

COMPUTER SCIENCE TRIPOS - PART II PROJECT

Language Modelling for Text Prediction

May 12, 2017

supervised by
Dr Marek Rei & Dr Ekaterina Shutova

Proforma

Name: **Devan Kuleindiren**
College: **Robinson College**
Project Title: **Language Modelling for Text Prediction**
Examination: **Computer Science Tripos – Part II, June 2017**
Word Count: **11977**
Project Originator: **Devan Kuleindiren & Dr Marek Rei**
Supervisors: **Dr Marek Rei & Dr Ekaterina Shutova**

Original Aims of the Project

The primary aim of the project was to implement and benchmark a variety of language models, comparing the quality of their predictions as well as the time and space that they consume. More specifically, I aimed to build an n -gram language model along with several smoothing techniques, and a variety of recurrent neural network-based language models. An additional aim was to investigate ways to improve the performance of existing language models on error-prone text.

Work Completed

All of the project aims set out in the proposal have been met, resulting in a series of language model implementations and a benchmarking framework for comparing their performance. I have also proposed and evaluated a novel extension to an existing language model which improves its performance on error-prone text. Additionally, as an extension, I implemented a mobile keyboard on iOS that uses my language model implementations as a library.

Special Difficulties

None.

Declaration

I, Devan Kuleindiren of Robinson College, being a candidate for Part II of the Computer Science Tripos, hereby declare that this dissertation and the work described in it are my own work, unaided except as may be specified below, and that the dissertation does not contain material that has already been used to any substantial extent for a comparable purpose.

SIGNED

DATE

Contents

1	Introduction	5
2	Preparation	7
2.1	<i>n</i> -gram Models	7
2.1.1	An Overview of <i>n</i> -gram Models	7
2.1.2	Smoothing Techniques	8
2.2	Recurrent Neural Network Models	11
2.2.1	An Overview of Neural Networks	11
2.2.2	Recurrent Neural Networks	14
2.2.3	Word Embeddings	16
2.3	Software Engineering	16
2.3.1	Requirements	16
2.3.2	Tools and Technologies Used	17
2.3.3	Starting Point	18
3	Implementation	19
3.1	System Overview	19
3.1.1	The Language Model Hierarchy	20
3.1.2	Testing	21
3.2	<i>n</i> -gram Models	21
3.2.1	Computing the Vocabulary	22
3.2.2	Counting <i>n</i> -grams Efficiently	22
3.2.3	Precomputing Smoothing Coefficients	23
3.3	Recurrent Neural Network Models	24
3.3.1	TensorFlow	24
3.3.2	Network Structure	26
3.3.3	Long Short-Term Memory and Gated Recurrent Units	27
3.3.4	Word Embeddings	29
3.3.5	Putting Theory into Practice	30
3.4	Mobile Keyboard	32
3.4.1	Updating Language Model Predictions On the Fly	33
3.5	Extending Models to Tackle Error-Prone Text	34
3.5.1	The Approach	34
4	Evaluation	36
4.1	Evaluation Methodology	36
4.1.1	Metrics	36
4.1.2	Datasets	37
4.2	Results	38

4.2.1	Existing Models	38
4.2.2	On a Mobile Device	41
4.2.3	On Error-Prone Text	42
5	Conclusions	45
	Bibliography	46
A	Backpropagation Recurrence Relation	48
B	Perplexity	50
C	Project Proposal	51

Chapter 1

Introduction

Typing on a mobile device can be a tedious experience. Predictive keyboards serve to somewhat alleviate this, by enabling faster word completion and fewer mistakes. In this project, I investigate a variety of language models for text prediction, and implement a predictive mobile keyboard on iOS. I also introduce a novel extension to an existing language model that improves its performance on error-prone input.

Language models (LMs) produce a probability distribution over a sequence of words, which can be used to estimate the relative likelihood of words or phrases occurring in various contexts. Specifically, given a vocabulary V and a sequence of words $w_1w_2...w_N \in V^N$, they estimate $\mathbb{P}(w_1w_2...w_N)$. Using repeated application of the product rule, this is often reduced to the task of estimating the probability of seeing the word, w_i , given the words that precede it:

$$\mathbb{P}(w_1w_2...w_N) = \prod_{i=1}^N \mathbb{P}(w_i|w_1w_2...w_{i-1}) \quad (1.1)$$

From now on, I will use w_i^j as shorthand for the sequence of words $w_iw_{i+1}...w_j$, where $i \leq j$.

The predictive power of language models is useful in a variety of applications. For example, in speech recognition, if the speech recogniser has estimated two candidate word sequences from an acoustic signal; ‘*it’s not easy to wreck a nice beach*’ and ‘*it’s not easy to recognise speech*’, then a language model can be used to determine that the second candidate is more probable than the first. Language models are also used in machine translation, handwriting recognition, part-of-speech tagging and information retrieval.

$$\begin{array}{rcl} \overbrace{\text{do you want to grab a}}^{w_1^6} & \overbrace{\text{drink}}^{w_7} & \overbrace{(0.327)}^{\mathbb{P}(w_7|w_1^6)} \\ & \text{coffee} & (0.211) \\ & \text{bite} & (0.190) \\ & \text{spot} & (0.084) \\ & \vdots & \vdots \end{array}$$

My project focuses on language modelling in the context of text prediction. For instance, if a user has typed ‘*do you want to grab a* ’, then a language model could be used to

suggest probable next words such as ‘*coffee*’, ‘*drink*’ or ‘*bite*’.

In this project, I focused in depth on the two most prominent types of language model: n -gram models and recurrent neural network (RNN)-based models. I implemented and benchmarked n -gram models coupled with five different smoothing techniques and RNN models of three different flavours on a variety of datasets. I also assessed the trade-offs between prediction accuracy and resource consumption for my model implementations, which became relevant when I applied them in the context of a mobile keyboard.

One problem with existing language models is that their next-word predictions tend to be less accurate when they are presented with text containing errors. Unfortunately, humans are error-prone and inevitably do make mistakes when typing. In this dissertation, I present a novel extension to an RNN-based language model that improves its predictions on error-prone text.

Chapter 2

Preparation

My preparation consisted of background reading and thoroughly understanding various language models, as well as planning how to tie them together in an efficient implementation.

2.1 n -gram Models

This section describes n -gram language models and the various smoothing techniques implemented in this project.

2.1.1 An Overview of n -gram Models

An n -gram is a sequence of n words. n -gram language models provide a way of estimating the probability, $\mathbb{P}(w_i|w_1^{i-1})$, of seeing a word, w_i , given the sequence of words that precede it, w_1^{i-1} . They make the Markov assumption that w_i only depends on the previous $(n-1)$ words. That is:

$$\mathbb{P}(w_i|w_1^{i-1}) \approx \mathbb{P}(w_i|w_{i-n+1}^{i-1}) \quad (2.1)$$

Using the maximum likelihood estimation, $\mathbb{P}(w_i|w_{i-n+1}^{i-1})$ can be estimated as follows:

$$\mathbb{P}(w_i|w_{i-n+1}^{i-1})_{MLE} = \frac{c(w_{i-n+1}^i)}{\sum_w c(w_{i-n+1}^{i-1}w)} \quad (2.2)$$

where $c(W)$ denotes the number of times that the word sequence W was seen in the training set.

This model provides reasonable results and is simple to compute, but it does have one major issue: if, for example, a 3-gram model does not encounter ‘*the cat sat*’ in its training data, then it will assign a probability of 0 to that word sequence. This is problematic, because ‘*the cat sat*’ and many other plausible sequences of words might not occur in the training data. In fact, there are $|V|^n$ possible n -grams for a language model with vocabulary V , which means that as the value of n is increased, the chances of encountering a given n -gram in the training data becomes exponentially less likely.

A crude solution to this problem is to exponentially increase the size of the training set, but this requires significantly more memory and computation, and assumes that additional training data is available in the first place. An alternative solution is to adopt a

technique called *smoothing*. The idea behind smoothing is to ‘smooth’ the probability distribution over the words in the vocabulary such that rare or unseen n -grams are given a non-zero probability. There are a variety of methods that achieve this, and the ones which I have implemented are described in the next section.

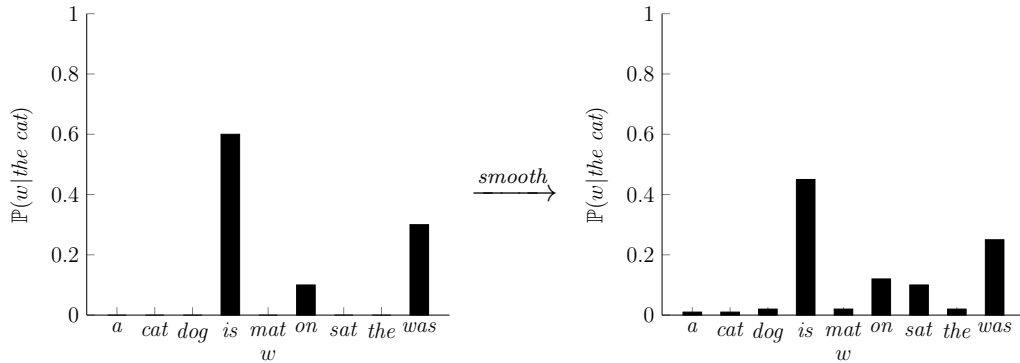


Figure 2.1: A toy example of smoothing, where the vocabulary is $V = \{a, cat, dog, is, mat, on, sat, the, was\}$.

2.1.2 Smoothing Techniques

Add-One Smoothing

Add-one smoothing [1] simply involves adding 1 to each of the n -gram counts, and adding $|V|$ to the denominator ensures the probabilities sum to 1:

$$\mathbb{P}(w_i | w_{i-n+1}^{i-1})_{\text{ADD-ONE}} = \frac{c(w_{i-n+1}^i) + 1}{\sum_w c(w_{i-n+1}^{i-1}w) + |V|} \quad (2.3)$$

One issue with add-one smoothing is that it gives an equal amount of probability to all n -grams, regardless of how likely they actually are. For example, if both ‘*cat is*’ and ‘*cat pizza*’ are unseen in the training data of a bigram model, then $\mathbb{P}(is | cat)_{\text{ADD-ONE}} = \mathbb{P}(pizza | cat)_{\text{ADD-ONE}}$. A simple way to reduce this problem is to employ *backoff*, a technique whereby you recurse on the probability calculated by the $(n - 1)$ -gram model. In this case, it is likely that $\mathbb{P}(is)_{\text{ADD-ONE}} > \mathbb{P}(pizza)_{\text{ADD-ONE}}$, which could be used to deduce that ‘*cat is*’ is more likely than ‘*cat pizza*’.

Absolute Discounting

Absolute discounting employs backoff by interpolating higher and lower order n -gram models. It does this by subtracting a fixed discount $0 \leq D \leq 1$ from each non-zero count:

$$\begin{aligned} \mathbb{P}(w_i | w_{i-n+1}^{i-1})_{\text{ABS}} &= \frac{\max\{c(w_{i-n+1}^i) - D, 0\}}{\sum_w c(w_{i-n+1}^{i-1}w)} \\ &\quad + \frac{D}{\sum_w c(w_{i-n+1}^{i-1}w)} N_{1+}(w_{i-n+1}^{i-1} \bullet) \mathbb{P}(w_i | w_{i-n+2}^{i-1})_{\text{ABS}} \end{aligned} \quad (2.4)$$

where

$$N_{1+}(w_{i-n+1}^{i-1} \bullet) = |\{w \mid c(w_{i-n+1}^{i-1}w) \geq 1\}| \quad (2.5)$$

and the base case of recursion is given by $\mathbb{P}(w)_{\text{ABS}} = \mathbb{P}(w)_{\text{MLE}}$. $N_{1+}(w_{i-n+1}^{i-1}\bullet)$ is the number of unique words that follow the sequence w_{i-n+1}^{i-1} , which is the number of n -grams that D is subtracted from. It is not difficult to show that the coefficient attached to the $\mathbb{P}(w_i|w_{i-n+2}^{i-1})_{\text{ABS}}$ term ensures that the probabilities sum to 1.

Ney, Essen and Kneser [2] suggest setting D to the value:

$$D = \frac{n_1}{n_1 + 2n_2} \quad (2.6)$$

where n_1 and n_2 are the total number of n -grams with 1 and 2 counts respectively.

Kneser-Ney Smoothing

Kneser and Ney proposed an extension to absolute discounting which takes into account the number of unique words that precede a given n -gram [3]. As a motivating example, consider the bigrams ‘*bottle cap*’ and ‘*bottle Francisco*’. If neither have been seen in the training data, and ‘*Francisco*’ occurs more frequently than ‘*cap*’, then the absolute discounting model would backoff onto the unigram distribution and assign a higher probability to the ‘*bottle Francisco*’ bigram, despite the fact that ‘*Francisco*’ usually only follows ‘*San*’.

The idea behind Kneser-Ney smoothing is to instead assign a higher probability to ‘*cap*’, because ‘*cap*’ follows a larger number of unique words, such as ‘*shower*’, ‘*flat*’, ‘*price*’ and so on. This intuition is captured by replacing some of the absolute counts $c(w_i^j)$ with the number of unique words that precede the word sequence w_i^j , $N_{1+}(\bullet w_i^j)$:

$$N_{1+}(\bullet w_i^j) = |\{w \mid c(w w_i^j) \geq 1\}| \quad (2.7)$$

Kneser-Ney smoothing¹ is defined as follows:

$$\mathbb{P}(w_i|w_{i-n+1}^{i-1})_{\text{KN}} = \frac{\max\{\gamma(w_{i-n+1}^{i-1}) - D, 0\}}{\sum_w \gamma(w_{i-n+1}^{i-1}w)} \quad (2.8)$$

$$+ \frac{D}{\sum_w \gamma(w_{i-n+1}^{i-1}w)} N_{1+}(w_{i-n+1}^{i-1}\bullet) \mathbb{P}(w_i|w_{i-n+2}^{i-1})_{\text{KN}} \quad (2.9)$$

where

$$\gamma(w_{i-k+1}^i) = \begin{cases} c(w_{i-k+1}^i) & \text{for the outermost level of recursion (i.e. } k = n) \\ N_{1+}(\bullet w_{i-k+1}^i) & \text{otherwise} \end{cases} \quad (2.10)$$

and the unigram probability is given as:

$$\mathbb{P}(w_i)_{\text{KN}} = \frac{N_{1+}(\bullet w_i)}{\sum_w N_{1+}(\bullet w)} \quad (2.11)$$

¹This is actually the interpolated version of Kneser-Ney smoothing, which differs slightly in form to the equation presented in the original paper.

Modified Kneser-Ney Smoothing

Chen and Goodman noticed that the ideal average discount value, D , for n -grams with one or two counts is substantially different from the ideal average discount for n -grams with higher counts. Upon this discovery, they introduced a modified version of Kneser-Ney smoothing [4]:

$$\mathbb{P}(w_i|w_{i-k+1}^{i-1})_{\text{MKN}} = \frac{\max\{\gamma(w_{i-k+1}^i) - D(c(w_{i-k+1}^i), 0)\}}{\sum_w \gamma(w_{i-k+1}^{i-1}w)} + \lambda(w_{i-k+1}^{i-1})\mathbb{P}(w_i|w_{i-k+2}^{i-1})_{\text{MKN}} \quad (2.12)$$

where γ is defined in Equation 2.10, and λ is defined as:

$$\lambda(w_{i-k+1}^{i-1}) = \frac{D_1 N_1(w_{i-k+1}^{i-1} \bullet) + D_2 N_2(w_{i-k+1}^{i-1} \bullet) + D_{3+} N_{3+}(w_{i-k+1}^{i-1} \bullet)}{\sum_w \gamma(w_{i-k+1}^{i-1}w)} \quad (2.13)$$

and

$$D(c) = \begin{cases} 0 & \text{if } c = 0 \\ D_1 & \text{if } c = 1 \\ D_2 & \text{if } c = 2 \\ D_{3+} & \text{if } c \geq 3 \end{cases} \quad (2.14)$$

where N_1 , N_2 and N_{3+} are defined analogously to N_{1+} in Equation 2.5. Chen and Goodman suggest:

$$D_1 = 1 - 2D \frac{n_2}{n_1} \quad D_2 = 2 - 3D \frac{n_3}{n_2} \quad D_{3+} = 3 - 4D \frac{n_4}{n_3} \quad (2.15)$$

where D is as defined in Equation 2.6.

