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## ZADÁNÍ BAKALÁŘSKÉ PRÁCE

**Název:** Rozpoznávání souvislé řeči s využitím neuronových sítí  
**Student:** Adam Zvada  
**Vedoucí:** Ing. Miroslav Skrbek, Ph.D.  
**Studijní program:** Informatika  
**Studijní obor:** Teoretická informatika  
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### Pokyny pro vypracování

Provedte rešerši metod pro rozpoznávání souvislé řeči s využitím neuronových sítí. Uvažujte rekurentní neuronové sítě a zvažte také možnost použití neuronových turingových strojů. Na základě rešerše a po dohodě s vedoucím práce vyberte vhodné řešení pro robota NAO. Maximálně využijte existujících knihoven s implementacemi potřebných metod. Navržené řešení otestujte na reálných datech. Rozsah práce upřesněte po dohodě s vedoucím práce.

### Seznam odborné literatury

Dodá vedoucí práce.

doc. Ing. Jan Janoušek, Ph.D.  
vedoucí katedry

doc. RNDr. Ing. Marcel Jiřina, Ph.D.  
děkan

V Praze dne 13. února 2018



CZECH TECHNICAL UNIVERSITY IN PRAGUE  
FACULTY OF INFORMATION TECHNOLOGY  
DEPARTMENT OF THEORETICAL INFORMATICS



Bachelor's thesis

# Continuous Speech Recognition by Neural Networks

*Adam Zvada*

Supervisor: Ing. Miroslav Skrbek Ph.D

May 9, 2018



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# Acknowledgements

THANKS



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## Declaration

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## Abstrakt

V několika větách shrňte obsah a přínos této práce v českém jazyce.

**Klíčová slova** Replace with comma-separated list of keywords in Czech.

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## Abstract

In my bachelor thesis Summarize the contents and contribution of your work in a few sentences in English language.

**Keywords** Replace with comma-separated list of keywords in English.



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# Introduction

What is speech recognizer? History... Why to build? Why it helps? How i will do it.





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# Neural Network

Neural networks have a remarkable ability to derive meaning from complicated data. They can be used to extract patterns and detect trends that are too complex to be noticed by either humans or other computer techniques[3]. Even though they have been around since the 1950s, it is only in the last decade when they started to outperform robust system or even humans in specify tasks. However, they require a huge amount of training examples and computational power to be trained for preforming a reasonable prediction. Fortunately, GPUs has seen enormous increase in performance<sup>1</sup> and 90% of the data in the world today has been created in the last two years alone, at 2.5 quintillion bytes of data a day[2]. That's why ANN is big topic in Computer Science and in the technology industry and it currently provides the best solutions to many problems such as speech recognition, image recognition, and natural language processing.

## 1.1 Inspiration in Nature

Artificial neural network (ANN) is heavily inspired by the way how biological neural networks process information in the human brain. Even though our brain is extremely complex and still not fully understand, we just need to know how information is being transferred. The basic building block is nerve cell called *neuron*. It receives, processes, and transmits information through electrical and chemical signals. It's estimated that an average human has 86 billion neurons[1].

As shown on Figure 1.1, *dendrites* are extensions of a nerve cell that propagate the electrochemical stimulation received from other neurons to the cell body. You may think of them as inputs to neuron, whereas neuron's output is called *axon*, a long nerve fiber that conducts electrical impulses away from

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<sup>1</sup>GPUs are explicitly designed to handle multiple matrix calculations at the same time. Evaluation and training of artifical neural networks are mostly matrix operations.

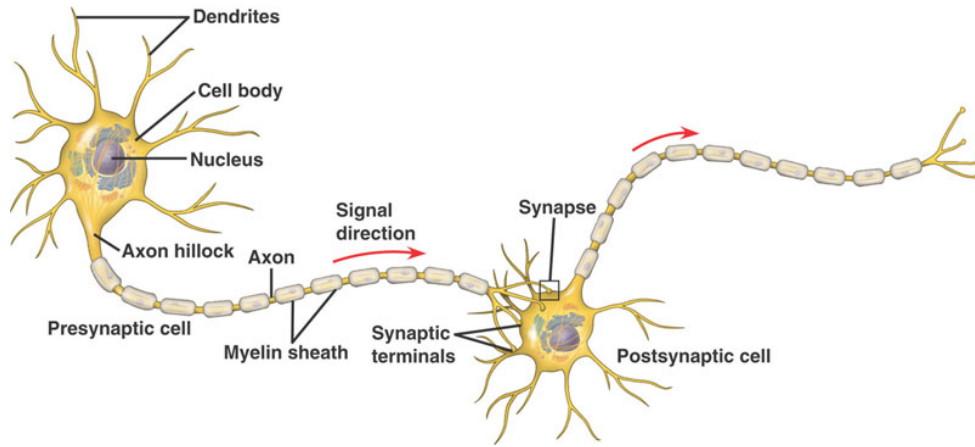


Figure 1.1: Illustration of nerve cell and communication flow

the cell body. The end of axon is branched to many axon terminals which can be again connected to other dendrites. The connection is managed by *synapses* that can permit the passing of electrical signal to cell body. Once the cell reaches a certain threshold, an action potential will fire, sending the electrical signal down the axon to other connected neurons.

### 1.2 Artificial Neuron

Artificial neuron is a generic computational unit, basic building block for artificial neural network (ANN). It's simplified version of the biological counterpart and we are able to map parts of biological neuron with the artificial one. It takes  $n$  inputs represented as a vector  $x \in \mathbb{R}^n$  which correspond to dendrites. Generally artificial neuron produces single output  $y \in \mathbb{R}$  as biological neuron where we call it axon. Each neuron's input  $i = 1, 2, \dots, n$  has assigned weight (synapse)  $w_1, w_2 \dots w_n$ , they refer to the connection strength between neurons. Weights and same as for synapse are the backbone of learning because in training phases, they keep changing to produce wanted output. Inside the artificial neuron, input vector with their weights are combined and run through an activation function producing some output  $y$ . This process is illustrated in Figure 1.2.

