allows us to convert the signal in the frequency domain and deconstruct it in a series of simple sine waves. These sine waves have specific amplitudes and frequencies and while plotting the FFT of the signal, the dominant waves can be identified by the intensity of the amplitude.

Chapter 4

Machine Learning for Speech

4.1 TensorFlow

TensorFlow provides multiple Application Programming Interfaces (APIs) for machine learning. The lowest level API "TensorFlow Core" provides complete programming control [4]. For those who require fine levels of control over their models, TensorFlow Core is a well-suited tool for the job. There are higher level APIs that are built on top of TensorFlow Core. These higher level APIs are typically easier to learn and use than Tensorflow Core [4]. In addition, the higher level APIs such as "tf.estimator" helps with managing data sets, estimators, training and inferences (testing your trained network), as well as, making repetitive tasks easier and more consistent [4].

The subsection below will start with TensorFlow Core. In order to gain an understanding of the basic principles that TensorFlow has to offer, a model shall be made. In the subsection after that, the same model will be implemented with tf.estimator. Knowing TensorFlow Core principles will give us a great mental model of how things are working internally when we use the more compact higher level API.

4.1.1 Tensors

The central unit of data in TensorFlow is the tensor. A tensor consists of a set of primitive values shaped into an array of any number of dimensions. A tensor's rank is its number of dimensions. Here are some examples of tensors:

```
3 # a rank 0 tensor; this is a scalar with shape []  
[1., 2., 3.] # a rank 1 tensor; this is a vector with shape [3]  
[[1., 2., 3.], [4., 5., 6.]] # a rank 2 tensor; a matrix with shape [2, 3]  
[[[1., 2., 3.]], [[7., 8., 9.]]] # a rank 3 tensor with shape [2, 1, 3]
```

4.1.2 TensorFlow Core

4.1.3 tf.estimator

4.2 Deep neural network for speech

4.2.1 Deep Feed Forward (DFF)

The simplest of all neural networks, the feedforward neural network, moves information in one direction only. Data moves from the input nodes to the output nodes, passing through hidden nodes (if any). The feedforward neural network has no cycles or loops in its network.

reference this
part with [https :
//www.aller.in.com/blog/six-
types - of - neural -
networks](https://www.aller.in.com/blog/six-types-of-neural-networks)

4.2.2 Recurrent Neural Network (RNN)

The recurrent neural network, unlike the feedforward neural network, is a neural network that allows for a bi-directional flow of data. The network between the connected units forms a directed cycle. Such a network allows for dynamic temporal behavior to be exhibited. The recurrent neural network is capable of using its internal memory to process arbitrary sequence of inputs. This neural network is a popular choice for tasks such as handwriting and speech recognition.

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above

Long/Short Term Memory (LSTM)

Chapter 5

Distributed speech recognizer

5.1 The DSR system

5.2 Acoustic models

5.3 Language model

Chapter 6

Knowledge-based IR system

6.1 The document retrieval system

6.2 The question answering system

Chapter 7

Discussion

Chapter 8

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