Katz Smoothing

Katz smoothing [5] is a popular smoothing technique based on the Good-Turing estimate [6], which states that an n -gram that occurs r times should be treated as occurring r^* times, where:

$$r^* = (r + 1) \frac{n_{r+1}}{n_r} \quad (2.16)$$

and n_r is the number of n -grams that occur r times. Converting this count into a probability simply involves normalising as follows:

$$\mathbb{P}(w_i|w_{i-n+1}^{i-1})_{\text{GT}} = \frac{c^*(w_{i-n+1}^i)}{\sum_{r=0}^{\infty} n_r r^*} \quad (2.17)$$

Katz smoothing is then defined as:

$$\mathbb{P}(w_i|w_{i-n+1}^{i-1})_{\text{KATZ}} = \begin{cases} \mathbb{P}(w_i|w_{i-n+1}^{i-1})_{\text{GT}} & \text{if } c(w_{i-n+1}^i) > 0 \\ \alpha(w_{i-n+1}^{i-1})\mathbb{P}(w_i|w_{i-n+2}^{i-1})_{\text{KATZ}} & \text{otherwise} \end{cases} \quad (2.18)$$

where

$$\alpha(w_{i-n+1}^{i-1}) = \frac{1 - \sum_{\{w_i \mid c(w_{i-n+1}^i) > 0\}} \mathbb{P}(w_i|w_{i-n+1}^{i-1})_{\text{KATZ}}}{1 - \sum_{\{w_i \mid c(w_{i-n+1}^i) > 0\}} \mathbb{P}(w_i|w_{i-n+2}^{i-1})_{\text{KATZ}}} \quad (2.19)$$

In practice, the infinite sum in Equation 2.17 cannot be computed. A modified form of Equation 2.18 is presented in Katz's original paper, which makes an approximation to get around this issue.

2.2 Recurrent Neural Network Models

In this section I give a brief introduction to neural networks, recurrent neural networks (RNNs) and how RNNs can be used in the context of language modelling.

2.2.1 An Overview of Neural Networks

The human brain is a furiously complicated organ, packed with a network of approximately 86 billion neurons² that propagate electrochemical signals across connections called synapses. Artificial neural networks were originally developed as a mathematical model of the brain [8]. Despite the fact that they are oversimplified to the point of little resemblance to actual biological neurons, they provide an effective tool for classification and regression in machine learning.

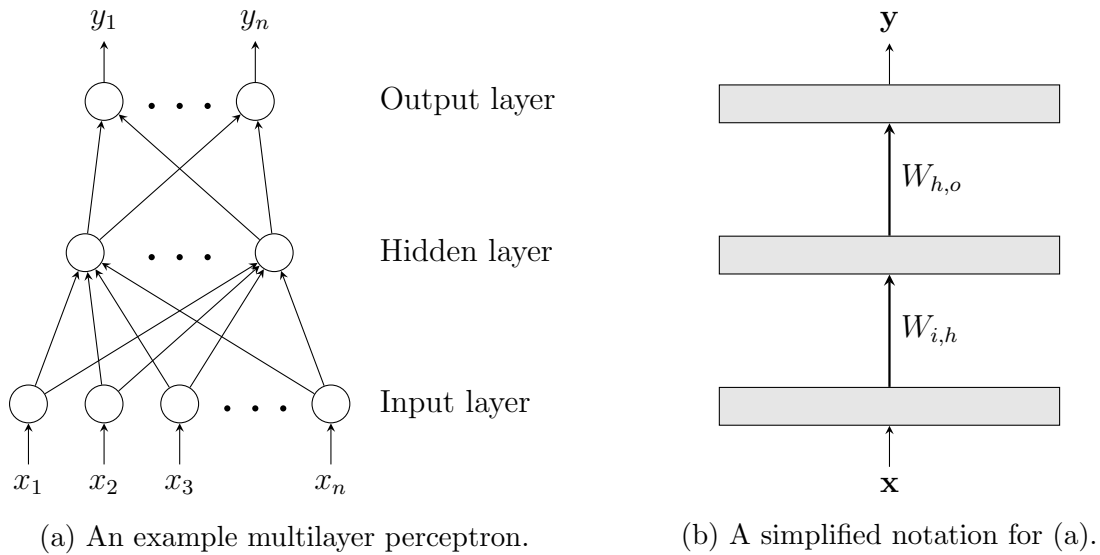


Figure 2.2: Two alternative notations for the same network.

Neural networks consist of a series of nodes which are joined by directed and weighted connections. Inputs are supplied to some subset of the nodes and then propagated along the weighted connections until they reach the designated output nodes. The nodes and weighted connections are loosely analogous to neurons and synapses respectively.

An important distinction to be made is whether the neural network is cyclic or not. Acyclic neural networks are called feed-forward neural networks (FNNs), whereas cyclic neural networks are denoted recurrent neural networks (RNNs) which are covered in Section 2.2.2. There are a variety of FNNs, but the most prominent is the multilayer perceptron (MLP) [9].

The Multilayer Perceptron

The multilayer perceptron consists of layers of neurons, where each layer is fully connected to the next one. The first and last layers are the input and output layers, and any layers

²According to a study by Azevedo et al. [7].

in between are called *hidden layers*. The input neurons simply pass on the input values that they are given. Neurons in the subsequent layers, however, compute the following:

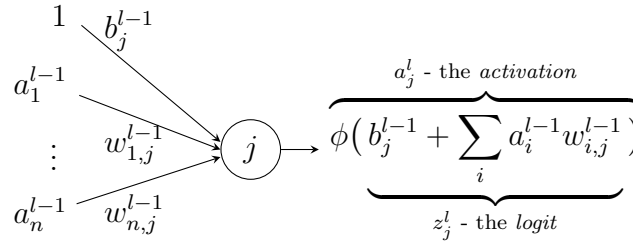


Figure 2.3: The computation carried out by each neuron. The outputs or *activations* of the neurons from the previous layer, $(l - 1)$, are denoted a_i^{l-1} , the weights on the input connections are $w_{i,j}^{l-1}$, the *activation function* is ϕ and b_j^{l-1} is the bias variable.

The constant-valued input 1 is known as a *bias* input, which is weighted by a bias variable b_j^{l-1} . Its purpose is to allow the neuron to shift the output of the activation function left and right by adjusting the value of b_j^{l-1} . Bias inputs are usually attached to every non-input neuron in the network.

It is important that the activation function, ϕ , is chosen to be non-linear. Any combination of linear operators is linear, which means that any linear MLP with multiple hidden layers is equivalent to an MLP with a single hidden layer. Non-linear neural networks, on the other hand, are more powerful, and can gain considerable performance by adding successive hidden layers to re-represent the input data at higher levels of abstraction [10, 11]. In fact, it has been shown that a non-linear MLP with a single hidden layer containing a sufficient number of neurons can approximate any continuous function on a compact input domain to arbitrary precision [12].

Frequently used activation functions include the sigmoid and the hyperbolic tangent functions. Their step-like shape is attributed to the fact that activation functions were originally designed to model the firing activity of neurons. Another important property of these functions is that they are differentiable, which allows for the network to be trained using *gradient descent*, which is discussed below.

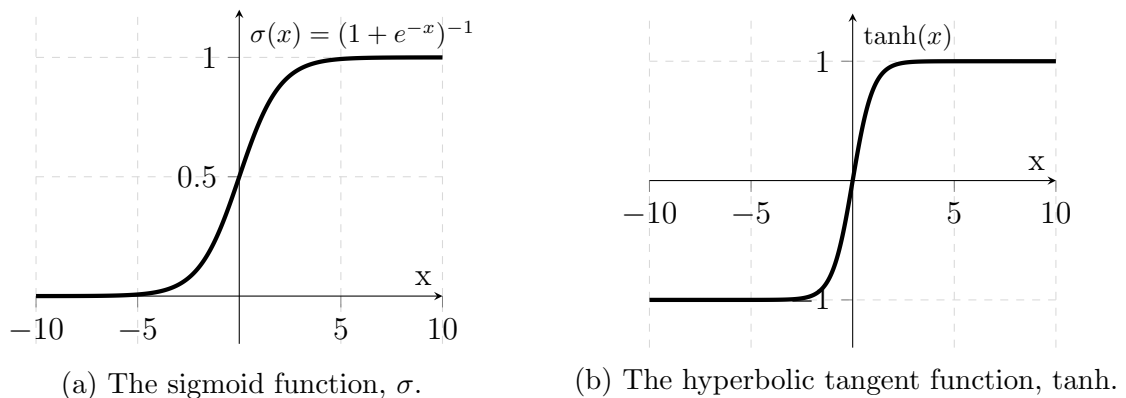


Figure 2.4: Two popular activation functions.

Backpropagation and Gradient Descent

FNNs compute a function, f , parameterised by the weights \mathbf{w} of the network, mapping an input vector $\mathbf{x} = (x_1, \dots, x_n)^T$ to an output vector $\mathbf{y} = (y_1, \dots, y_m)^T$.

$$\mathbf{y} = f_{\mathbf{w}}(\mathbf{x})$$

The entire point of training a neural network is to get it to learn a particular mapping from input vectors to output vectors. What these vectors represent depends on the problem at hand. For example, in the context of classifying pictures of animals, the input vector might represent the pixel values of an image and the output vector might represent a probability distribution over the set of animals in the classification task.

In order to train a neural network, it is supplied with a training set, which is just a list of input-target pairs:

$$\mathbf{s} = ((\mathbf{x}_1, \mathbf{t}_1), \dots, (\mathbf{x}_N, \mathbf{t}_N))$$

where \mathbf{t}_k is the vector that the network should output given the example input \mathbf{x}_k . A differentiable loss function, $\mathcal{L}(\mathbf{y}, \mathbf{t})$, is also defined, which quantifies how well the network output \mathbf{y} matches the target output \mathbf{t} . Then, a measure of how much error the network produces over the whole training set can be defined as follows:

$$\mathcal{L}_{\text{TOTAL}} = \sum_{k=1}^N \mathcal{L}(f_{\mathbf{w}}(\mathbf{x}_k), \mathbf{t}_k) \quad (2.20)$$

The end product of the *backpropagation* algorithm is the partial derivative:

$$\frac{\partial}{\partial w_{i,j}^l} (\mathcal{L}_{\text{TOTAL}})$$

for every weight $w_{i,j}^l$ in the neural network. It is a two stage algorithm that works as follows:

1. Calculate, $f_{\mathbf{w}}(\mathbf{x}_k)$ for each input example \mathbf{x}_k , and use those values to derive $\mathcal{L}_{\text{TOTAL}}$.
2. Given $\mathcal{L}_{\text{TOTAL}}$, calculate $\frac{\partial}{\partial w_{i,j}^l} (\mathcal{L}_{\text{TOTAL}})$ for each weight in each layer of the network by applying the chain rule backwards from the output layer to the input layer.

$\frac{\partial}{\partial w_{i,j}^l} (\mathcal{L}_{\text{TOTAL}})$ is calculated as follows:

$$\frac{\partial}{\partial w_{i,j}^l} (\mathcal{L}_{\text{TOTAL}}) = \sum_{k=1}^N \underbrace{\frac{\partial \mathcal{L}}{\partial f_{\mathbf{w}}(\mathbf{x}_k)} \frac{\partial f_{\mathbf{w}}(\mathbf{x}_k)}{\partial z_j^{l+1}}}_{\delta_j^{l+1}} \frac{\partial z_j^{l+1}}{\partial w_{i,j}^l} = \delta_j^{l+1} a_i^l \quad (2.21)$$

The delta term $\delta_j^{l+1} = \frac{\partial \mathcal{L}}{\partial z_j^{l+1}}$ varies in form for each layer. For the output layer, it has the following form:

$$\delta_j^L = \frac{\partial \mathcal{L}}{\partial f_{\mathbf{w}}(\mathbf{x}_k)} \phi'(z_j^L) \quad (2.22)$$

For the preceding layers, δ_j^l can be defined recursively in terms of the delta values for the neurons in subsequent layers in the network. This is defined below, and a detailed justification of it is given in appendix A:




$$\delta_i^l = \phi'(z_i^l) \sum_j \delta_j^{l+1} w_{i,j}^l \quad (2.23)$$

Gradient descent is an optimisation technique that uses the derivatives produced by the backpropagation algorithm to adjust the weights such that the loss is minimised over the training set. It does this by changing each weight value in the direction of the negative gradient of the loss, which can be visualised as moving downhill on the error surface plotted with respect to the weights in the network:

$$w_{i,j}^l \leftarrow w_{i,j}^l - \eta \frac{\partial}{\partial w_{i,j}^l} (\mathcal{L}_{\text{TOTAL}}) \quad (2.24)$$

$\eta > 0$ is known as the *learning rate*, which determines how large the steps are in the negative direction of the gradient. Ways for setting η , and other useful techniques when training neural networks in practice are described in Chapter 3.

2.2.2 Recurrent Neural Networks

In the context of language modelling, the input is a sequence of words and the output is a sequence of probability distributions over the possible next words at each point. Ideally, the word with the highest probability at each step should be the next word in the input sequence. For example, given (‘the’, ‘cat’, ‘sat’, ...) as input, the network should produce something along the lines of (, , , ...) as output. The problem with FNNs is that these sequences may have varying length, yet FNNs have fixed input and output vector sizes. Recurrent neural networks (RNNs) naturally lend themselves to tackling this issue.

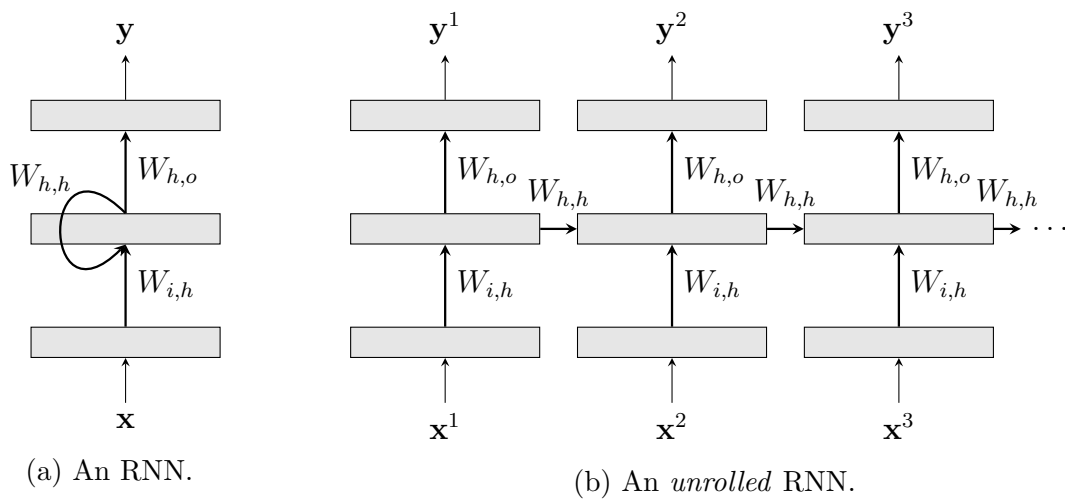


Figure 2.5: Two different representations of the same RNN.

A recurrent neural network can be constructed from an FNN by adding connections from each neuron in a hidden layer back to every neuron in that layer, including itself. A nice

property about RNNs is that they can be unrolled to an arbitrary number of steps without the need for any additional parameters, because the same weight matrices are reused at each step. For this reason, RNNs are well suited for operating over sequences of input data, which is of course ideal in the context of language modelling. The problem of representing textual words as numerical input vectors for an RNN is addressed in Section 2.2.3.

In most applications, the sequences of input vectors are typically ordered by time, so I use the notation \mathbf{x}^t and \mathbf{y}^t to denote the input and output vectors at time step t .