### 1.3 Perceptron

Perceptron is the simplest ANN with just one neuron and since we covered the basic intuition about artificial neuron we may proceed further and take a look at how output is actually calculated. The equation for a perceptron can

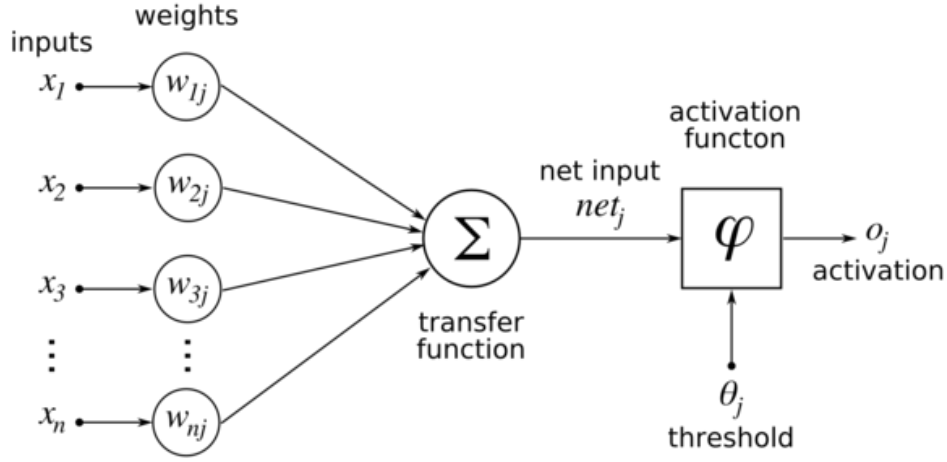


Figure 1.2: Illustration of nerve cell and communication flow

be written as

$$y = f\left(\sum_{i=1}^N w_i \cdot x_i + b\right) \quad (1.1)$$

where

- $x$  - input vector
- $y$  - predicted output
- $f$  - activation function
- $w$  - weights
- $b$  - bias

Perceptron is a basically linear classifier, therefore the data has to be linearly separable otherwise we would not be able to make the correct prediction. Problems such as speech recognition are not definitely linearly separable, however we can solve non-linear decisions for example by introducing another layer of neurons, thus creating *Multilayered Perceptron*.

### 1.3.1 Activation Functions

We have stated that biological neuron fires electrical signal to other connected neurons whenever it reaches a certain threshold of incoming electrical impulses. Activation function is based on that concept and inside an artificial neuron it is used for calculating output signal via equation 1.1. It introduces non-linear properties to our ANN and without an activation function would

be just a regular linear regression model. Nowadays many different activation function are being used and their performance varies from model to model.

List of some activation function:

- *Sigmoid*

$$\sigma(x) = \frac{1}{1 + e^{-x}}$$

- *Hyperbolic Tangent*

$$\tanh(x) = \frac{(e^x - e^{-x})}{(e^x + e^{-x})}$$

- *ReLU*

$$f(x) = \begin{cases} 0 & \text{for } x < 0 \\ x & \text{for } x \geq 0 \end{cases}$$

- *Softmax*

$$f_i(\vec{x}) = \frac{e^{x_i}}{\sum_{j=1}^J e^{x_j}}, \quad i = 1, 2 \dots J$$

where  $i$  is number of output

\* TODO: Activation Functions features... differentiable and non-linear\*

### 1.3.2 Bias

We can think of bias as a value stored inside neuron and being used to calculate it's output. The bias value allows the activation function to be shifted to the left or right, to better fit the data.

## 1.4 Topology of Artificial Neuron Network

Basic ANN as feedforward model is a directed graph with nodes as neurons and edges with weights representing connection to other neurons. ANN can be divided to three important layers as shown in Figure 1.3. Yellow nodes is an input layer which takes input data, dimension of input vector has to correspond to number of input nodes. Hidden layer as the green nodes is most important to ANN and that is where the training and evaluation happens. Number of hidden layers and neurons needs to be in a good ratio between its size and its effectiveness. Output layer produces output vector as the prediction for given input.

## 1.5 Network Evaluation

Evaluation of ANN is done layer by layer.

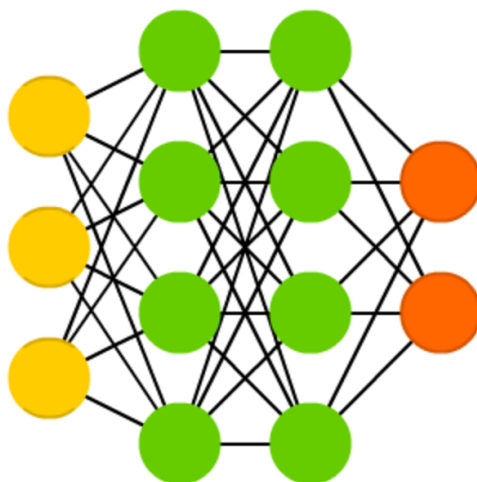


Figure 1.3: Basic topology of fully connected artificial neuron network with input vector of size 3, output vector of size 2 and two hidden layers.

## 1.6 Training

The greatest trait of ANN is ability to learn from given data and then make the best approximate prediction. The aim of the learning process is to find the most optimal values for network's weights and biases while minimizing error on predicated values. For ANN to learn we have to introduce training data consisted of input vector which will be feeded to the network and desired output value (label) for calculating our loss. This approach is called supervised learning<sup>2</sup>.

### 1.6.1 Loss Function

Loss function compares the prediction from ANN with the desired output and returns the error of the prediction. During a training ANN, the goal is to minimize given loss function. The most common and most intuitive loss function is Mean squared Error (MSE),

$$\text{MSE}(y, \hat{y}) = \frac{1}{n} \sum_{i=1}^n (y_i - \hat{y}_i)^2.$$

### 1.6.2 Backpropagation

Backpropagation algorithm is responsible for the ability to learn from given training data. It is an iterative algorithm which for each training data from

<sup>2</sup>ANN can be also trained using unsupervised learning.

given training dataset backpropagates the error and adjust the weights and biases accordingly to get desired output.

### 1.6.2.1 Optimization

Backpropagation requires optimizer to minimize the error on the training data. We will describe backpropagation with using *gradient descent* as the most common optimization algorithm.

Weights and biases are updated using formula,

$$W_{jk}^l := W_{jk}^l - \alpha \frac{\partial E}{\partial W_{jk}^l}$$

$$b_j^l := b_j^l - \alpha \frac{\partial E}{\partial b_j^l}$$

where  $W_{jk}^l$  is weight with connection between unit  $j$  in layer  $l$  and unit  $i$  in layer  $l + 1$ ,  $b_j^l$  is bias associated with unit  $i$  in layer  $l + 1$ ,  $\alpha$  is a learning rate [4], and  $\frac{\partial E}{\partial W_{jk}^l}$  or  $\frac{\partial E}{\partial b_j^l}$  can be interpreted as minimizing loss function with respect to given weight and bias respectively.