Backpropagation Through Time (BPTT)

Training an RNN is not so much different to training an FNN: the only difference is that gradients must also be calculated across time steps. As shown Figure 2.5b, each weight is reused at every time step, so the weight derivative is now summed across the time steps:

$$\frac{\partial}{\partial w_{i,j}^l}(\mathcal{L}_{\text{TOTAL}}) = \sum_t \delta_j^{l+1,t} a_i^{l,t} \quad (2.25)$$

The delta term for the output layer is defined analogously to Equation 2.22:

$$\delta_j^{L,t} = \frac{\partial \mathcal{L}}{\partial f_{\mathbf{w}}(\mathbf{x}_k^t)} \phi'(z_j^{L,t}) \quad (2.26)$$

The delta term for the preceding layers can be shown to have the form below. This is justified in more detail in appendix A:

$$\delta_i^{l,t} = \phi'(z_i^{l,t}) \left(\sum_j \delta_j^{l+1,t} w_{i,j}^l + \sum_j \delta_j^{l,t+1} v_{i,j}^l \right) \quad (2.27)$$

where $v_{i,j}^l$ denotes the weight on the recurrent connection from neuron i in layer l back to neuron j in the same layer.

RNN Architectures

So far, the equations presented have assumed a *vanilla RNN*. That is, the j^{th} neuron, or *cell*, in layer l at time step t simply computes:

$$a_j^{l,t} = \phi \left(\sum_i a_i^{l-1,t} w_{i,j}^{l-1} + \sum_i a_i^{l,t-1} v_{i,j}^l + b_j^{l-1} \right) \quad (2.28)$$

This computation can be written for an entire layer of hidden neurons, $\mathbf{a}^{l,t} = (a_1^{l,t}, a_2^{l,t}, \dots, a_H^{l,t})$, using matrix notation as follows:

$$\mathbf{a}^{l,t} = \phi(\mathbf{W}^{l-1} \mathbf{a}^{l-1,t} + \mathbf{V}^l \mathbf{a}^{l,t-1} + \mathbf{b}^{l-1}) \quad (2.29)$$

Various improvements to this architecture have been proposed, which capture longer-term information across the inputs. In this project, I have implemented the two most prominent alternatives, which are Long Short-Term Memory (LSTM) [13] and the Gated Recurrent Unit (GRU) [14]. These are outlined in Figure 2.7 and discussed in much more detail in Section 3.3.3.

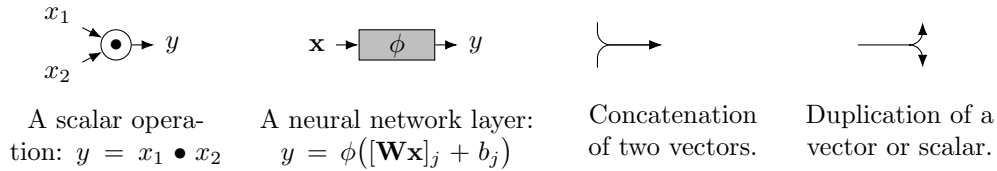


Figure 2.6: A notation for drawing RNN architectures.

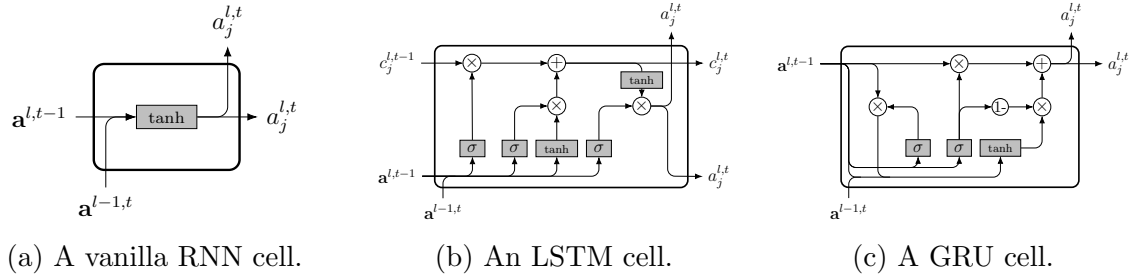


Figure 2.7: Various architectures of RNN neurons, using the notation from Figure 2.6. Each diagram represents the computation of the j^{th} neuron or cell in layer l .

2.2.3 Word Embeddings

In order for an RNN to take a sequence of words as input, each word must first be mapped to a vector representation, known as a *word embedding*, which represents a point in a high-dimensional space. Ideally, similar words will be assigned points which are closer together, so that it is easier for the RNN to generalise to words that it has not seen before.

In this dissertation, I get the RNN to learn good word embeddings through the application of backpropagation and gradient descent to the vector elements.

2.3 Software Engineering

This section details the project requirements and early design decisions that were made.

2.3.1 Requirements

The success requirements set out in the project proposal were:

R1	Language models (LMs) using the following techniques are implemented: <ul style="list-style-type: none"> • n-gram LMs with various smoothing techniques. • A vanilla RNN-based LM. • An LSTM-based LM.
R2	Comprehensible and reliable comparisons between the various LM implementations and their combinations are made regarding their accuracy, speed and resource consumption during both training and inference.
R3	A simple console application is developed to demonstrate the capability of the aforementioned LMs in the context of next-word prediction.

The project proposal also suggested two possible extensions:

E1	Explore extensions to existing language models to improve their performance on error-prone text.
E2	Build a mobile keyboard that exhibits the functionality of the language models.

All requirements and extensions depend on **R1**, so I implemented that first. Afterwards, I built **R2** and **R3** in parallel before completing **E1** and **E2**.

2.3.2 Tools and Technologies Used

Below I describe and justify where necessary the tools and technologies that I used.

Version Control and Build Tools

I hosted my project in a repository on GitHub³, used Git for version control, and used Bazel for running builds and tests on my project.

Machine Learning

I chose to use TensorFlow [15], an open source machine learning library, to assist in the implementation of my recurrent neural networks. There are two key reasons for this:

1. TensorFlow supports *automatic differentiation*. That is, each tensor operation has a partial derivative associated with it, so when you want to apply the second stage of backpropagation, TensorFlow can automatically apply the chain rule through any sequence of tensor operations. Without this, I would have to hard-code new gradient calculations every time I change the structure of the network, which is tedious and error-prone.
2. TensorFlow supports GPU-accelerated matrix operations.

Of course, there are many other machine learning libraries that offer comparable features. The reason I picked TensorFlow in particular was because I was already familiar with its codebase from using and contributing to it during an internship at Google in 2016.

In order to train large scale models on NVIDIA GPUs, I used the University's High Performance Computing service. This uses SLURM for cluster management and job scheduling.

Languages

At the time of writing my code, TensorFlow had both a C++ and a Python API. However, the C++ API only had support for loading and running inference on models, not for training them.

One of the possible extensions I set out in my project proposal was to implement a mobile keyboard that makes use of my language model implementations. Android and iOS both provide little or no support for applications written in Python, but they do support code written in C++ via the JNI and Objective-C++ respectively.

³<https://github.com/DevanKuleindiren/language-modelling>

For these reasons, I wrote code to train and export my RNN-based models in Python, and then wrote some C++ classes for loading and running inference on them. The benchmarking framework and my n -gram models were all written in C++. In order to export and import trained language models across different languages, I used Google’s protocol buffers: a language neutral mechanism for serialising structured data.

Testing

I used Google Test for C++ unit tests and the `unittest` package for testing in Python.

2.3.3 Starting Point

My project codebase was written from scratch, with the assistance of the tools and libraries mentioned above. Apart from a basic knowledge of recurrent neural networks, I had to learn about language modelling, n -gram smoothing techniques, Long Short-Term Memory and Gated Recurrent Units through personal reading.

Experience	Tools and Technologies
<i>Significant</i>	C++, Python, Git, GitHub, TensorFlow
<i>Some</i>	Bazel, iOS, Protocol Buffers
<i>None</i>	Google Test, Objective-C++, SLURM

Table 2.1: A summary of my prior experience.

Chapter 3

Implementation

This chapter details how I implemented a variety of n -gram and RNN-based language models, how I built the mobile keyboard on iOS and how I extended an existing language model to improve its performance on error-prone text.

3.1 System Overview

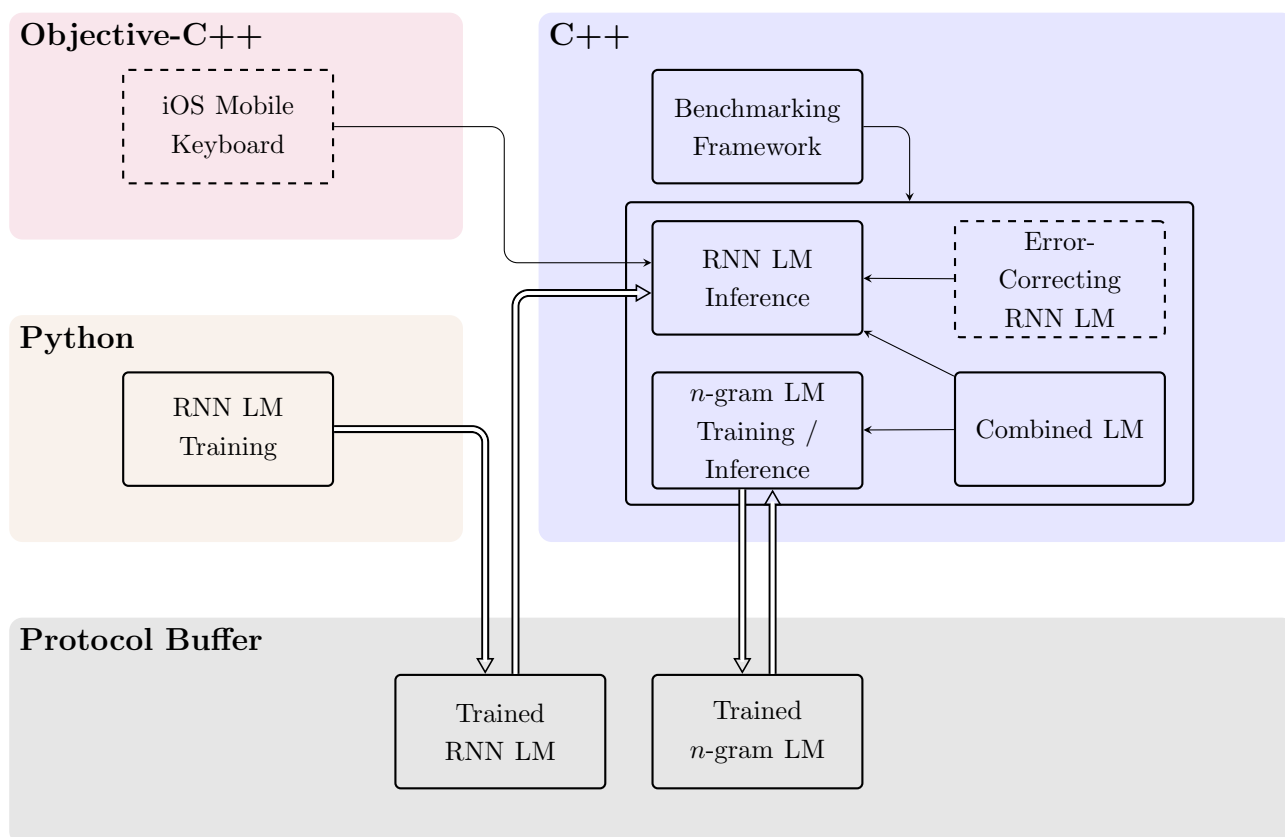


Figure 3.1: An overview of the project. Outlined rectangles denote groups of related files or code, of which the ones with a dashed outline represent project extensions. Thin arrows denote dependencies and thick arrows represent the movement of data.

As shown, all of the code for training and running inference on n -gram models was written in C++, and is described in Section 3.2. The RNN-based language models were trained

in Python using TensorFlow, before being exported into a small group of protocol buffers that can be loaded into C++ for running inference, all of which is detailed in Section 3.3. The ‘Combined LM’ component represents a language model which interpolates the probabilities across one or more n -gram and/or RNN-based language models. This part is not described in depth, because it is mostly trivial. I outline in Section 3.4 how I built a mobile keyboard for iOS that utilises one of the RNN-based language models. Finally, the ‘Error-Correcting RNN LM’ component denotes an extension of my RNN-based language models which I made in order to improve their performance on error-prone text, which is described in Section 3.5.

3.1.1 The Language Model Hierarchy

All of my language model implementations in C++ were joined by the same inheritance hierarchy, at the top of which was a single language model superclass, LM. This is the only class that the benchmarking framework depended on:

```

1 class LM {
2 protected:
3     Vocab *vocab;
4     ...
5 public:
6     ...
7     // Initialise ‘preds’ with the top k most probable next words that follow ‘seq’, in
8     // order of decreasing probability.
9     virtual void PredictTopK(std::list<std::string> seq,
10                             std::list<std::pair<std::string, double>> &preds, int k);
11
12     // Initialise ‘pred’ with the most probable next word that follows ‘seq’.
13     virtual void Predict(std::list<std::string> seq,
14                          std::pair<std::string, double> &pred);
15
16     // Return the probability of the word sequence, ‘seq’.
17     virtual double Prob (std::list<std::string> seq) = 0;
18
19     // Given a sequence of words, ‘seq’, initialise ‘probs’ with the probability of each
20     // word in the vocabulary following that sequence.
21     virtual void ProbAllFollowing (std::list<std::string> seq,
22                                   std::list<std::pair<std::string, double>> &probs) = 0;
23
24     // The same as above, except the word-probabilities are stored in a character trie.
25     virtual void ProbAllFollowing (std::list<std::string> seq, CharTrie *probs) = 0;
26     ...
27 };

```

Every language model object stores a `Vocab` object, which contains the set of words in the language model vocabulary, and a mapping from those words to integer IDs. As a simple memory optimisation, the integer IDs are used in place of the string representations of such words in various language model data structures that are discussed later in this chapter.

`Predict()` and `PredictTopK()` both call `ProbAllFollowing()`. `Predict()` takes the maximum over the result and `PredictTopK()` uses a priority queue to extract the top k predictions. `ProbAllFollowing()` could be implemented by calling `Prob()` for each word in the vocabulary, but this would be inefficient in the case of an RNN-based language

model, which implicitly computes the probability distribution over every word in each forward pass of the network, hence why the methods are kept separate.

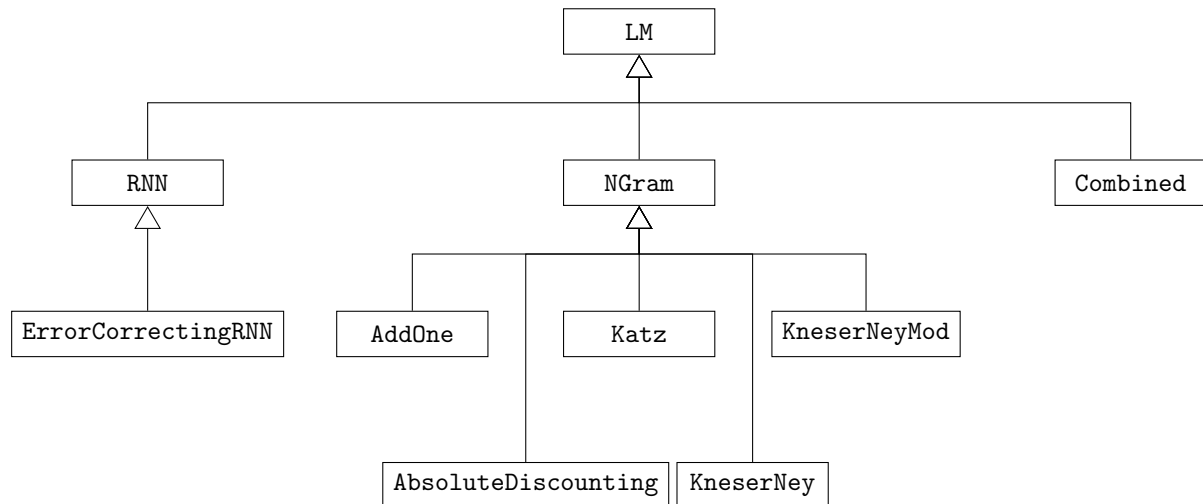


Figure 3.2: An outline of the C++ classes in the language model inheritance hierarchy. Methods and members are omitted for simplicity.

For the RNN-based language models, the differences between the vanilla RNN, GRU and LSTM implementations arise in the Python code where the models are constructed and trained. The C++ code for the RNN-based models just loads and runs inference on the model graph described in the exported protocol buffer. See 3.3.1 for more details.

3.1.2 Testing

To encourage correctness, every non-trivial C++ and Python class in the codebase has an accompanying test class which contains one or more unit tests. In particular, the language model classes are paired with tests that query their `Prob()` and `ProbAllFollowing()` methods and compare the results with values that I calculated by hand using the equations presented in Chapter 2.

All new code was tested before being committed to the codebase, and new test cases were immediately added upon discovery of any bugs.

3.2 n -gram Models

Below I detail my implementation of the n -gram models and their associated smoothing techniques.

One of the first things I discovered when implementing n -gram language models is that there is a trade-off between how much computation is done at training time, and how much computation is done at inference time. For example, at one extreme you could compute only the number of times each n -gram occurs at training time, and leave the computation of the coefficients in the smoothing equations to be calculated on the fly. At the other extreme, you could compute and store the final value of the probability for every

possible n -gram that can be formed from the words in the language model vocabulary, and then return the relevant probability immediately at inference time. The problem with the first approach is that the language model would be too slow at inference time, and the problem with the second approach is that storing absolute probabilities for every possible n -gram would occupy far too much space to be practical. In my implementation, I try to strike a balance between these two extremes by precomputing and efficiently storing all of the relevant coefficients for the smoothing equations, and then combining them at inference time to compute the required n -gram probability.

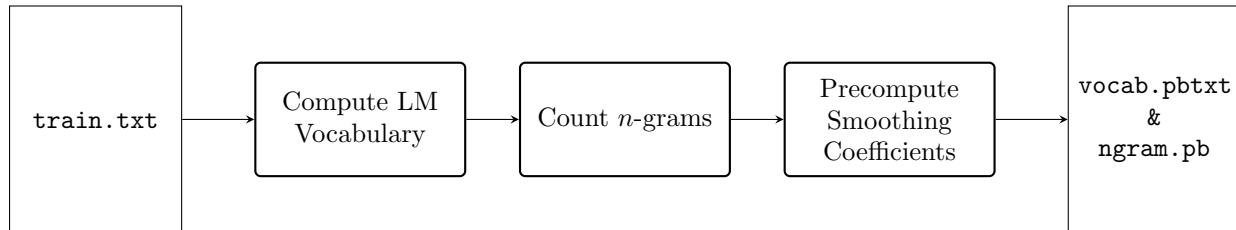


Figure 3.3: The n -gram training pipeline.

The three stages of my n -gram training pipeline are outlined in sections 3.2.1, 3.2.2 and 3.2.3 respectively.

3.2.1 Computing the Vocabulary

The first stage involves a single pass over `train.txt`, where words that occur at least k times are added to the vocabulary and mapped to an integer ID. k is called the *minimum frequency*. A special marker `<unk>` is included in the vocabulary to represent any out-of-vocabulary words. That is, when the model encounters a word it has not seen before, it treats it as `<unk>`. The marker `<s>` is also included and is used to denote the end of a sentence.

3.2.2 Counting n -grams Efficiently

For n -gram models without any smoothing, the only information that needs to be computed at this stage is $c(w_{i-n+1}^i)$ and $\sum_w c(w_{i-n+1}^{i-1}w)$, which are the number of times each n -gram w_{i-n+1}^i occurred in `train.txt`, and the sum of all such counts for all words that follow the $(n-1)$ -gram w_{i-n+1}^{i-1} respectively. Then, when the n -gram model is queried for the probability of w_{i-n+1}^i , it can simply return the first number divided by the second, as required in Equation 2.2.

This situation becomes more complex once smoothing techniques are included. For instance, absolute discounting requires the knowledge of $N_{1+}(w_{i-n+1}^{i-1}\bullet)$, the number of unique words that follow w_{i-n+1}^{i-1} , for every such $(n-1)$ -gram. Moreover, because it employs backoff, it actually requires this knowledge for every k -gram where $1 \leq k < n$. Kneser-Ney smoothing additionally requires $N_{1+}(\bullet w_{i-k+1}^i)$, the number of unique words that precede w_{i-k+1}^i , and $\sum_w N_{1+}(\bullet w_{i-k+1}^i w)$, the number of unique pairs of words that follow and precede w_{i-k+1}^i , for every k -gram where $1 \leq k < n$.

In my implementation, I precompute and store all of these parameters in a temporary word-level trie, as shown in Figure 3.4. This way, each parameter is only ever computed once and is accessible in $O(n)$ time. Given that I only consider n -grams for $n = 2, 3, 4$ or 5 , the access time is practically constant.

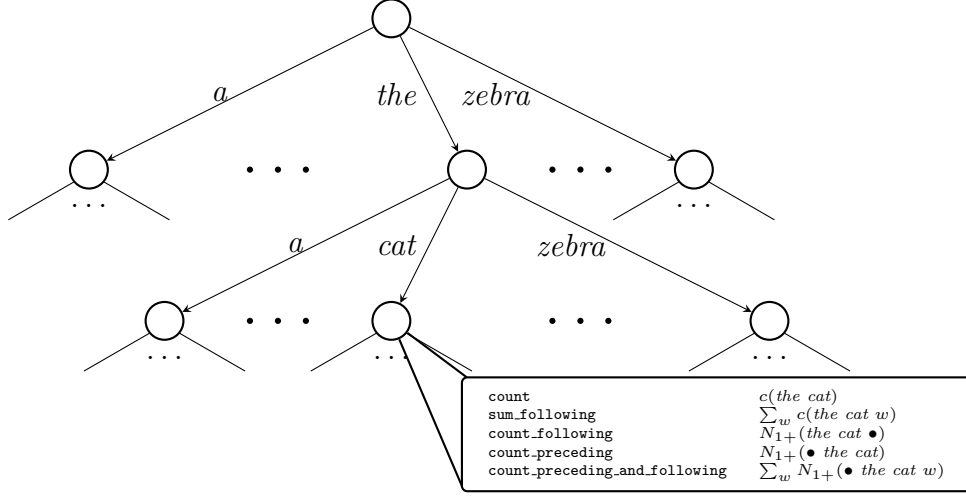


Figure 3.4: An example instance of the **CountTrie** class, which is the end result of the ‘Count n -grams’ stage in Figure 3.3.

Populating a **CountTrie** instance works in two phases. In the first phase, all of the k -grams, where $1 \leq k \leq n$, from the training set are inserted into the trie. At every insertion, the relevant **count** and **count_preceding** values are updated. The second phase involves a full traversal of the trie in which the **sum_following**, **count_following** and **count_following_and_preceding** values are computed. Each of these values can be derived for a particular node by looking at the **count** or **count_preceding** values of its children.

It is worth noting that this data structure is temporary. That is, it is only used to enable fast computation of the smoothing coefficients in the next stage.

3.2.3 Precomputing Smoothing Coefficients

With a few exceptions, most smoothing techniques can be generalised to the following form:

$$\mathbb{P}(w_i | w_{i-n+1}^{i-1})_{\text{SMOOTH}} = \alpha(w_{i-n+1}^i) + \beta(w_{i-n+1}^{i-1}) \mathbb{P}(w_i | w_{i-n+2}^{i-1})_{\text{SMOOTH}} \quad (3.1)$$

The probabilities of many different n -grams share some of the same α and β coefficients. For example, $\mathbb{P}(\text{sat} | \text{the cat})_{\text{SMOOTH}}$ and $\mathbb{P}(\text{jumped} | \text{the cat})_{\text{SMOOTH}}$ would both require $\beta(\text{the cat})$. To avoid duplicated computation, the α and β values for every k -gram in the **CountTrie** instance are computed exactly once and stored in a new word-level trie, called **ProbTrie**. Each node in this trie stores two values, $\alpha(w_i^j)$ and $\beta(w_i^j)$, where w_i^j is the path of words that lead to the node. This way, the parameters can be fetched in $O(n)$ time at n -gram model query time, which again is practically constant since I only use $n \leq 5$. The **CountTrie** instance supplies the necessary parameters for deriving the smoothing coefficients used to populate the **ProbTrie** instance.

Once the `ProbTrie` instance has been populated, computing the probability of any n -gram can be done by traversing the data structure as shown in algorithm 1. Using this implementation, the only thing that each smoothing subclass has to override is the way in which the coefficients α and β are calculated when the `ProbTrie` instance is initially populated. This made it much easier to add and test new smoothing techniques.

Algorithm 1 Computing $\mathbb{P}(w_i | w_{i-n+1}^{i-1})_{\text{SMOOTH}}$

```

1: procedure PROB( $w_i, w_{i-n+1}^{i-1}$ )
2:   if length( $w_{i-n+1}^i$ ) > 0 then
3:      $node \leftarrow \text{GETNODE}(w_{i-n+1}^i)$ 
4:      $\alpha \leftarrow 0$ 
5:      $\beta \leftarrow 1$ 
6:     if  $node$  then
7:        $\alpha \leftarrow node.\alpha$ 
8:       if length( $w_{i-n+1}^{i-1}$ ) > 0 then
9:          $b\_node \leftarrow \text{GETNODE}(w_{i-n+1}^{i-1})$ 
10:        if  $b\_node$  then
11:           $\beta \leftarrow b\_node.\beta$ 
12:       return  $\alpha + (\beta \cdot \text{PROB}(w_i, w_{i-n+2}^{i-1}))$ 
13:   return 0

```

Katz smoothing is an exception in that it does not quite match the form presented in Equation 3.1. To get around this issue, the Katz subclass simply overrides algorithm 1 to traverse the same data structure in a slightly different way.

Once the `ProbTrie` instance is populated, it is exported to a protocol buffer in binary format, `ngram.pb`. The vocabulary is exported to a protocol buffer in textual format, `vocab.pbtxt`.

3.3 Recurrent Neural Network Models

Below, I explain how I implemented my RNN-based language models. To do this, I first give a brief overview of TensorFlow in Section 3.3.1. Then, in Section 3.3.2, I explain the structure of the RNNs I use and how they apply to language modelling. In Section 3.3.3 I give details of the GRU and LSTM architectures. Section 3.3.4 describes the word embeddings used in the models. Finally, Section 3.3.5 outlines how I tuned the RNN hyperparameters and applied techniques to encourage convergence and avoid overfitting.

3.3.1 TensorFlow

TensorFlow [15] is an open-source machine learning library that I used to implement my RNN-based language models.

Computation in TensorFlow is defined in terms of a directed graph, where each node represents an operation and each directed edge represents the flow of a tensor, hence the name. TensorFlow programs consist of two key sections: the first is constructing the graph, and the second is executing one or more paths through the graph in what is called

a *session*. In order to execute a path through a graph, you specify one or more nodes to evaluate, and then TensorFlow automatically executes all operations that are required to evaluate the tensor(s) at the specified node(s).

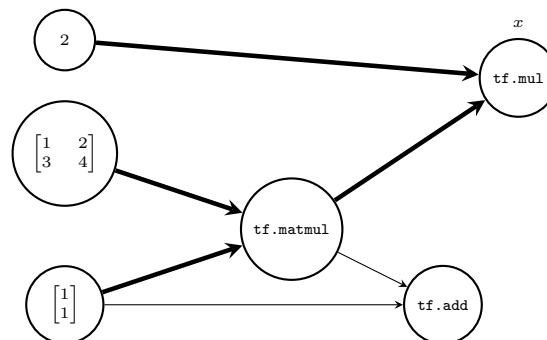


Figure 3.5: A toy TensorFlow computation graph. On evaluation of node x , only the operations on the highlighted paths are computed, and 20 (a rank-0 tensor) is returned.

Each operation takes zero or more tensors as input, and produces a tensor as output. There are some notable operations that take zero inputs: a *constant*, which outputs a tensor whose value cannot be changed; a *variable*, which is the same except the tensor that it outputs can be changed; and a *placeholder*, which is a promise to provide a tensor at execution time.

In my implementation, I described my RNN-based language models as a directed graph of tensor operations. For example: layers of neurons were represented as vectors and the weights in-between each layer were represented as a matrix. In the case of the vanilla RNN, I computed a layer of activations using the matrix operations presented in Equation 2.29. I used a *placeholder* to provide the input training data, and *variables* to store all of the trainable parameters of the network, namely the weights, bias variables and the word embeddings.

Once I constructed my RNN, I ran forward propagation through the network by evaluating the output nodes of the network in a *session*. TensorFlow has a useful method called `tf.gradients(ys, xs)`, which returns the partial derivatives of the sum of the tensor(s) in `ys` with respect to the variables in `xs`. This is known as *automatic differentiation*, which allowed me to compute backpropagation by setting `xs` to the trainable parameters of the network and `ys` to the loss function. Using these partial derivatives, I then applied gradient descent to update the trainable parameters of the network.

After having trained the RNN in Python, I would *freeze* the computation graph and export it into a format that can be loaded into C++. This involved converting all of the variables in the graph to constants and saving the network, along with some metadata, into a protocol buffer. Using the C++ API, I could then load the network and run inference on it by evaluating the relevant tensors in a session, without having to know anything about the internal structure of the RNN.

3.3.2 Network Structure

The general structure of all of my RNN-based language models is described below.

Before the RNN is constructed, an initial pass is made over the training set to construct a vocabulary V . This is done in exactly the same way as for the n -gram models, as described in Section 3.2.1, so I will not repeat the detail.

At the very start of the network is a word embedding lookup table, that maps words to vectors that represent points in a high dimensional space. I set the number of vector elements in each word embedding to be the same as the number of hidden neurons in each hidden layer of the RNN. This was a configurable parameter, H , which was typically set to 256.

Each word embedding retrieved from the lookup table is then fed into the input layer of the RNN, which is followed by the hidden layers and the output layer. Hermans and Schrauwen [16] demonstrate that RNNs with more than one hidden layer naturally employ a temporal hierarchy, where lower layers capture short term interactions among words and higher layers reflect longer-term dependencies. For this reason, I chose to have two hidden layers in my architecture.

There are $|V|$ neurons in the output layer, each of which represent a word in the language model vocabulary. Rather than feeding the logits of the output neurons through an activation function, they are passed through the softmax function. That is, if the vector of logits of the output neurons is denoted \mathbf{z} , then the output of neuron j after the softmax function is defined as $\text{SOFTMAX}(\mathbf{z})_j$, where:

$$\text{SOFTMAX}(\mathbf{z})_j = \frac{e^{z_j}}{\sum_i e^{z_i}} \quad (3.2)$$

The outputs of the softmax function sum to 1 and are used to represent a probability distribution over all of the possible next words. That is, $\mathbb{P}(\text{next word} = j) = \text{SOFTMAX}(\mathbf{z})_j$, where j is the ID of a particular word in the vocabulary. Each ID lies in the range $[0, |V|)$.

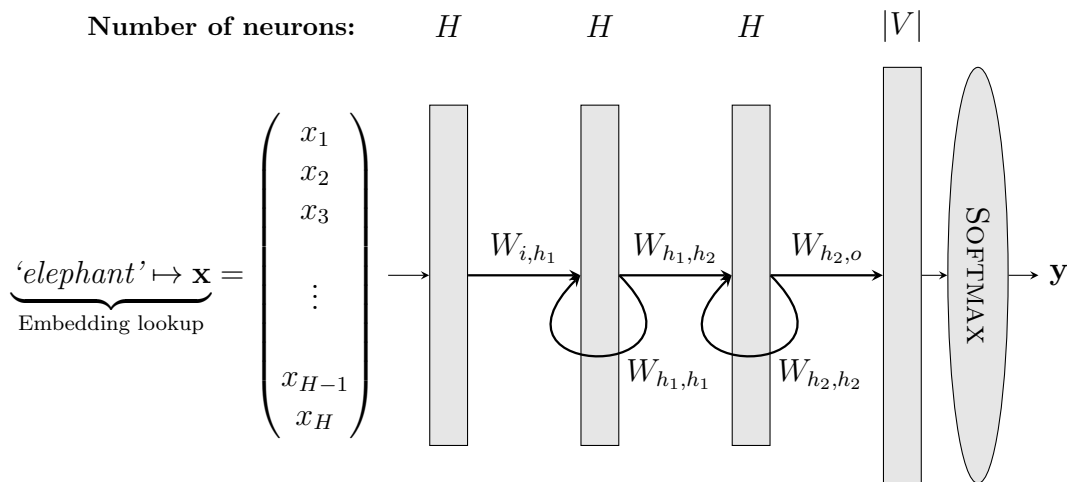


Figure 3.6: The structure of my RNN-based language models.