By applying a chain rule twice on the partial derivative of the loss function with respect to a weight, we get

$$\frac{\partial E}{\partial W_{jk}^l} = \frac{\partial E}{\partial a_j^l} \frac{\partial a_j^l}{\partial z_j^l} \frac{\partial z_j^l}{\partial W_{jk}^l}$$

where  $z_j^l$  is a sum of weighted inputs to unit  $j$  in layer  $l$

$$z_j^l = b_j^l + \sum_{k=1}^K w_{jk}^l a_k^{l-1}$$

and  $a_j^l$  is an output of node  $j$  in layer  $l$

$$a_j^l = f(z_j^l).$$

Let's calculate the last two products of equation \*Reference\*:

$$\frac{\partial a_j^l}{\partial z_j^l} = f'(z_j^l)$$

$$\frac{\partial z_j^l}{\partial W_{jk}^l} = \frac{\partial W_{jk}^l a_k^{l-1}}{\partial W_{jk}^l} = a_k^{l-1}$$

We introduce a new variable  $\delta_j^l$  which represents the error in unit  $j$  in layer  $l$  and helps us to better understand and calculate real interested value of  $\frac{\partial E}{\partial W_{jk}^l}$  and  $\frac{\partial E}{\partial b_j^l}$ .

$$\delta_j^l = \frac{\partial E}{\partial z_j^l}$$

We will simplify the error equation on neuron  $j$  in output layer  $L$  as

$$\delta_j^L = \frac{\partial E}{\partial z_j^L} = \frac{\partial E}{\partial a_j^L} \frac{\partial a_j^L}{\partial z_j^L} = \frac{\partial E}{\partial a_j^L} f'(z_j^L)$$

Now we have enough information to reformulate equation \*reference\* for output layer to

$$\frac{\partial E}{\partial W_{jk}^l} = \delta_k^L a_j^L.$$

However, to be able to update weights inside the hidden layers, we have to redefine the calculation of  $\delta_j^l$ . We know that the error produced by an output neuron is just influencing the output value but inside a hidden layer the produced error propagates to all following layers. Therefore we have calculate the  $\delta_j^l$  where layer  $l$  is inside a hidden layer and take into account all  $\delta^{l+1}$  from following layer  $l + 1$ .

$$\begin{aligned} \delta_j^l &= \frac{\partial E}{\partial z_j^l} = \sum_i \frac{\partial E}{\partial z_i^{l+1}} \frac{\partial z_i^{l+1}}{\partial z_j^l} = \sum_i \frac{\partial E}{\partial z_i^{l+1}} \frac{\partial z_i^{l+1}}{\partial a_j^l} \frac{\partial a_j^l}{\partial z_j^l} \\ &= \sum_i \delta_i^{l+1} W_{ij}^{l+1} f'(z_j^l) \end{aligned}$$

where the sum index  $i$  iterates over all neurons in layer  $l+1$  and Notice that we have substituted  $\frac{\partial E}{\partial z_i^{l+1}}$  with  $\delta_i^{l+1}$  which is calculated from previous iteration. Finally, we may calculate all weights adjustments through the whole network as

$$W_{jk}^l := W_{jk}^l - \alpha \delta_k^l a_j^l$$

where

$$\delta_k^l = \frac{\partial E}{\partial a_k^L} f'(z_k^L), \quad l = L$$

or

$$\delta_k^l = \sum_i \delta_i^{l+1} W_{ij}^{l+1} f'(z_j^l), \quad l = 2, \dots, L-1.$$

We won't be exemplifying the equation for biases adjustments because it follows a similar process shown above with just little changes, resulting to equation

$$b_j^l := b_j^l - \alpha \delta_j^l$$

### 1.6.2.2 Backpropagation Algorithm

---

**Algorithm 1** Backpropagation

---

- 1: Initialize network weights and biases
  - 2: **for each** training data from training dataset **do**
  - 3:     Forward pass and calculate network prediction for given training input
  - 4:     Calculate error  $\delta^L$  for output layer
  - 5:     Calculate errors  $\delta^l$  for hidden layers
  - 6:     Update weights and biases using precalculated  $\delta^l$
-



# Recurrent Neural Network

Neural networks are powerful learning models that achieve state-of-the-art results in a wide range of machine learning tasks. Nevertheless, they have limitations in the field of sequential data. Standard ANNs rely on the assumption of independence among the training examples but if data points are related in time or space then ANNs would not be the right model for the task\*Reference - arxiv\*.

Recurrent neural network (RNN) is type of neural network which is precisely designed to work with sequential data through time. The key difference is that RNN's neurons in hidden layer have a special edge (recurrent edge) to a next time step which can be interpreted as a loop. In RNN, the neuron's output is dependent on the previous computations which is sent through the recurrent edge. Basically, the recurrent edges or loops allow persistence of information from one time step to the next one as shown on \*Figure 3.1\*.

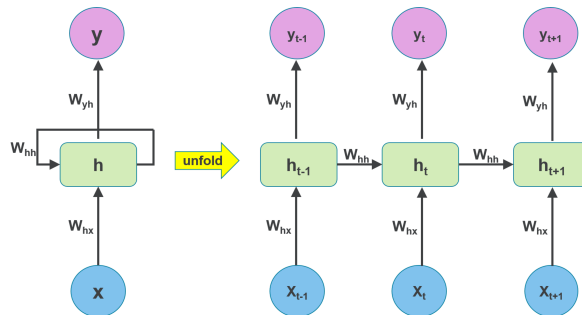


Figure 2.1: Simple RNN topology and illustration of unrolled RNN through time

## 2.1 Evaluation

In \*Figure 3.1.\* we may see simplification of evaluation process of RNN through the time steps. RNN's neuron cell in hidden layer takes two inputs,  $x_t$  and  $h_{t-1}$  which is value (hidden state) sent through the recurrent edge from previous time-step. The cell also produces two outputs,  $h_t$  as hidden state for upcoming time-setp

$$h_t = f(W_{hx}x_t + W_{hh}h_{t-1} + b_h)$$

where  $f$  is arbitrary non-linear activation function,  $W_{hx}$  is matrix of conventional weights,  $W_{hh}$  is the matrix of recurrent weights and  $b_h$  is a bias. The second output from cell is  $y_t$  which outputs the predication using precalculated hidden state  $h_t$ ,

$$y_t = W_{hy}h_t + b_y$$

where  $W_{hy}$  is matrix of output weights.

### 2.1.1 Softmax Fucntion

It is very common for RNN models to use *softmax* as activation function for output layer. Softmax function helps to get probability distribution of outputs so it's useful for finding most probable occurrence of output with respect to other outputs.

$$\text{softmax}(y)_j = \frac{e^{z_j}}{\sum_{k=1}^K e^{z_k}}, \quad \text{for } j = 1, \dots, K$$

Softmax is being used for calculating output value of  $y_t$  resulting to formula

$$y_t = \text{softmax}(W_{hy}h_t + b_y).$$

## 2.2 Training

Training a RNN is similar to training a traditional ANN. We also use the backpropagation algorithm, but since the parameters are shared by all time-steps in the network, the gradient at each output depends not only on the calculations of the current time-step, but also the previous time-steps.