3.3.3 Long Short-Term Memory and Gated Recurrent Units

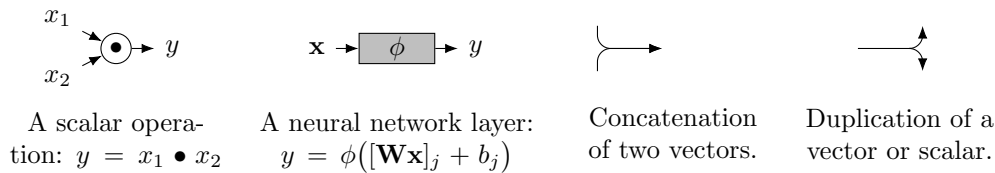
As mentioned in Section 2.2.2, there are various architectures that can be used for the neurons of RNNs, three of which I implemented in this project. Under the vanilla RNN architecture, the vector of neuron activations, $\mathbf{a}^{l,t}$, for layer l at time step t is computed as shown in Equation 2.29, which is repeated below:

$$\mathbf{a}^{l,t} = \phi(\mathbf{W}^{l-1}\mathbf{a}^{l-1,t} + \mathbf{V}^l\mathbf{a}^{l,t-1} + \mathbf{b}^{l-1})$$

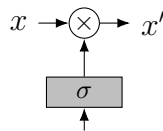
Unfortunately, it is difficult to train vanilla RNNs to capture long-term dependencies, because the influence of the inputs as they cycle through the recurrent connections in the network often either decay or blow up exponentially [17, 18], resulting in the weight gradients either vanishing or exploding disproportionately. These problems are known as the *vanishing gradient problem* and the *exploding gradient problem* respectively. Various RNN architectures have been constructed to improve upon these problems, two of which I implemented in my project and are outlined below.

Long Short-Term Memory

Hochreiter and Schmidhuber [13] originally proposed an architecture that directly tackles the vanishing gradient problem, called Long Short-Term Memory (LSTM). The state of each hidden layer l in their architecture at time step t consists of a tuple $(\mathbf{a}^{l,t}, \mathbf{c}^{l,t})$. $\mathbf{a}^{l,t}$, just like in the vanilla RNN cells, allows the LSTM cells to make decisions over a short period of time. The second state vector, $\mathbf{c}^{l,t}$, serves to retain longer-term information. At time step $t + 1$, each element $c_j^{l,t}$ of $\mathbf{c}^{l,t}$ passes through a single LSTM cell in layer l , and can be modified by a series of three *gates*: the *forget gate*, the *input gate* and the *output gate*. These gates can remove, add and output information respectively from $c_j^{l,t}$.



In what follows, I use the notation from Figure 2.6, which is repeated above for convenience. Each LSTM gate has the following structure:



Each gate controls how much information from x is let through to x' . The sigmoid layer outputs a scalar value in the range $[0, 1]$, where 0 prevents any information from x passing through and 1 allows all of it to pass through.

The LSTM forget, input and output gates in cell j of layer l at time step t are denoted $f_j^{l,t}$, $i_j^{l,t}$ and $o_j^{l,t}$ respectively. If $[\mathbf{x}_1, \mathbf{x}_2]$ denotes the concatenation of vectors \mathbf{x}_1 and \mathbf{x}_2 ,

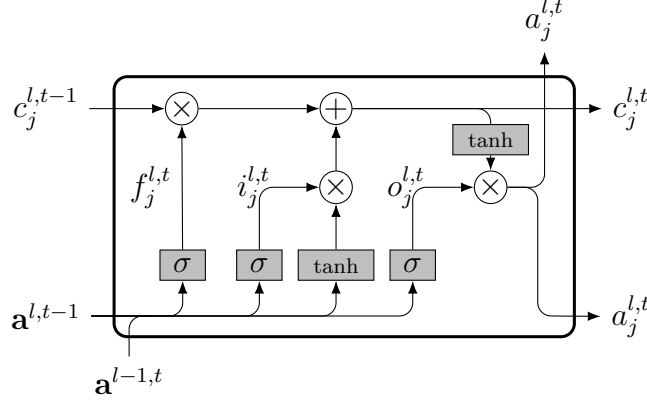


Figure 3.7: A summary of the structure of an LSTM cell. Specifically, this diagram represents the computation of the j^{th} cell in layer l at time step t .

and $[\mathbf{x}]_j$ denotes the j^{th} element of vector \mathbf{x} , then the gate values are computed as follows:

$$f_j^{l,t} = \sigma([\mathbf{W}_f^l[\mathbf{a}^{l,t-1}, \mathbf{a}^{l-1,t}]]_j + [\mathbf{b}_f]_j) \quad (3.3)$$

$$i_j^{l,t} = \sigma([\mathbf{W}_i^l[\mathbf{a}^{l,t-1}, \mathbf{a}^{l-1,t}]]_j + [\mathbf{b}_i]_j) \quad (3.4)$$

$$o_j^{l,t} = \sigma([\mathbf{W}_o^l[\mathbf{a}^{l,t-1}, \mathbf{a}^{l-1,t}]]_j + [\mathbf{b}_o]_j) \quad (3.5)$$

Firstly, the forget gate determines how much of the long-term state value, $c_j^{l,t-1}$, should be ‘forgotten’. Next, there is the opportunity for information derived from the short-term state, $\mathbf{a}^{l,t-1}$, and the previous layer activation, $\mathbf{a}^{l-1,t}$, to be added to the long-term state. The input gate decides how much of this information should be added. The application of the forget and input gates gives the new value of the long-term state value, $c_j^{l,t}$:

$$c_j^{l,t} = f_j^{l,t} \cdot c_j^{l,t-1} + i_j^{l,t} \cdot \tanh([\mathbf{W}_c^l[\mathbf{a}^{l,t-1}, \mathbf{a}^{l-1,t}]]_j + [\mathbf{b}_c]_j) \quad (3.6)$$

Finally, the output gate is used to control how much of the long-term state is used for the cell output, $a_j^{l,t}$:

$$a_j^{l,t} = o_j^{l,t} \cdot \tanh(c_j^{l,t}) \quad (3.7)$$

This structure is summarised in Figure 3.7, and is the version of LSTM that I implemented in my project. There are many variants of LSTM, but I chose this one in particular because it represents a good baseline for the architecture. That is, most of the other versions are some form of extension of the one presented above, such as the notable use of *peephole connections* from Gers and Schmidhuber [19].

Gated Recurrent Unit

Whilst the LSTM architecture provides a significant improvement for retaining long-term information, it is computationally expensive to run. Cho et al. propose the Gated Recurrent Unit [14], which is a simplified version of LSTM that is cheaper to compute. The most significant changes are that it uses two gates rather than three, and it merges the state vectors into one. It consists of a *reset gate*, $r_j^{l,t}$, and an *update gate*, $u_j^{l,t}$:

$$r_j^{l,t} = \sigma([\mathbf{W}_r^l[\mathbf{a}^{l,t-1}, \mathbf{a}^{l-1,t}]]_j) \quad (3.8)$$

$$u_j^{l,t} = \sigma([\mathbf{W}_u^l[\mathbf{a}^{l,t-1}, \mathbf{a}^{l-1,t}]]_j) \quad (3.9)$$

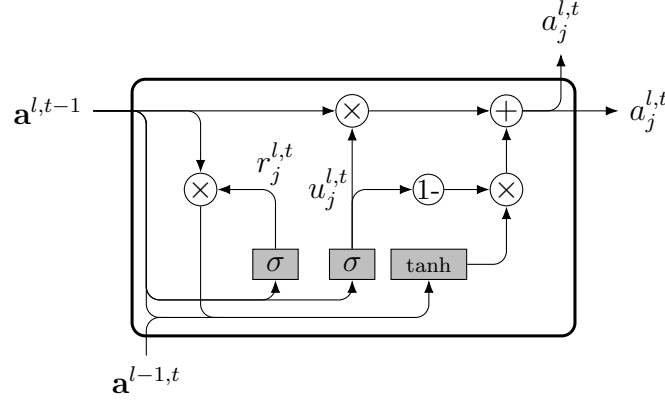


Figure 3.8: A summary of the structure of a GRU cell. Specifically, this diagram represents the computation of the j^{th} cell in layer l at time step t .

The cell output, $a_j^{l,t}$, is then computed as follows:

$$a_j^{l,t} = u_j^{l,t} \cdot a_j^{l,t-1} + (1 - u_j^{l,t}) \cdot \tilde{a}_j^{l,t} \quad (3.10)$$

where

$$\tilde{a}_j^{l,t} = \tanh \left([\mathbf{W}_a^l [\mathbf{a}^{l,t-1} \odot \mathbf{r}^{l,t}, \mathbf{a}^{l-1,t}]]_j \right) \quad (3.11)$$

and \odot represents element-wise multiplication.

The reset gate acts similarly to the LSTM forget gate, in that it allows the cell to drop information that it deems irrelevant. The update gate controls how much information from the previous state carries over to the new one. Cells in this architecture learn to capture either short or long-term information, with the short-term cells having more active reset gates, and the long-term cells having more active update gates.

Abstraction Over the RNN Cells

The LSTM architecture maintains 2-tuples of state vectors, whereas the vanilla RNN and GRU architectures only maintain 1-tuples. Apart from this, the three RNN cells only differ in how they are connected internally. I exploited this in my code by abstracting the implementation of each RNN cell from the overall network structure that was presented in Section 3.3.2, so that new cell architectures could easily be added or tested.

3.3.4 Word Embeddings

As mentioned in Section 3.3.2, each input word is first mapped to a vector representation known as a *word embedding* which is learned as the network is trained. The aim is that similar words will be assigned points in the high-dimensional space which are closer together. This way, the RNN-based language model can more easily generalise to combinations of words that it has not seen before, by understanding which words can be substituted for one another.

In order to visualise such word embeddings, I used principle component analysis (PCA) to reduce the word embeddings to two-dimensional points that can be plotted on a scatter

diagram. PCA is a dimensionality reduction technique that transforms the input data to a new coordinate system such that the first coordinate represents some projection that achieves the greatest variance, the second coordinate the second greatest variance, and so on.

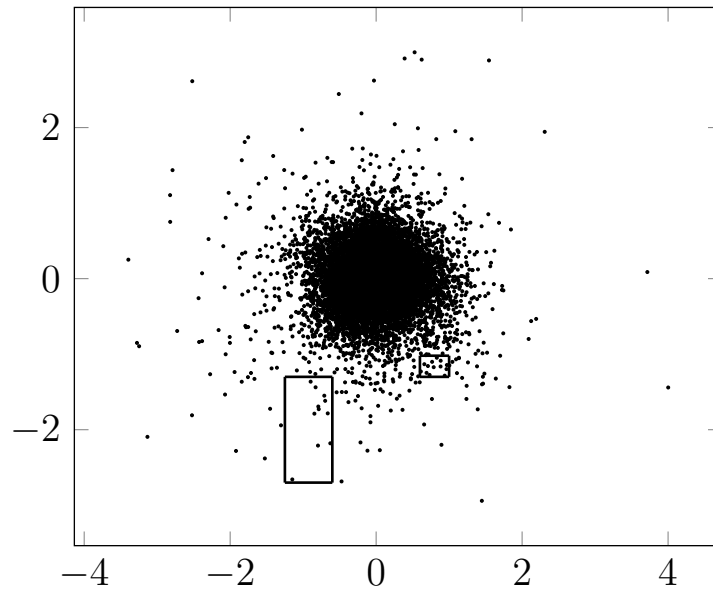


Figure 3.9: LSTM-based language model word embeddings projected onto two dimensions using PCA. The highlighted regions and are shown in Figure 3.10.

The words that have been assigned a more distinguishable representation are given by the points which are further from the centre in Figure 3.9.

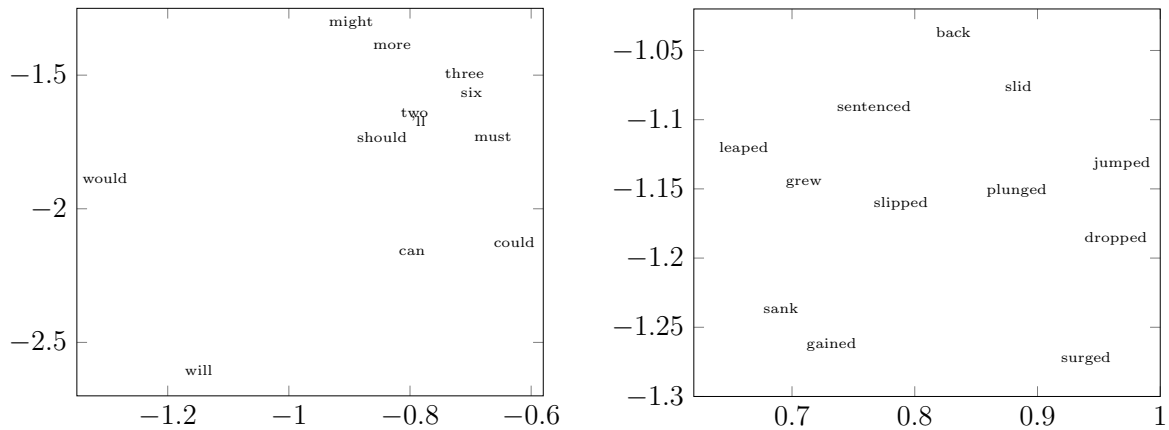


Figure 3.10: Two zoomed in regions of Figure 3.9.

3.3.5 Putting Theory into Practice

In this section, I address a number of problems that are faced when training RNNs in practice and describe how I went about mitigating them.

Underfitting and Overfitting

In any supervised machine learning problem, there are one or more underlying patterns in the training data that need to be learned. *Underfitting* occurs when the model fails to capture these patterns, and is typically the case when the model is too simple for the training data, or it is not trained for long enough. This problem was easy to mitigate by adding more parameters to the model, such as the number of hidden neurons, and training it for a sufficient amount of time.

As well as underlying patterns in the training data, there also often exists some noise which you do not want the model to learn. *Overfitting* occurs when the model learns this noise, which makes it difficult for it to generalise to new data that it has not seen before. When each of my models were trained, a portion of the data called the *test set* was reserved for evaluating the performance of the model. The model was not allowed to see any of the data from the test set during training, so that its ability to generalise to unseen data was fairly assessed. In order to avoid overfitting my RNN-based language models, I also set aside a portion of data called the *validation set*. I used this to monitor, during training, the performance of my models on unseen data. As soon as this performance plateaued, I stopped the training process.

In my implementation, I evaluate the perplexity¹ of my language model on the validation set at the end of each training iteration. The learning rate starts at some initial value, η_0 , and is then halved on every new iteration once the improvement of the perplexity on the validation set between two iterations drops below some threshold, Δ . After this point, when the improvement of the perplexity on the validation set drops below 0.1%, the training process is halted. I tried various values for η_0 and Δ , and chose the best ones for each RNN architecture: $\eta_0 \in \{1, 0.5, 0.1, 0.01, 0.001\}$, $\Delta \in \{10\%, 5\%, 1\%, 0.5\%\}$.

RNNs with a large number of parameters can be particularly susceptible to overfitting. Srivastava proposes a technique called *dropout* for reducing this problem [20], whereby some of the neurons and their connections are randomly dropped from the network during training. Zaremba et al. found that dropout works best in RNNs if it is only applied to the non-recurrent connections [21], which is the method I used in my RNN implementation.

Convergence

Whilst the LSTM and GRU architectures directly combat the aforementioned vanishing gradient problem, there is still the possibility that the gradients can explode, which can make the network unstable during training. In my implementation, I used *gradient clipping* to mitigate this issue, whereby the gradients are clipped once their norm exceeds a certain threshold.

The selection of the initial learning rate value, η_0 , was also important for ensuring convergence in my RNNs. If the learning rate is too large, then the weight updates may overshoot the minimum which they are directed towards. On the other hand, learning rates that are too small result in very slow convergence or the network becoming stuck in local minima.

¹Perplexity is the standard performance measure for language models, detailed in Section 4.1.1.

3.4 Mobile Keyboard

As a project extension, I decided to apply one of my RNN-based language models in a practical context by building a mobile keyboard.