### 2.2.1 Backpropagation Through Time

The most used algortihm to train RNN is *backpropagation through time* (BPTT), introduced by Werbos in 1990. BPTT is basically an extended version of back-propagation algortihm where we not only propagate the error to all following layers but also through the hidden states. We may think of it as unrolling the RNN to sequence of identical ANNs where the recurrent edge connects

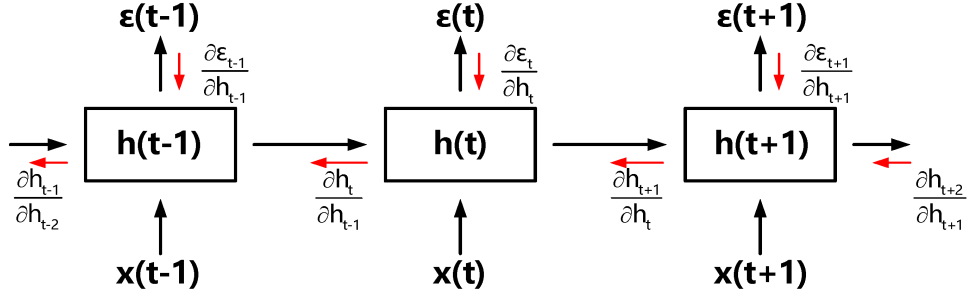


Figure 2.2: Deriving the gradients according to the back-propagation through time (BPTT) method. Notation for output value  $\epsilon(t)$  corresponds to our  $y_t$ .

the sequences of neurons in hidden layer together as shown on \*Figure 3.1 and 3.2\*. On \*Figure 3.2\* is also indicated how the errors are propagated. The propagation of errors through hidden states allows the RNN to learn long term time dependencies. The calculated gradients of the loss function for defined parameter ( $W, b$ ) through the sequence of unrolled RNN are then sum up, producing the final gradient for updating the weights or biases, \*Equation bottom\*.

$$\frac{\partial E}{\partial W_{ij}^l} = \sum_{t=1}^T \frac{\partial E_t}{\partial W_{ij}^l}$$

where  $E$  is predefined loss function,  $W_{jk}^l$  is weight with connection between unit  $j$  in layer  $l$  and unit  $i$  in layer  $l + 1$ ,  $T$  is number of input sequences and  $\frac{\partial E_t}{\partial W_{ij}^l}$  is calculated similarly as in backpropagation with just considering existence of recurrent edges

$$\frac{\partial E_t}{\partial W_{ij}^l} = \sum_{k=1}^t \frac{\partial E_t}{\partial y_t} \frac{\partial y_t}{\partial h_t} \frac{\partial h_t}{\partial h_k} \frac{\partial h_k}{\partial W_{ij}^l}$$

To compute the  $\frac{\partial h_t}{\partial h_k}$  we use simple chain rule over all hidden states in interval  $[k, t]$ .

$$\frac{\partial h_t}{\partial h_k} = \prod_{i=k+1}^t \frac{\partial h_i}{\partial h_{i-1}}$$

Putting equations together, we have the following relationship.

$$\frac{\partial E}{\partial W_{ij}^l} = \sum_{t=1}^T \sum_{j=1}^t \frac{\partial E_t}{\partial y_t} \frac{\partial y_t}{\partial h_t} \left( \prod_{i=k+1}^t \frac{\partial h_i}{\partial h_{i-1}} \right) \frac{\partial h_k}{\partial W_{ij}^l}$$

### 2.2.2 Exploding and Vanishing Gradients

Even though, RNNs had achieved success in learning short-range dependencies, they haven't been showing any worth mentioning achievement with learn-

## 2. RECURRENT NEURAL NETWORK

ing mid-range dependencies. That was mainly caused by problems of *vanishing* and *exploding gradients*, introduced in Bengio et al. (1994).

The exploding gradient problem occurs when backpropagating the error across many time steps, that could lead to exponentially grow of gradient for long-term components. Basically, a small change in parameters at initial stages can get accumulated through the time-steps resulting to the exponential growth. The values of weights can become so large as to overflow and result in *NaN* values.

The vanishing gradient problem refers to opposite behavior when the gradient values are shrinking exponentially fast and eventually vanishing completely. Gradient contributions from later time-steps become zero and the states at those steps don't contribute so we end up not learning long-range dependencies. Vanishing gradients aren't exclusive to RNNs, they also happen in deep ANN.

### 2.2.2.1 Solutions

To overcome problem with exploding gradient we can *gradient clipping* method. To fix the problem of vanishing gradient is little more complicated. We can avoid it by initializing the weights carefully.

## 2.3 LSTM

Long-Short-Term-Memories (LSTM) is special kind of RNN cell, introduced by Hochreiter and Schmidhuber in 1997 \*REFERENCE\*. Conventional RNNs are only just able to learn short-term dependencies because of vanishing gradient problem. However, LSTM are not affected by it and are capable of learning long-term dependencies. RNN are not used anymore, just LSTM.

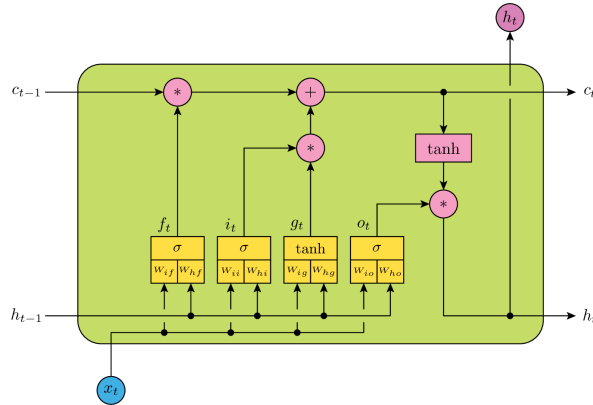


Figure 2.3: Diagram of LSTM cell.

## 2.4 CTC

Connectionist temporal classification (CTC) is a loss function used for classification of sequential data.



# Speech Recognition

Speech recognition is the task of converting speech audio to text representation. It has been attracting researchers for many years with a goal to produce efficient speech recognizer, because it's a very easy and natural human-machine interface tool.

Speech recognition system (ASR) takes audio signal as an input and predicts its transcript. ASR are normally divided into two important stages as shown on \*Figure 1.1\*. The Feature Extractor block generates a sequence of feature vectors which are then fed to the recognizer block generating the correct output word.

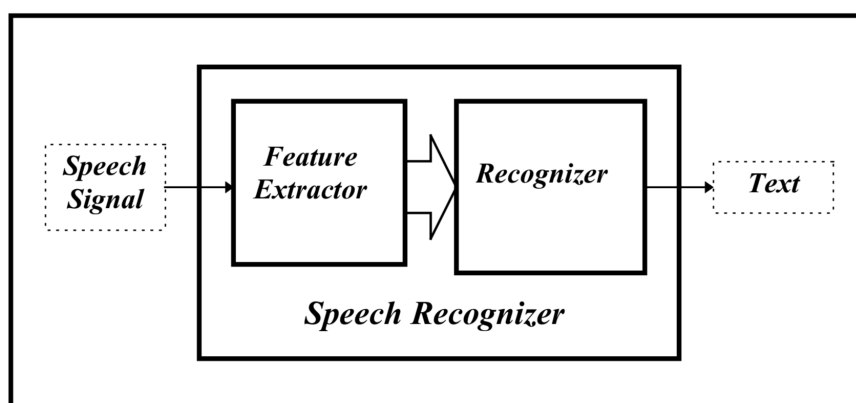


Figure 3.1: Basic building blocks of a Speech Recognizer

## 3.1 Feature Extraction

The feature extraction (FE) block used in speech recognition should aim towards reducing the complexity of the problem, it should derive descriptive features from speech signal to enable a classification of sounds. It is needed because the raw speech signal contains other information besides the linguistic message which would be counterproductive for recognizer.

### 3.1.1 Preprocessing

It is adventegous to apply preprocessing to raw speech signal before moving to feature extraction block. Using some type of preprocessing leads to easier feature extraction and faster training phase.

Usually, speech is recorded with a sampling frequency of 44.1kHz (44,100 readings per second). According to The Shannon Theorem \*Reference\*, a bandwidth limited signal can be reconstructed if the sampling frequency is more than double the maximum frequency meaning that frequencies up to almost 8kHz\* are constituted correctly. Other part of preprocessing is to remove the parts between the recording starts and the user starts talking as well as after the end of speech. That helps to speed up the training phase because it reduces the size of training data\*Reference\*.

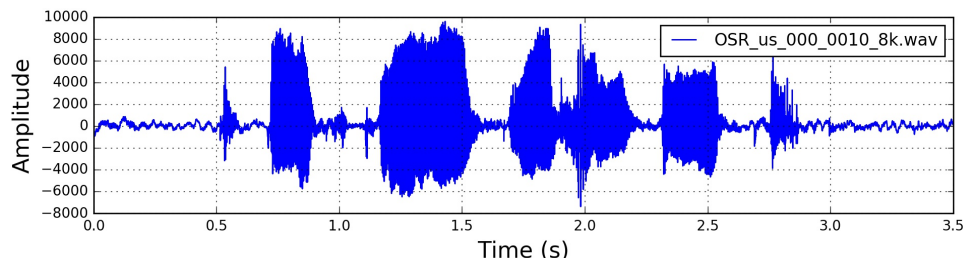


Figure 3.2: Illustration of raw speech signal from wav file with sampling frequency of 8kHz

### 3.1.2 MFCC

Mel Frequency Cepstral Coefficients (MFCCs) are a feature widely used in speech recognition. They were introduced by Davis and Mermelstein in the 1980's, and have been state-of-the-art ever since.

MFCC mimics the logarithmic perception of loudness and pitch of human auditory system and tries to eliminate speaker dependent characteristics by excluding the fundamental frequency and their harmonics.

To obtain MFCC features we have to follow operation steps as shown on \*Figure\*:



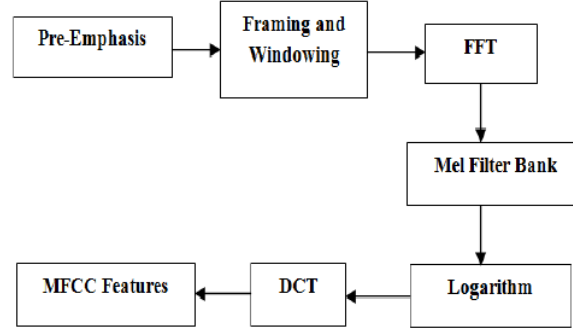


Figure 3.3: Steps of MFCC.

- **Pre-Emphasis** - This step applies filter on the speech signal to amplify the high frequencies. It balances the frequency spectrum and avoids numerical problems during the Fourier transform operation.

$$y(t) = x(t) - \alpha x(t - 1)$$

where  $x(t)$  is amplitude of signal in time  $t$  and  $\alpha$  is filter coefficient which typical values are 0.95,  $y(t)$  pre-emphasis speech signal.

- **Framing** - The process of segmenting the speech signal into a small frames with the length within the range of 10 to 40 milliseconds. Speech is non-stationary signal but we consider all frames behave stationary so they describe a phonemes. In SR we process overlapping frames because phonemes can dependent, resulting to smoother changes in values. Popular settings are 25 ms for the frame size, 10 ms stride (15 ms overlap)\*Reference\*.
- **Windowing** - This step applies Hamming window function\*Reference\* on each speech signal frame. This is common operation for sound signal before applying FFT. \*Reference why\*
- **FFT** - This step converts all speech frames from time domain into frequency domain using Fast Fourier Transform (FFT).\*Reference FFT\*
- **Mel Filter Banks** - This step applies the mel-filterbank which consists of triangular overlapping windows that are spread over the whole frequency range, outputting mel frequency spectrum. It mimics the non-linear human ear perception of sound, these filters are more discriminative at lower frequencies and less discriminative at higher frequencies.
- **Logarithm** - This step computes the logarithm of the mel frequency spectrum, to mimic the human perception of loudness because perceive loudness on a logarithmic scale\*Reference\*.

### 3. SPEECH RECOGNITION

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- **DCT** - This step converts mel spectrum into time domain using Discrete Cosine Transform (DCT)\*Reference\*, resulting to MFCC vectors.

We have just given a theoretical overview how MFCC is calculated, for more detailed explanation consider reading \*Reference MFCC explanation\*. On \*figure mfcc\* is vector of MFCCs calculated from speech signal \*figure signal\* where number of cepstral coefficients is set to 13. We have extracted the features of speech signal and vectors of MFCCs can be feeded to recognizer.

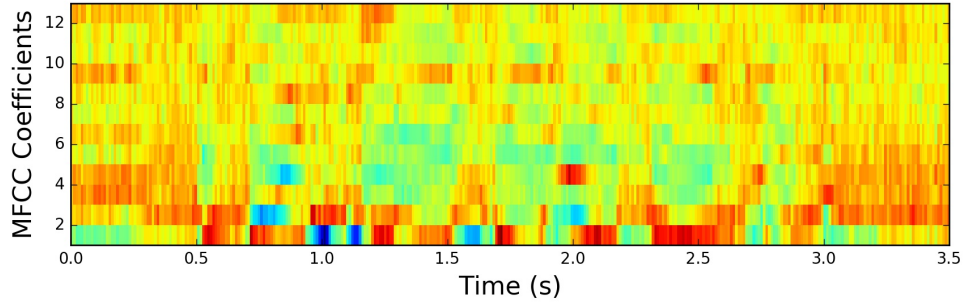


Figure 3.4: Vector of Mel Frequency Cepstral Coefficients through time.