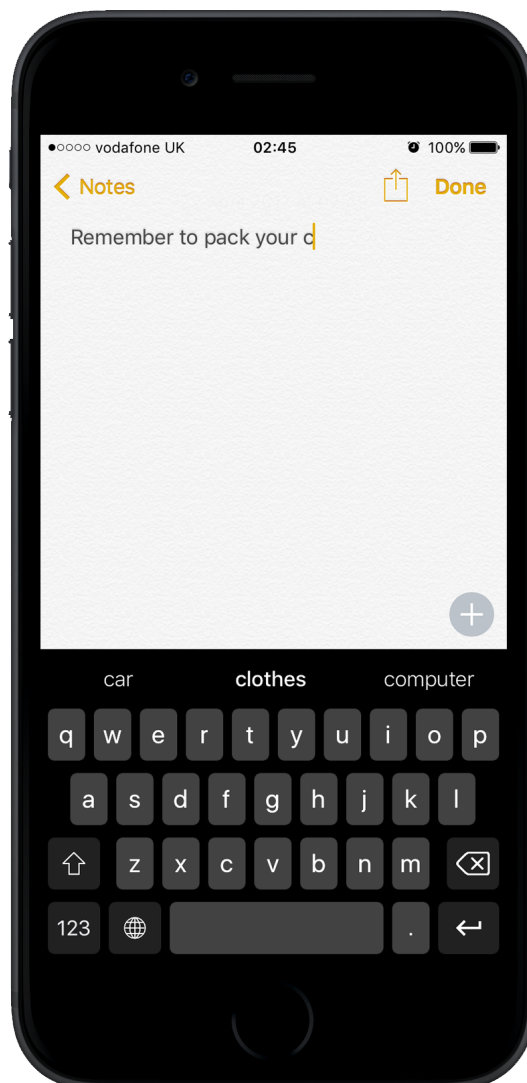


Figure 3.11: The mobile keyboard on iOS. The top three predictions are displayed at the top of the keyboard, and are updated with every character.

iOS applications are written in either Objective-C or Swift. I chose Objective-C so that I could use Objective-C++ to link C++ code into my application. My C++ code, however, depended on TensorFlow, so in order to make my language models accessible within the mobile keyboard, I compiled both my RNN language model implementation and a subset of TensorFlow’s C++ API into a single static library by modifying one of the makefiles already present in the TensorFlow codebase.

To build a mobile keyboard in iOS, you have to implement what is called an ‘App Extension’. App Extensions differ from normal applications in iOS in that their functionality is system-wide. Apple insist that such ubiquitous components should be extremely responsive and light-weight, which means that they have much stricter memory and CPU

limitations than normal applications. Although they do not specify exactly what these limitations are, many people in the developer community estimate that the memory limit is approximately 30 MB.

In order to get an RNN-based language model to run under such memory and CPU limitations, I had to trade-off some of its prediction accuracy. In particular, I reduced the number of hidden layers, the number of hidden neurons and the size of the vocabulary to cut down both the CPU and memory pressure from the model at run time. The final model I used was a 1-layer vanilla RNN with 32 hidden neurons. I also removed the softmax layer, which is technically is not needed to rank the predictions, to further reduce the CPU overhead.

3.4.1 Updating Language Model Predictions On the Fly

A mobile keyboard should present the user with next-word predictions that are updated as they type. In my implementation, I model the state of the input text as a tuple (W, S, C) , where W , S and C represent a sequence of words, spaces and non-space characters respectively. For example, the text in Figure 3.11 would be represented as:

$$\underbrace{\text{Remember_to_pack_your}}_W \quad \underbrace{_}_S \quad \underbrace{c}_C$$

With every new character insertion or deletion, this state is implicitly updated. The interaction with the language model as a response to each insertion or deletion is illustrated in Figure 3.12.

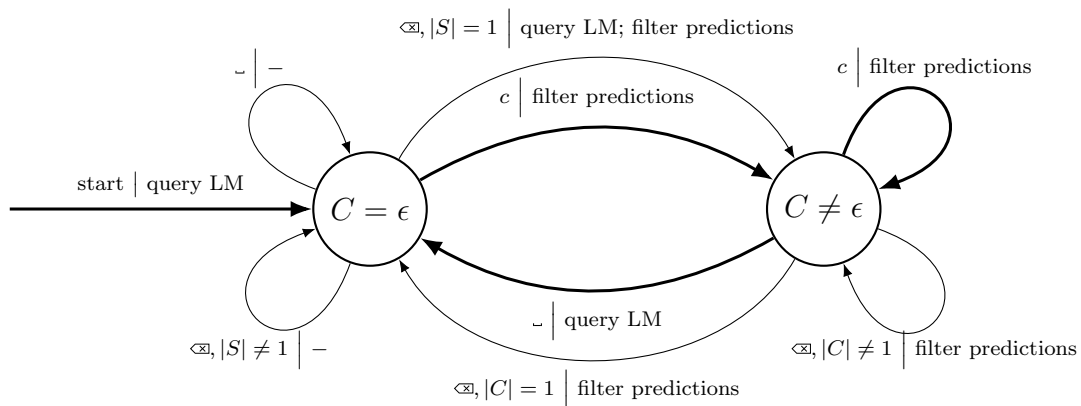


Figure 3.12: An FSM representing the language model interaction. The thick arrows denote the path taken when typing without any deletions or unnecessary spaces.

c , $_$ and \boxtimes represent a non-space character insertion, a space insertion and a backspace respectively. ‘Query LM’ represents the process of querying the language model with W . The language model vocabulary is stored in a character-level trie, and the probabilities for each word from the query result are inserted at the nodes of the trie. ‘Filter predictions’ represents the process of traversing this trie using C and taking the top three most probable words on the current branch. This approach finds the most probable next-word predictions that begin with the characters that the user has already typed for the current

word, whilst avoiding unnecessary language model queries.

As well as the basic input events illustrated in Figure 3.12, my implementation also caters for: the shift key, caps lock, dragging the cursor to a new position, switching between letters and special characters, selecting displayed predictions and auto-capitalisation of new sentences.

3.5 Extending Models to Tackle Error-Prone Text

An issue with existing language models is that their predictions suffer when presented with error-prone text. For example, consider the two word sequences ‘*the weather is*’ and ‘*the whether is*’. Whilst in the first case a language model might successfully predict words such as ‘*sunny*’ or ‘*nice*’, in the second case it will likely struggle to suggest reasonable word candidates. Evidence to support this claim is given in Table 4.2 in Section 4.2.3.

In the context of text prediction, it is unfortunately very rare that a human user will not make any mistakes, which makes this an important issue to address. In this section, I propose an extension to a language model which aims to improve its performance on text containing errors.

3.5.1 The Approach

My approach is to add a layer of error correction between the input text and the language model. There are lots of different types of error that can occur in the English language, but my implementation only focuses on correcting spelling mistakes.

Error-correction can be broken up into two parts: detecting the errors and making suggestions to fix them. In the case of spelling mistakes, detection can be done with a single dictionary lookup. Candidate corrections for a spelling mistake should be similar to the original word and probable in the surrounding context. In my implementation, I use the Levenshtein distance as a measure of how similar two words are, which is the minimum number of single character insertions, deletions and/or substitutions that need to be made to get from one word to the other. I also use the language model itself to gauge how probable candidate corrections are in the surrounding context. Specifically, when searching for candidate replacements for a word w that is preceded by the words w_i^j , I feed w_i^j to the language model, iterate through the next-word predictions in order of decreasing probability and select the first word that has an edit distance within some threshold δ of the original spelling, as described in algorithm 2.

I implemented this approach as an extension to my RNN-based language models, hence the use of `FEEDFORWARD()` and *state* in algorithm 2.

Algorithm 2 Computing $\mathbb{P}(w_j | w_i^{j-1})_{\text{ERROR_CORRECTING}}$

```

1: procedure PROB( $w_j, w_i^{j-1}$ )
2:    $outputs \leftarrow \emptyset$ 
3:    $state = 0$ 
4:   for  $w$  in  $w_i^{j-1}$  do
5:      $w_{correct} \leftarrow w$ 
6:     if  $w \notin \text{dictionary}$  and  $outputs \neq \emptyset$  then
7:       SORTBYDECREASINGPROBABILITY( $outputs$ )
8:       for  $w'$  in  $outputs$  do
9:          $d \leftarrow \text{LEVENSHTEINDISTANCE}(w, w')$ 
10:        if  $d \leq \delta$  then
11:           $w_{correct} \leftarrow w'$ 
12:          break
13:     $outputs, state \leftarrow \text{FEEDFORWARD}(w_{correct}, state)$ 
14:  return  $outputs[w_i]$ 

```

Chapter 4

Evaluation

In this chapter, I first describe the benchmarking framework that I built to evaluate the language models, before proceeding onto the results. The results section is threefold: firstly I present the performance of the existing language models that I implemented, secondly I focus on the tradeoffs faced when employing those models on a mobile device, and finally I display my findings in language modelling on error-prone text.

4.1 Evaluation Methodology

4.1.1 Metrics

In the context of text prediction, there are essentially two questions one might want to answer when evaluating a language model:

Q1. How accurately does the language model predict text?

Q2. How much resource, such as CPU or memory, does the language model consume?

I implemented a benchmarking framework that can return a series of metrics which are outlined below. The first two are concerned with **Q1** and the latter three relate to **Q2**.

Perplexity

Perplexity is the most widely-used metric for language models, and is therefore an essential one to include so that my results can be compared with those of other authors. Given a sequence of words $w_1^N = w_1 w_2 \dots w_N$ as test data, the perplexity PP of a language model L is defined as:

$$\text{PP}_L(w_1^N) = \sqrt[N]{\frac{1}{\mathbb{P}_L(w_1^N)}} = \sqrt[N]{\prod_{i=1}^N \frac{1}{\mathbb{P}_L(w_i | w_1^{i-1})}} \quad (4.1)$$

where $\mathbb{P}_L(w_i | w_1^{i-1})$ is the probability computed by the language model L of the word w_i following the words w_1^{i-1} . This somewhat arbitrary-looking formulation is more thoroughly justified with a bit of information theory in appendix B. The key point is that *lower values of perplexity indicate better prediction accuracy* for language models trained on a particular training set.

One issue with perplexity is that it is undefined if $\mathbb{P}_L(w_i|w_1^{i-1})$ is 0 at any point. This is problematic for n -gram models with no smoothing, since they assign 0 probabilities to unseen n -grams. In order to make this model comparable to the others under this metric, I replaced probability values of 0 with the small constant `1e-9`.

Average-Keys-Saved

Average-keys-saved is a more user-oriented metric, which is defined as the number of characters that they can avoid typing as a result of the correct next words appearing in the top three predictions, averaged over the number of characters in the test data. As an example, if the user is typing `science` and the word `science` appears in the top three predictions after they have typed `sc`, then that would count as 5 characters being saved, averaging at $\frac{5}{7}$ keys saved per character. Averaging over the number of characters ensures that the results are not biased by having longer words in the test data.

Memory Usage

This is measured as the amount of physical memory in megabytes occupied by the process running the language model, retrieved from the `resident_size` member of the `task_basic_info` struct¹ for the current process.

Training Time

The amount of time it took to train the language model.

Average Inference Time

This is measured as the amount of time in milliseconds that the language model takes to assign a probability to all of the words in its vocabulary given a sequence of words, averaged over a large number of sequences.

4.1.2 Datasets

Penn Tree Bank (PTB) Dataset

The PTB [22] is a widely-adopted dataset of 1,036,580 words for measuring the quality of language models, derived from text from the Wall Street Journal. I used it for all tests in which the size of the training data is fixed.

One Billion Word (1BW) Benchmark

This is a much larger dataset produced by Google of approximately 1 billion words [23]. I used this dataset for tests in which the size of the training data is a variable under investigation.

¹http://web.mit.edu/darwin/src/modules/xnu/osfmk/man/task_basic_info.html

First Certificate in English (FCE) Dataset

The FCE dataset is a subset of the Cambridge Learner Corpus, which is a dataset of exam scripts written by candidates sitting an FCE examination in 2000 and 2001 [24]. In this project I make use of a preprocessed version of this dataset, consisting of 88,130 words, in which error-prone and error-free versions of the exam scripts are aligned line by line in separate files. I used this dataset when exploring the performance of language models on error-prone text.

4.2 Results

4.2.1 Existing Models

Smoothing techniques and the value of n in n -gram models

The first set of language models that I built were n -gram models, along with a series of smoothing techniques for improving their predictions on less frequent n -grams.

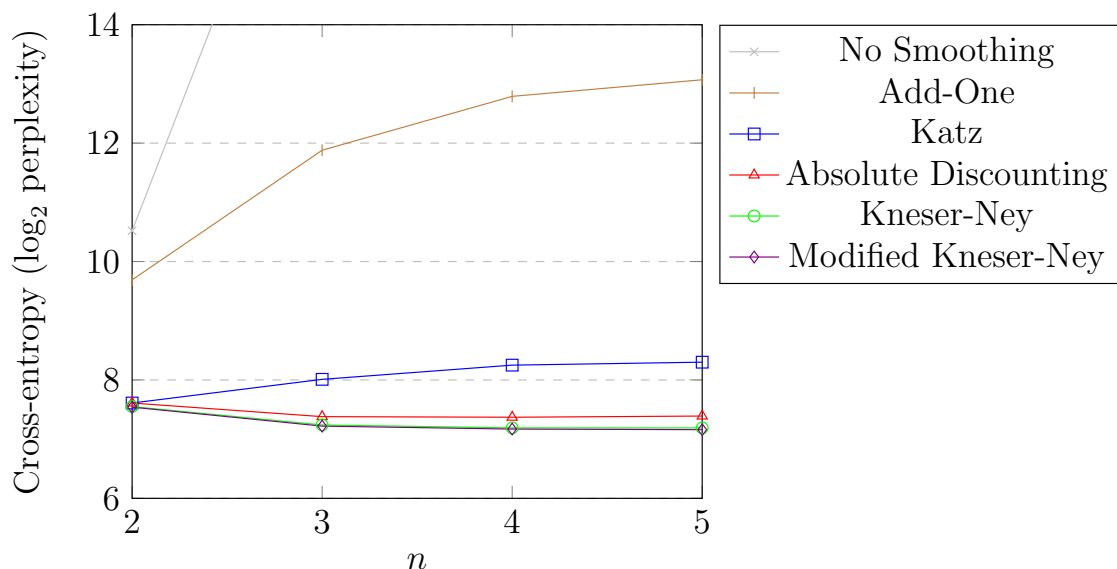


Figure 4.1: The cross-entropy of n -gram models trained on the PTB dataset.

Cross-entropy is the binary logarithm of perplexity, and lower perplexity scores indicate better prediction accuracy. With this in mind, it is clear that Kneser-Ney-based smoothing offers the best prediction performance amongst the n -gram models.

The change in performance with the value of n is interesting, because one might expect that increasing n will always yield better results. For n -gram models with add-one or no smoothing, this is not the case, because they do not employ backoff. At higher values of n , n -grams are much more sparse, so without any backoff these models can only rely on sparse counts, resulting in lower probabilities being assigned to plausible sequences of words. Katz smoothing does employ backoff, and achieves much better performance, but it still distributes too much probability to rare n -grams, which is why its performance also drops with n .

A comparison of RNN-based models with n -gram models

I implemented three RNN-based language models which differ in their cell architecture: vanilla RNN, Gated Recurrent Unit and Long Short-Term Memory.

Some datasets are more predictable than others, which means that metrics like perplexity and average-keys-saved depend on which dataset is used. In order to produce results that are comparable with the work of other authors, I have tested my language models on the PTB dataset, as shown in Table 4.1. The RNN-based models used 2 hidden layers each with 256 neurons. Average-keys-saved, an expensive metric to compute, was only taken over the first 1000 words of the test set.

Language Model	Perplexity	Average-Keys-Saved	Memory Usage (MB)	Training Time (min,secs)*	Average Inference Time (ms)
3-gram	4.54×10^5	0.35014	266.91	11s	62
3-gram + add-one	3764.96	0.53063	266.94	11s	41
3-gram + Katz	256.95	0.68482	266.71	14s	88
3-gram + absolute disc.	166.03	0.72178	266.78	13s	63
3-gram + KN	150.73	0.72466	266.88	14s	54
3-gram + modified KN	149.54	0.72355	266.97	14s	54
5-gram	1.96×10^8	0.07167	737.36	26s	130
5-gram + add-one	8610.45	0.33886	737.30	26s	63
5-gram + Katz	314.49	0.67154	737.43	41s	156
5-gram + absolute disc.	167.38	0.72333	737.43	40s	126
5-gram + KN	146.35	0.72598	737.37	44s	114
5-gram + modified KN	142.68	0.72554	737.53	50s	116
Vanilla RNN	131.03	0.72776	253.67	15m 10s	39
Gated Recurrent Units	114.52	0.73993	271.39	28m 35s	37
Long Short-Term Memory	112.47	0.73617	287.13	18m 59s	38
LSTM, 5-gram + MKN (av)	96.07	0.75719	929.39	19m 49s	189
LSTM, 5-gram + MKN (int)	94.70	0.75830	927.20	19m 49s	190

**The n -gram models were trained on my laptop, whereas the neural models were trained on a GPU cluster on the High Performance Computing service.*

Table 4.1: A benchmark of various language models on the PTB dataset.

There is a lot of information that can be drawn from Table 4.1, so I will highlight only the most important and interesting points:

- The RNN-based language models outperform the n -gram models in terms of prediction accuracy. In my implementation, they run slightly faster at inference time and also consume less memory, although this is partly because the n -gram models are entirely unpruned. Obtaining the correct hyperparameters and training the RNN-based language models, however, does take significantly longer.
- Mikolov benchmarked 3-gram + MKN, 5-gram + MKN, a 1-layer vanilla RNN and

a 1-layer LSTM model with 100 hidden neurons on the PTB dataset, for which he obtained perplexity scores of 148.3, 141.2, 124.7 and 115 respectively [25, 26]. All of these results closely match my own, with slightly more deviation in the RNN-based results due to his models having a different number of neurons and hidden layers.

- Ignoring insignificant differences in memory and time, absolute discounting, Kneser-Ney smoothing and modified Kneser-Ney smoothing equal or outperform add-one and Katz smoothing in every metric.
- LSTM yields marginally better prediction accuracy than the GRU architecture, but requires more memory. Apart from the memory overhead, GRUs outperform the vanilla RNN architecture.
- Interestingly, a substantial improvement in perplexity and average-keys-saved can be obtained by simply averaging the probabilities produced by an n -gram and an RNN-based language model. This seems to imply that the two classes of language model complement one another, in the sense that RNN-based models make strong predictions when some of the n -gram predictions are weak, and vice versa.
- The combined language models can be improved even more by interpolating their probabilities rather than just averaging them. The result shown for the interpolation of 5-gram modified Kneser-Ney with LSTM used $\lambda = 0.38$, where the probability was calculated as $\lambda(\textit{n-gram probability}) + (1 - \lambda)(\textit{LSTM probability})$. The perplexity as a function of λ was approximately U-shaped, so I obtained the value $\lambda = 0.38$ by repeatedly evaluating the models on a subset of the test data and narrowing down the interval in which the minimum was positioned.

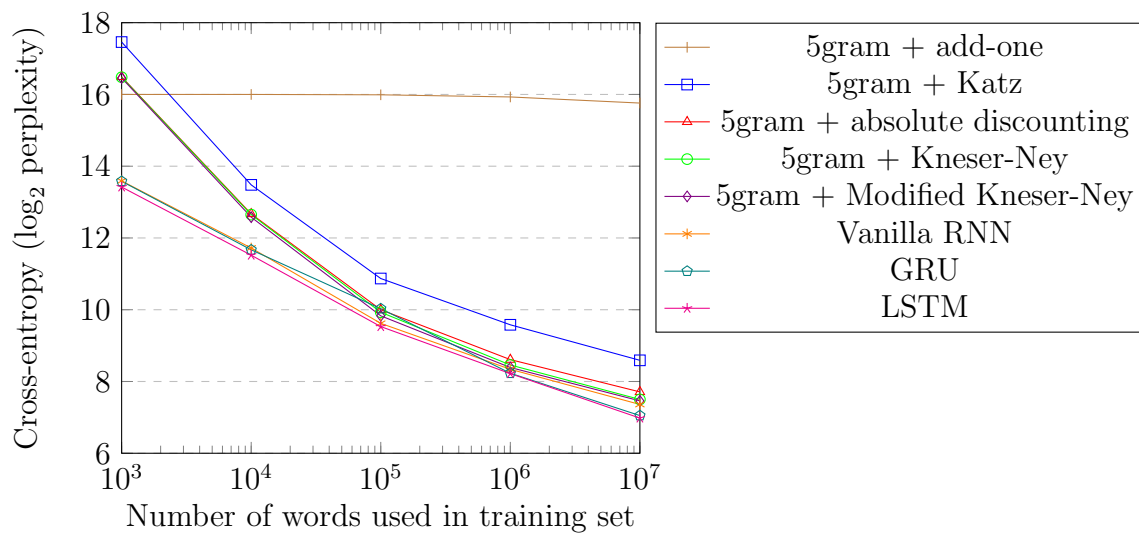


Figure 4.2: The cross-entropy of various language models with respect to the training set size, using the 1BW dataset.

As shown in Figure 4.2, I also investigated the effect of the training set size on prediction accuracy. The RNN-based language models consistently outperform the n -gram models. This difference is most significant for small training sets, and starts to become more pronounced again once the training set grows large beyond 10^6 words.

Finally, I present a qualitative comparison of a smoothed n -gram model with an RNN-based model in Figure 4.3.

money is the root of a new york city.	money is the root of the nation's largest public bank.
the meaning of life is a unit of the company's shares outstanding.	the meaning of life is the best way to get the way to the public.
science and was up N N from \$ N million or N cents a share.	science's most recent issue of the nation's largest economic indicators.

(a) 5gram + modified Kneser-Ney smoothing (b) 2-layer LSTM with 256 hidden neurons

Figure 4.3: Sentences generated from two different language models trained on the PTB dataset. The bold words were given to the model, and the non-bold words were generated until a full stop was predicted.

4.2.2 On a Mobile Device

In this section I explore the trade-offs between language model resource consumption and prediction performance.

Two obvious ways of decreasing the memory footprint of a language model are decreasing the vocabulary size and, for RNN-based models, decreasing the number of hidden neurons. What is less obvious, is how quickly the average-keys-saved drops with these two parameters.

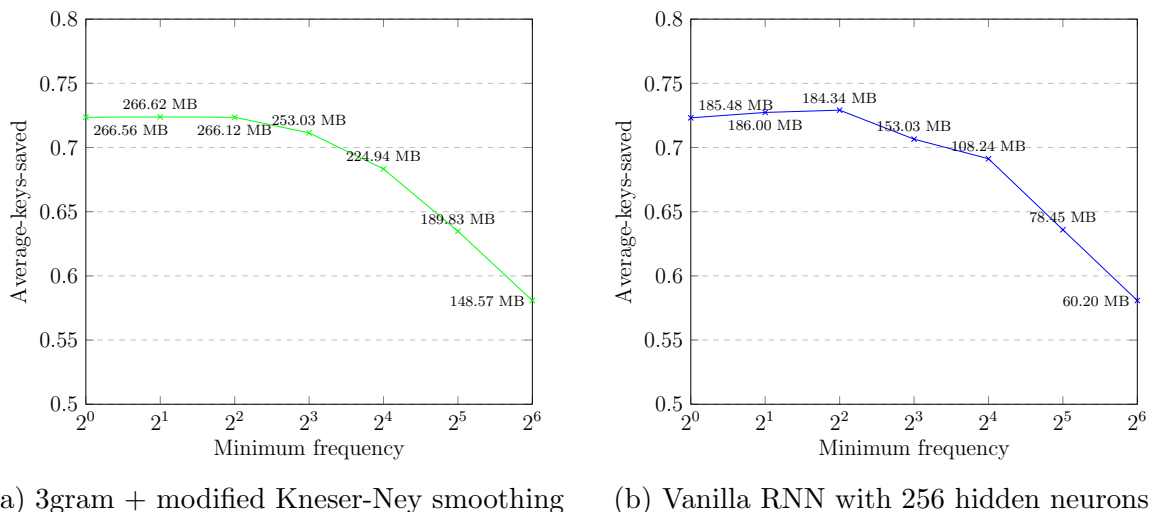


Figure 4.4: The effect of changing the vocabulary size for models trained on the PTB dataset.

I altered the vocabulary size by changing a parameter called the *minimum frequency*, which defines the minimum number of times a word must appear in the training set to be considered part of the language model vocabulary.

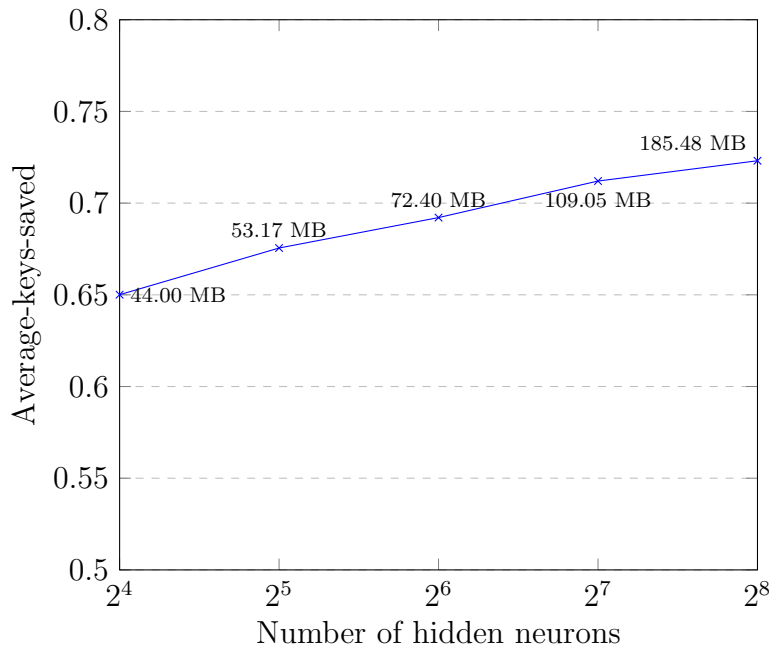


Figure 4.5: The effect of changing the number of hidden neurons in a 2-layer vanilla RNN trained on the PTB dataset.

Comparing figures 4.4 and 4.5, it can be seen that much greater reductions in memory usage can be achieved for a given loss in average-keys-saved by decreasing the number of hidden neurons. Dropping from 256 to 32 hidden neurons in each layer gives a memory saving of 132.31 MB, with an average-keys-saved loss of 0.04758. It is intriguing that such a reduced vanilla RNN model can achieve a better average-keys-saved score than the 5-gram model with Katz smoothing from Table 4.1.

Of course, there are several other techniques that can be used to lower the memory and CPU pressure of language models on mobile platforms. These include, but are not limited to: reducing the number of hidden layers; using 16-bit floats for model parameters; removing the softmax layer from the RNN; and memory-mapping the model.

4.2.3 On Error-Prone Text

In this section I present the performance of my language model extension on error-prone text. The goal is to produce a language model whose predictions are not deterred by errors in the input text, as illustrated in Figure 4.6.

Normally, language models are evaluated by feeding them a sequence of input words (the *inputs*) and seeing how well they predict that same sequence of input words shifted forward in time by one word (the *targets*). When evaluating my error-correcting language models, I did exactly the same thing, except that the inputs were allowed to contain errors in them and the targets were not.

I used the FCE dataset, which consisted of six files:

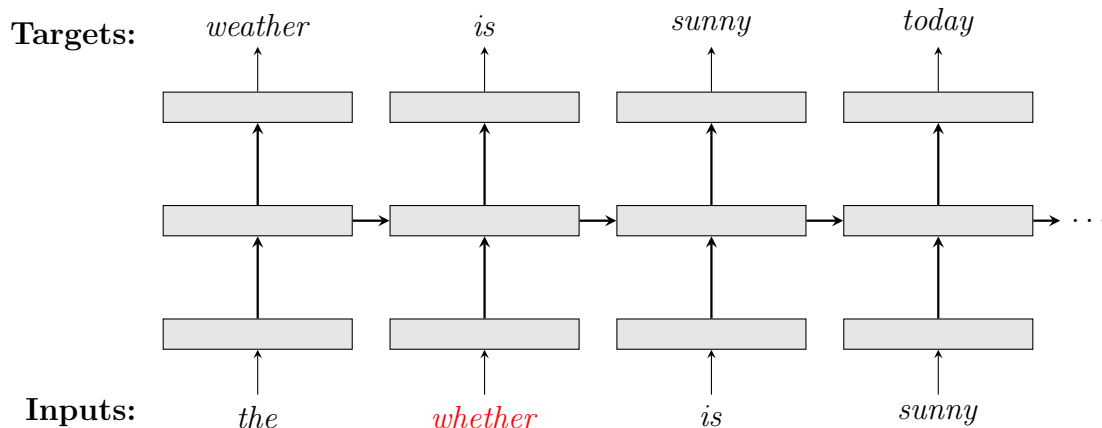


Figure 4.6: The ideal error-correcting language model.

```

train.correct  train.incorrect
valid.correct  valid.incorrect
test.correct   test.incorrect

```

The training, validation and test sets all consisted of one file containing uncorrected text and another file containing the corresponding corrected text. I trained a 2-layer LSTM-based language model with 256 hidden neurons on `train.correct`, and used `valid.correct` to guide the learning rate decay. In order to focus the evaluation on the error-correcting ability of my language models, I removed any pairs of lines from the test set files that were identical. I also removed any pairs of lines that contained corrections that involve the insertion or deletion of words, because my model was not designed to handle these types of errors, as explained in Section 3.5. This left 6672 words in each of the test files.

The ideal error-correcting language model would predict words as if the input were not error-prone, therefore, to obtain an *upper bound* in performance, I evaluated the LSTM-based language model using `test.correct` for the input words and `test.correct` for the target words. In order to establish a *lower bound*, I evaluated the performance of the same model using `test.incorrect` for the input words and `test.correct` for the target words. This model did not contain any error-correcting capabilities.

Language model	Perplexity	Average-keys-saved
<i>Upper bound</i>	77.83	0.65483
<i>Error-correcting LSTM; $\delta = 1$</i>	87.99	0.64453
<i>Error-correcting LSTM; $\delta = 2$</i>	88.47	0.64376
<i>Error-correcting LSTM; $\delta = 3$</i>	88.57	0.64273
<i>Lower bound</i>	89.70	0.63912

Table 4.2: The performance of my error-correcting language model with respect to the upper and lower bound, using the FCE dataset.

The performance of my error-correcting language model that was outlined in Section 3.5 is given in Table 4.2. It can be seen that it improves predictions on text containing errors that do not require the insertion or deletion of words. The best result arises from an edit

distance threshold of 1, which seems attributed to the fact that a larger threshold allows for more false corrections, as shown in Figure 4.7b.

Input	i am more than happy to give you the nessessary information.
Target	i am more than happy to give you the necessary information.
Prediction	i am more than happy to give you the necessary information.

(a) An accurate correction.

Input	the bus will pick you up right at your hotel entery .
Target	the bus will pick you up right at your hotel entrance .
Prediction	the bus will pick you up right at your hotel every .

(b) A false correction.

Input	appart of that, there is no recommendation as to what to wear.
Target	apart from that, there is no recommendation as to what to wear.
Prediction	apart of that, there is no recommendation as to what to wear.

(c) A missed correction.

Figure 4.7: Example corrections made by my error-correcting RNN with $\delta = 2$.

There is still a gap to be closed, which is demonstrated in Figure 4.7c by the fact that my model does not correct non-spelling mistakes.

Chapter 5

Conclusions

The project was a success: I implemented a variety of n -gram and RNN-based language models, and benchmarked them in terms of both their prediction performance and their applicability to running on a mobile platform, as set out in my project proposal. Additionally, I built a mobile keyboard for iOS that uses my language models as a library, and proposed a novel extension to an existing language model that improves its performance on error-prone text.

My implementation of existing language models saw a level of performance that was comparable with measures made by existing authors. For example, my modified Kneser-Ney smoothing n -gram model, vanilla RNN model and LSTM model all matched the results from Mikolov on the Penn Tree Bank dataset [25, 26]. This allowed me to make comprehensive and reliable comparisons between each of my models, and assess the trade-offs between accuracy and resource consumption. My work also saw such models complement one another, in that the interpolation of their probability distributions surpassed the performance of all of my stand-alone models, achieving state-of-the-art prediction accuracy.