## 3.2 Traditional Speech Recognizers

Historically, most speech recognition systems have been based on a set of statistical models representing the various sounds of the language to be recognized. We can define a problem of speech recognition as maximizing a probability of the word sequence given some utterance.

$$W^* = \underset{W}{\operatorname{argmax}} P(W|X)$$

where  $X$  are acoustic vectors and transcribed  $W^*$  word sequence. However, calculating directly  $W^*$  is a very difficult task. We may simplify it by using Bayes rule resulting to equivalent equation

$$W^* = \underset{W}{\operatorname{argmax}} P(X|W)P(W)$$

where the likelihood  $P(X|W)$  is called the acoustic model and the prior  $P(W)$  is the language model. In traditional speech recognizers we don't form words directly but we concatenating phonemes which are basic building block of words and they are defined by pronunciation model. As shown on \*Figure diagram\*, the decoder block works with language, acoustic and pronunciation model. The language model has a word sequences probabilities, while the acoustic model is generated by Hidden Markov Model (HMM) which is a tool

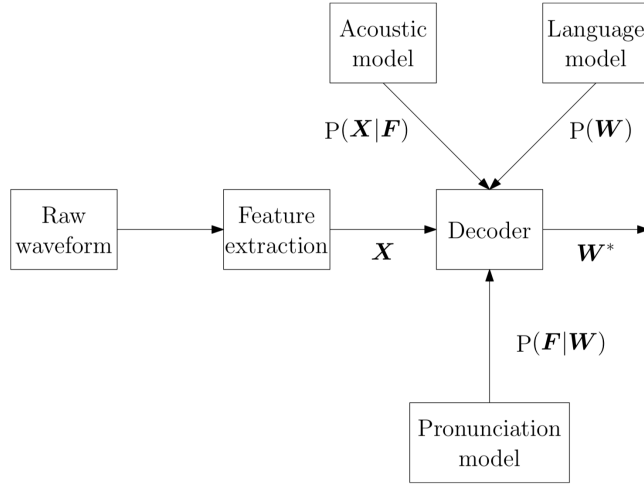


Figure 3.5: Diagram of traditional speech recognizer

for representing probability distribution over sequences of phonemes using pronunciation model.

In this thesis we just give a basic overview how traditional speech recognizers, our main focus is on end-to-end recognizers.

### 3.3 End-to-End Speech Recognizers

Recent advances in algorithms and computer hardware have made it possible to train neural networks(\*section\*) in an end-to-end fashion for tasks that previously required significant human expertise. All of the state-of-the-art speech recognizers were HMM-based, they required pronunciation, acoustic and language model which were hand-engineered and trained separately. Not only speech recognizers based on neural networks require less human effort than traditional approaches, they generally deliver superior performance. Training independent components is complex and suboptimal compared to training all components as one. Because it replaces entire pipelines of hand-engineered components with neural networks, end-to-end learning allows us to handle a diverse variety of speech including noisy environments, accents and different languages. End-to-end speech recognizers simplifies the training and deployment process altogether.

#### 3.3.1 Connectionist Temporal Classification

Connectionist Temporal Classification (CTC) were introduced in \*section\* First step towards end-to-end speech recognizers were due to the introduction

### 3. SPEECH RECOGNITION

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of recurrent neural network (RNN) \*section\* since they are designed to deal with sequenctional data through time.

#### 3.3.2 CTC

Connectionist Temporal Classification make it possible to train RNNs for sequence labelling problems where the input-output alignment is unknown.

## Implementation

The goal is to implement end-to-end speech recognizer using neural network. High-level concept, how the implemented speech recognition system works is illustrated on \*figure\*. It takes a wav file as an input generated from given microphone and performs preprocessing and feature extraction. The data are feeded to the recognizer which outputs the prediction of transcribed text from speech.

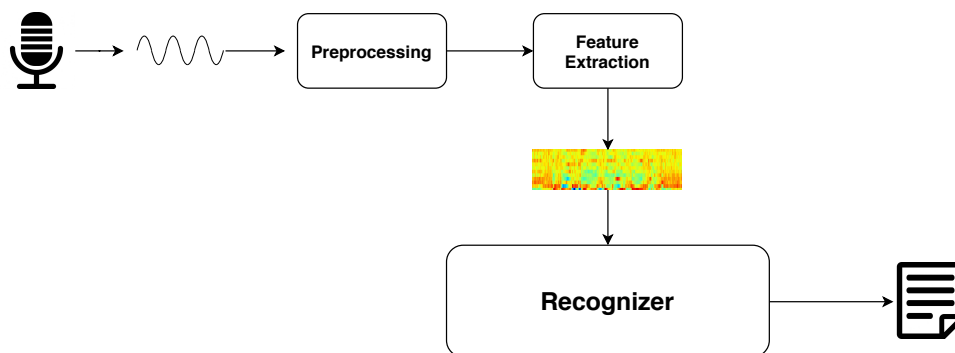


Figure 4.1: Speech Recognition System

The implemented recognizer is build on recurrent neural networks, therefore they need to be trained, in order to make a successful predictions. On \*figure 4.2\* is shown how the recognizer is being trained. It's done by providing speech and transcribed text from the training dataset. RNN feed-forwards all the vectors of MFCC and the RNN's output are processed by CTC. Obtaining the prediction text of the speech signal. Using backpropagation through time algorithm we update the weights and biases of RNN which minimize the error of the loss function resulting to better prediction in future.

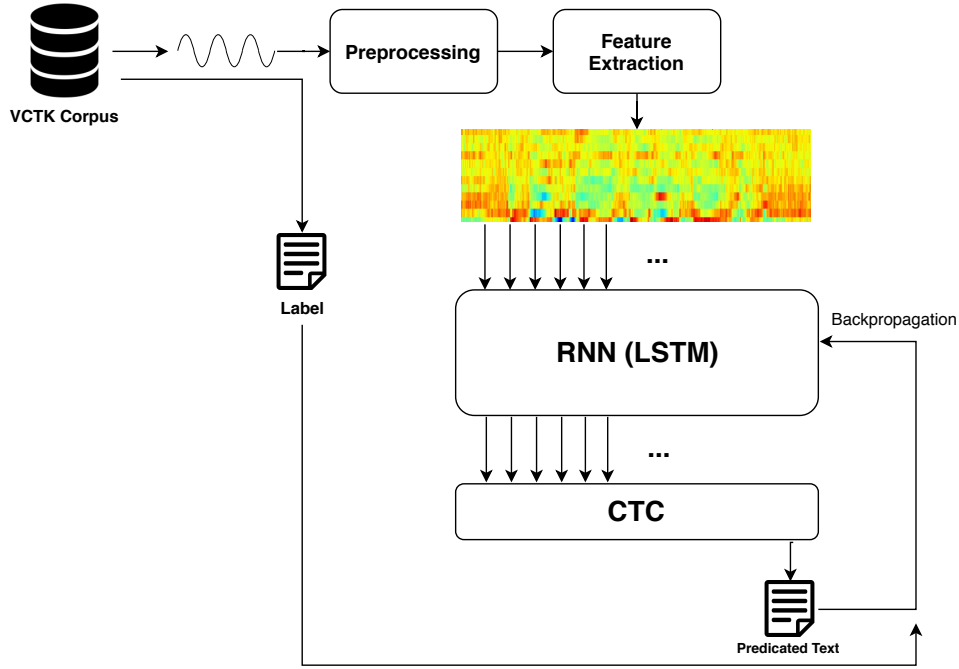


Figure 4.2: Diagram of the learning phase for the speech recognition system

## 4.1 Tools

### 4.1.1 Python

Speech recognition system is implemented in programming language **Python** which is currently most popular approach in machine learning and AI. Python is a very powerful, flexible, open source language that is easy to learn. The greatest strength however is wide range of libraries and frameworks for ML and AI.

### 4.1.2 Tensorflow

TensorFlow is open-source library developed by Google for deep learning and other algorithms involving large number of mathematical operations. The primary unit in TensorFlow is a tensor. A tensor consists of a set of primitive values shaped into an array of any number of dimensions. These massive numbers of large arrays are the reason that GPUs and other processors designed to do floating point mathematics excel at speeding up these algorithms.  
\*Reference\*

TensorFlow programs are structured into a construction phase that assembles a computational graph, and an execution phase that uses a session to execute operation in the graph. However, TensorFlow programs are hard

to debug because of the structure. Fortunately, TensorFlow offers a built-in function for visualization of the computation called TensorBoard.

## 4.2 Training Data

Training data are essential for neural networks performance and its quality, variety, and quantity determine the success of the learning models. Since we use approach of supervised learning for our recognizer, we have to provide labeled data.

### 4.2.1 Dataset Base Class

In source code of the speech recognition system we have class `DatsetBase` which stores path to audios and transactions (labels) from our training dataset. It has also method `next_batch` which takes as a parameter `batch_size` and returns next batch of MFCC vectors and its labels. In method `next_batch` we retrieve speech signal data from audio file and perform preprocessing and feature extraction, then also the text labels are loaded from its file path. Upon the text labels is called preprocessing method which simplifies the text and eliminates all the non-alphabetic characters.

However, retrieving data from file system and performing processing and feature extraction upon them during a training phase is slowing down the process. One of the solution could be to prepare the data beforehand and store it as some variable which would lead to lower retrieving latency.

### 4.2.2 Numbers

Before using my learning model on large training dataset, I had been debugging and validating it on smaller dataset. I have used **Free Spoken Digit Dataset** from Github *\*reference\**. The dataset provides three english speakers with 1500 recordings, 50 recordings for each digit per speaker.

In source code we have a class `DigitDataset` which extends the base class `DatsetBase`. Class `DigitDataset` provides method called `read_digit_dataset` which takes argument of digit dataset path and stores all training data paths in audios and labels variables. They are later used in `next_batch` method.

### 4.2.3 VCTK Corpus

VCTK Corpus is training dataset which includes speech data uttered by 109 native speakers of English with various accents. Each speaker reads out about 400 sentences, most of which were selected from The Herald newspaper *\*reference\**. Even though, this dataset was designed to maximise the contextual and phonetic coverage for HMM-based speech recognizers, we might as well use it for ANN-based speech recognizer. The dataset size is around 15GB which is

still not enough to create robust production ready speech recognizer but it's enough for the purpose of this thesis.

### 4.3 Config Reader

To efficiently use different hyperparameters, datasets or feature extraction configurations. We run the speech recognizer training with *YAML* configuration file.

## 4.4 Preprocessing and Feature Extraction

### 4.4.1 Audio

In source code we have python file `audio_utils` with implements a function called `audiofile_to_input_vector`. The function takes as parameter file path to wav file and the number of cepstrum coefficients. First it loads the wav file from the file system and downsize the sample rate to 16kHz as a part of preprocessing. Even this reduced sample rate contains enough speech information for our recognizer to make successful predication. Then feature extraction is called upon the preprocessed wav file which is done by MFCC. Library `python_speech_features` provides implementation of MFCC method, we just need to configure the used parameters such as the number of cepstrum coefficients, length of window or the length of overlap.

### 4.4.2 Text

In python file `text_utils` we have function `get_refactored_transcript` which takes string and performs multiple operations for simplification. It converts string to lowercases, eliminates all non-alphabetic characters besides the spaces between words. Then string is converted to *numpy* array of characters which gets encoded to integers values. Thanks to the encoding we can simply calculate the loss function for given text label.

## 4.5 Recognizer

Recognizer was created by using TensorFlow library. Before we begin to assemble a computational graph we

### 4.5.1 Computational Graph

TensorFlow requires to assemble a computational graph which will represent the computational steps.



#### 4.5.1.1 CTC Network

In source code we have `CTCNetwork` class representing important features of the network such as input and output dimensions, loss function or used optimizer.

The first method is `generate_placeholders`. *Placeholders* are TensorFlow objects able to store tensors. They don't have to be initialized and input tensors are provided during runtime. Their main purpose is for input and output values. Therefore, the method `generate_placeholders` is creating input and output placeholders for the computational graph. Input placeholder for the network is created as three dimensional array. First dimension represents batch index, second is for number of timesteps and last is for the length of acoustic vector (MFCC vector). For input is also created another placeholder of sequence length for each one on the batched sentences. Output of network is represented by a sparse placeholder because it is required by TensorFlow's CTC.

Second method `loss_funtion` creates CTC loss function inside a computational graph. We use TensorFlow method `tf.nn.ctc_loss` which takes input parameters as a label in sparse matrix format, logits\*footnote\* which is the last layer of the network and sequence length. The TensorFlow method also performs softmax operation upon the input before applying CTC loss.

The third method is `train_optimizer` is defines the used optimizer in the graph. Optimizer is performing some type of gradient descent algorithm to minimize the error on the loss function. There are many optimizer to choose from but currently the recognizer uses one of the most popular and universal optimizer in deep learning which is *AdamOptimizer*.

Another method is `decoder` which decodes predicated sentence from outputted probabilities using argument of input sequence length placeholder and ouput from last layer. It uses TensorFlow method called `tf.nn.ctc_greedy_decoder`. The same output can be decoded also by using `tf.nn.ctc_beam_search_decoder` but it is little slower than the greedy decoder.

Last method is `compute_label_error_rate` which takes parameter as a decoded sparse label and computes its label error rate.

#### 4.5.1.2 LSTM CTC

Class `LSTMCTC` extends from the `CTCNetwork` class and it defines the inner structure of the network. The constructor sets number of layers, hidden neurons, input dimension and the size of acoustic vector.

The class has method `define` which creates the part of the computation graph. It calls parent method for generating placeholders. Creates LSTM cells using `tf.contrib.rnn.LSTMCell` method for all layers then we stack the cells into multilayer RNN networks with method `tf.contrib.rnn.MultiRNNCell`, the stacked network is used in method `tf.nn.dynamic_rnn` which finalizes it

with input placeholders. The method `define` returns the output layer of the network.

### 4.5.2 Training

Training phase of the recognizer is implemented in file `train.py` by method `train_network` which takes dataset and config reader object. The method first has to read the hyperparameters of the network from the config reader and then the computational graph is constructed using the LSTMCTC methods.

In TensorFlow the computation on created graphs are performed inside a `tf.Session()`, thus the training phase is happening inside the session where we loop thorough all the training epoches. In the epoch we train RNN on all training data which are provided using dataset object's method `next_batch`. To run the c we will use function `session.run(fetches, feed_data)`. The *fetches* will be graph operation which are responsible for the training and *feed\_data* are network's placeholder with assigned values from `next_batch` method in dictionary structuer. Example code of running the session for backpropagation algortihm:

```
feed = {
    lstm_ctc.input_placeholder : train_x ,
    lstm_ctc.label_sparse_placeholder : train_y_sparse ,
    lstm_ctc.input_seq_len_placeholder : train_sequence_length
}

batch_cost , _ = session.run([loss_operation , optimizer_operation] , feed)
```

TensorFlow also offers a way of restoring trained networks. During a training we may save checkpoint files with operations variables because `tf.Variable` maintains state in the graph across the computaions. It's achived by an object `tf.train.Saver()`, upon the object we either call method `save(session, checkpoint_path)` or `restore(session, checkpoint_path)`.

## 4.6 Robot NAO

Robot Nao is an autonomous, programmable humanoid robot and the goal is to use the implemented speech recognizer as a voice-user interface.

ALProxy provides remote connection to the NAO robot and gives us access to all the robot's methods. Speech recognizer will be python module running remotly and using ALProxy object we are able to fetch the recorded robot's sound data. The sound data will be processed by the speech recognizer and robot can react to the predicated text.



Figure 4.3: Robot Nao



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## Experiments

In this section we will review the speech recognizer performance. We will introduce some optimization to increase the learning model accuracy and also by tweaking hyperparameters of the network we can achieve better results.

### 5.1 Computing Power

Training neural networks could be considered as computationally difficult problem. However, with the right hardware we can speed up the process significantly. Backpropagation algorithm is mostly about multiplying matrices and GPUs are explicitly designed to handle multiple matrix calculations at the same time, therefore it is highly recommended to use GPUs for training neural networks.

Unfortunately, TensorFlow is just limited on using NVIDIA GPUs to properly work because the python library `tensorflow-gpu` which handles the TensorFlow GPUs computations is built upon CUDA toolkit. Therefore, I will be using CPU for the experiments section as the main computational resource. Because it would not be possible to train speech recognizer on the whole *VCTK dataset*, for the experiment part I will use *Free Spoken Digit Dataset*.

The final training of the speech recognizer using *VCTK dataset* is done on *Floyd Hub* which is commercial platform for cloud computing.

#### 5.1.1 Floyd Hub

FloydHub is a Platform-as-a-Service for training and deploying deep learning models in the cloud. \*TODO\*

### 5.2 First training

The hyperparameters of the first training the speech recognizer on *Free Spoken Digit Dataset*:

## 5. EXPERIMENTS

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- number of hidden neruons - 100
- number of hidden layers - 1
- batch size - 8
- number of epoches - 150
- learning rate - 0.001
- dimension of acoustic vector - 13

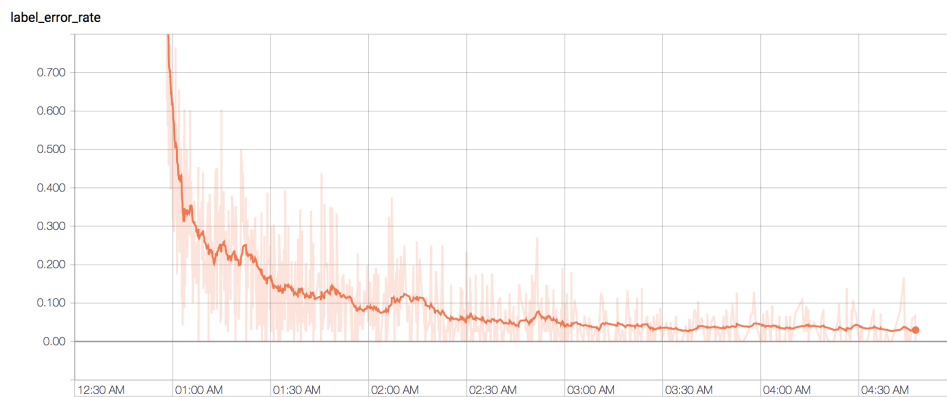


Figure 5.1: Learning Error rate for \*TODO\*

Validation of the speech recognizer performance is evaluated using label error rate. On \*Figure 5.1.\*

### 5.3 Dropout

Optimalization of the learning model can be achived by introducing dropout method.

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# Conclusion

The goal of the thesis was to implement speech recognizer using neural networks which will be later used as voice-user interface on Robot NAO.

First we have introduced the concept of artificial neural networks and basics of the internal network architecture. We have explained the training phase of artificial neural networks and how backpropagation algorithm is used to modify the weights and biases. We have extended the knowledge of neural networks by introducing recurrent neural networks and explained the improved version of backpropagation algorithm, compatible with RNN. However, RNNs presented a problem with vanishing and exploding gradient and we explained a possible solution in type of LSTM cells as an improvement upon basic RNN's neuron.

The speech recognition systems has drastically changed over the last decade. We have given an overview of HMM-based speech recognition system and talked about how the field is starting to shift to end-to-end speech recognizer systems.

We have introduced the speech recognition system based on recurrent neural networks with LSTM cells and CTC loss function on the output layer and proposed the implementation of such a system using TensorFlow library. Validation of such a model was first performed on audio number dataset where

In future work we want to finish the integration of implemented speech recognition with Robot NAO and most importantly train it on more advance speech corpus with deeper network.





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## Zdroje

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## Acronyms

**ANN** Artificial Neural Network  
**RNN** Recurrent Neural Network  
**CTC**  
**MFCC**  
**SR**  
**GPU**  
**CPU**



## Contents of enclosed CD

	readme.txt .....	the file with CD contents description
	exe .....	the directory with executables
	src .....	the directory of source codes
	wbdcm .....	implementation sources
	thesis .....	the directory of $\text{\LaTeX}$ source codes of the thesis
	text .....	the thesis text directory
	thesis.pdf .....	the thesis text in PDF format
	thesis.ps .....	the thesis text in PS format