I explicitly demonstrated that language model performance is compromised by error-prone text. Whilst I made some modifications to an existing model that saw an increase in performance on such text, there is still room for improvement. If I were given more time to work on the project, I would investigate ways to cater for non-spelling corrections. From the CoNLL shared tasks of 2013 [27] and 2014 [28], it is clear that there is a plethora of methods that already exist for grammatical error-correction. It would be interesting to see which of these could be applied to improve the performance of language models on error-prone text.

Throughout this dissertation, I have learnt a great deal about language modelling, but what have my language models learnt from me? Here is a one line summary of my work, in the words of an LSTM-based language model, trained on the significantly small training set that is the text of this L^AT_EX file:

‘the number of words that the language model is a mobile of the network, and the input is the input set.’

Bibliography

- [1] W. E. Johnson, “Probability: The deductive and inductive problems,” *Mind*, vol. 41, no. 164, pp. 409–423, 1932.
- [2] H. Ney, U. Essen, and R. Kneser, “On structuring probabilistic dependences in stochastic language modelling,” *Computer Speech & Language*, vol. 8, no. 1, pp. 1–38, 1994.
- [3] R. Kneser and H. Ney, “Improved backing-off for m-gram language modeling,” in *Acoustics, Speech, and Signal Processing, 1995. ICASSP-95., 1995 International Conference on*, vol. 1, pp. 181–184, IEEE, 1995.
- [4] S. F. Chen and J. Goodman, “An empirical study of smoothing techniques for language modeling,” *Comput. Speech Lang.*, vol. 13, pp. 359–394, Oct. 1999.
- [5] S. Katz, “Estimation of probabilities from sparse data for the language model component of a speech recognizer,” *IEEE transactions on acoustics, speech, and signal processing*, vol. 35, no. 3, pp. 400–401, 1987.
- [6] I. J. Good, “The population frequencies of species and the estimation of population parameters,” *Biometrika*, pp. 237–264, 1953.
- [7] F. A. Azevedo, L. R. Carvalho, L. T. Grinberg, J. M. Farfel, R. E. Ferretti, R. E. Leite, R. Lent, S. Herculano-Houzel, *et al.*, “Equal numbers of neuronal and nonneuronal cells make the human brain an isometrically scaled-up primate brain,” *Journal of Comparative Neurology*, vol. 513, no. 5, pp. 532–541, 2009.
- [8] W. S. McCulloch and W. Pitts, “A logical calculus of the ideas immanent in nervous activity,” *The bulletin of mathematical biophysics*, vol. 5, no. 4, pp. 115–133, 1943.
- [9] D. E. Rumelhart, G. E. Hinton, and R. J. Williams, “Learning internal representations by error propagation,” tech. rep., DTIC Document, 1985.
- [10] G. E. Hinton, S. Osindero, and Y.-W. Teh, “A fast learning algorithm for deep belief nets,” *Neural computation*, vol. 18, no. 7, pp. 1527–1554, 2006.
- [11] Y. Bengio, Y. LeCun, *et al.*, “Scaling learning algorithms towards AI,” *Large-scale kernel machines*, vol. 34, no. 5, pp. 1–41, 2007.
- [12] K. Hornik, M. Stinchcombe, and H. White, “Multilayer feedforward networks are universal approximators,” *Neural networks*, vol. 2, no. 5, pp. 359–366, 1989.
- [13] S. Hochreiter and J. Schmidhuber, “Long short-term memory,” *Neural computation*, vol. 9, no. 8, pp. 1735–1780, 1997.

- [14] K. Cho, B. Van Merriënboer, C. Gulcehre, D. Bahdanau, F. Bougares, H. Schwenk, and Y. Bengio, “Learning phrase representations using RNN encoder-decoder for statistical machine translation,” *arXiv preprint arXiv:1406.1078*, 2014.
- [15] M. Abadi, P. Barham, J. Chen, Z. Chen, A. Davis, J. Dean, M. Devin, S. Ghemawat, G. Irving, M. Isard, *et al.*, “TensorFlow: A system for large-scale machine learning,” in *Proceedings of the 12th USENIX Symposium on Operating Systems Design and Implementation (OSDI)*. Savannah, Georgia, USA, 2016.
- [16] M. Hermans and B. Schrauwen, “Training and analysing deep recurrent neural networks,” in *Advances in neural information processing systems*, pp. 190–198, 2013.
- [17] S. Hochreiter, *Untersuchungen zu dynamischen neuronalen Netzen*. PhD thesis, TU Munich, 1991.
- [18] Y. Bengio, P. Simard, and P. Frasconi, “Learning long-term dependencies with gradient descent is difficult,” *IEEE transactions on neural networks*, vol. 5, no. 2, pp. 157–166, 1994.
- [19] F. A. Gers and J. Schmidhuber, “Recurrent nets that time and count,” in *Neural Networks, 2000. IJCNN 2000, Proceedings of the IEEE-INNS-ENNS International Joint Conference on*, vol. 3, pp. 189–194, IEEE, 2000.
- [20] N. Srivastava, *Improving neural networks with dropout*. PhD thesis, University of Toronto, 2013.
- [21] W. Zaremba, I. Sutskever, and O. Vinyals, “Recurrent neural network regularization,” *arXiv preprint arXiv:1409.2329*, 2014.
- [22] M. P. Marcus, M. A. Marcinkiewicz, and B. Santorini, “Building a large annotated corpus of English: The Penn Treebank,” *Computational linguistics*, vol. 19, no. 2, pp. 313–330, 1993.
- [23] C. Chelba, T. Mikolov, M. Schuster, Q. Ge, T. Brants, P. Koehn, and T. Robinson, “One billion word benchmark for measuring progress in statistical language modeling,” *arXiv preprint arXiv:1312.3005*, 2013.
- [24] H. Yannakoudakis, T. Briscoe, and B. Medlock, “A new dataset and method for automatically grading ESOL texts,” in *Proceedings of the 49th Annual Meeting of the Association for Computational Linguistics: Human Language Technologies-Volume 1*, pp. 180–189, Association for Computational Linguistics, 2011.
- [25] T. Mikolov, “Statistical language models based on neural networks,” *Presentation at Google, Mountain View, 2nd April*, 2012.
- [26] T. Mikolov, A. Joulin, S. Chopra, M. Mathieu, and M. Ranzato, “Learning longer memory in recurrent neural networks,” *arXiv preprint arXiv:1412.7753*, 2014.
- [27] H. T. Ng, S. M. Wu, Y. Wu, C. Hadiwinoto, and J. Tetreault, “The CoNLL-2013 shared task on grammatical error correction,” in *CoNLL Shared Task*, pp. 1–12, 2013.
- [28] H. T. Ng, S. M. Wu, T. Briscoe, C. Hadiwinoto, R. H. Susanto, and C. Bryant, “The CoNLL-2014 shared task on grammatical error correction,” in *CoNLL Shared Task*, pp. 1–14, 2014.

Appendix A

Backpropagation Recurrence Relation

To see how the recurrence relation for δ_i^l is derived, consider the neuron i in the first hidden layer of the neural network shown in Figure A.1. Without loss of generality, suppose this layer is layer l .

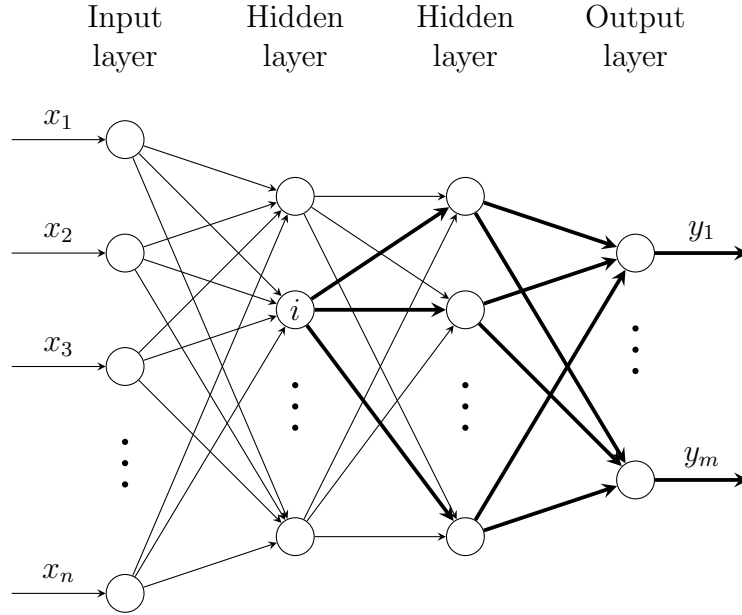


Figure A.1: An illustration of the connections along which the partial derivatives are chained for backpropagating to a particular node i .

From the diagram, it is clear that the derivative $\frac{\partial \mathcal{L}}{\partial z_i^l}$ depends on the derivatives of all of the neurons in layer $l + 1$, and so from the chain rule it follows that:

$$\frac{\partial \mathcal{L}}{\partial z_i^l} = \sum_j \frac{\partial \mathcal{L}}{\partial z_j^{l+1}} \frac{\partial z_j^{l+1}}{\partial z_i^l}$$

Plugging in the definition of δ_i^l , and simplifying gives:

$$\delta_i^l = \sum_j \delta_j^{l+1} \frac{\partial z_j^{l+1}}{\partial z_i^l} = \sum_j \delta_j^{l+1} \frac{\partial z_j^{l+1}}{\partial y_i^l} \frac{\partial y_i^l}{\partial z_i^l} = \phi'(z_i^l) \sum_j \delta_j^{l+1} w_{i,j}^l$$

as required.

These ideas can be extended to a recurrent neural network, whereby the derivative $\frac{\partial \mathcal{L}}{\partial z_i^l}$ depends not only on the derivatives of the neurons in layer $l + 1$, but also on the neurons in layer l at time step $t + 1$:

$$\frac{\partial \mathcal{L}}{\partial z_i^{l,t}} = \sum_j \frac{\partial \mathcal{L}}{\partial z_j^{l+1,t}} \frac{\partial z_j^{l+1,t}}{\partial z_i^{l,t}} + \sum_j \frac{\partial \mathcal{L}}{\partial z_j^{l,t+1}} \frac{\partial z_j^{l,t+1}}{\partial z_i^{l,t}}$$

Again, plugging in the definition of $\delta_i^{l,t}$ and simplifying gives:

$$\delta_i^{l,t} = \sum_j \delta_j^{l+1,t} \frac{\partial z_j^{l+1,t}}{\partial z_i^{l,t}} + \sum_j \delta_j^{l,t+1} \frac{\partial z_j^{l,t+1}}{\partial z_i^{l,t}} = \phi'(z_i^{l,t}) \left(\sum_j \delta_j^{l+1,t} w_{i,j}^l + \sum_j \delta_j^{l,t+1} v_{i,j}^l \right)$$

as required.

Appendix B

Perplexity

The formula for perplexity, presented in Equation 4.1 and repeated below for convenience, can be better understood from a touch of information theory.

$$\text{PP}_L(w_1^N) = \sqrt[N]{\frac{1}{\mathbb{P}_L(w_1^N)}} = \sqrt[N]{\prod_{i=1}^N \frac{1}{\mathbb{P}_L(w_i|w_1^{i-1})}}$$

In information theory, the cross-entropy $H(p, q)$ between a true probability distribution p and an estimate of that distribution q is defined as:¹

$$H(p, q) = - \sum_x p(x) \log_2 q(x)$$

It can be shown that $H(p, q) = H(p) + D_{KL}(p||q)$ where $D_{KL}(p||q) \geq 0$ is the Kullback-Leibler distance between p and q . Generally speaking, the better an estimate q is of p , the lower $H(p, q)$ will be, with a lower bound of $H(p)$, the entropy of p .

The perplexity PP of a model q , with respect to the true distribution p it is attempting to estimate, is defined as:

$$\text{PP} = 2^{H(p, q)}$$

Language models assign probability distributions over sequences of words, and so it seems reasonable to use perplexity as a measure of how accurately they model the real underlying distribution. Unfortunately, in language modelling, the underlying distribution, p , is not known, so it is approximated with Monte Carlo estimation by taking samples of p (i.e. sequences of words from the test data) as follows:

$$\text{PP}_L(w_1^N) = 2^{-\frac{1}{N} \sum_i \log_2 \mathbb{P}_L(w_i|w_1^{i-1})}$$

With a little algebra, this can be rearranged to give Equation 4.1.

¹Note that $H(p, q)$ is often also used to denote the joint entropy of p and q , which is a different concept.

Appendix C

Project Proposal

COMPUTER SCIENCE TRIPOS - PART II PROJECT PROPOSAL

Language Modelling for Text Prediction

October 20, 2016

supervised by
Dr. Marek Rei & Dr. Ekaterina Shutova

Introduction

Language models (LMs) produce a probability distribution over a sequence of words, which can be used to estimate the relative likelihood of words or phrases occurring in various contexts. The predictive power of such models is useful in speech recognition, text prediction, machine translation, handwriting recognition, part-of-speech tagging and information retrieval. For instance, in a text prediction context, if a user has typed:

‘Do you want to grab a ’

then a language model could be used to suggest probable next words such as ‘coffee’, ‘drink’ or ‘bite’, and these predictions could further be narrowed down when the user starts typing the next word.

With the recent shift from desktop to mobile computing, it has become increasingly important to facilitate accurate, high-speed language models that aren’t too memory hungry. The aim of this project is to investigate this area by implementing and benchmarking various language models, with an aim of providing useful insights into how their the accuracy, speed and resource consumption compare.

I will begin with the traditional n -gram language models and their various smoothing techniques, before proceeding onto neural network-based language models. Combinations of such models, which can be achieved by interpolating their probability distributions, will also be included in the benchmark.

In the context of text prediction, one issue with existing language models is that they assume that their input is error (such as spelling or grammatical errors) free. With the presence of typos, human error and people learning new languages, this assumption is often not so reliable. I aim to include some extensions to existing language models that seek to provide better predictions over error-prone text, and I will assess whether they offer any performance improvements by including them in the benchmark.

Starting point

Code

TensorFlow, an open-source machine learning library, will assist in the construction of the neural network-based language models.

Computer Science Tripos

The following Tripos courses will be useful when undertaking my project:

- **Natural Language Processing** - n -gram language models.
- **Artificial Intelligence I** - neural networks.
- **Machine Learning and Bayesian Inference** - tips and tricks around supervised learning.

- **Mathematical Methods for Computer Science** - Markov chains.
- **Information Theory** - entropy.

Algorithms, Object-Oriented Programming, Software and Interface Design, and **Software Engineering** will also prove useful for general programming and software engineering.

There are aspects of language modelling not covered by the Tripos, including various smoothing techniques for n -gram LMs and recurrent neural networks. I will fill this gap with personal reading.

Experience

Language modelling draws ideas from natural language processing and machine learning. I have no prior experience in the former field, but in the latter I have an understanding of neural networks (including RNNs) that comes from both the Artificial Intelligence I course and personal reading.

Resources required

Machines

The development of my project will be carried out on my personal laptop, a 2015 MacBook Pro running macOS Sierra. I accept full responsibility for this machine and I have made contingency plans to protect myself against hardware and/or software failure.

I intend to train the language models on my laptop. When experimenting with the neural network-based language models on larger datasets, I may utilise the University's High Performance Computing Service¹ in order to run the models on a GPU cluster.

Datasets

In order to train and evaluate the various language models, I will need to make use of various datasets, including, but not limited to:

- The Penn Treebank (PTB) dataset.²
- The One Billion Word dataset.³
- The CLC FCE dataset.⁴

¹<https://www.cl.cam.ac.uk/local/sys/resources/hpc/>

²<http://www.cis.upenn.edu/~treebank/>

³<http://www.statmt.org/lm-benchmark/>

⁴<http://illexir.co.uk/applications/clc-fce-dataset/>

Version Control & Backup

I will use Git for version control, with one Git repository for all project documentation and another for all project source code. Both repositories will be hosted on GitHub, and all local commits will be pushed no more than a few hours after they are created, meaning that I should never lose more than half a day's worth of work.

I will also make weekly backups on an external hard drive, so that in the unlikely event that both GitHub goes down and my laptop fails, I will be able to recover my project on a new machine.

Work to be done

Language models

The first stage of the project is to implement a variety of language modelling algorithms. Starting with the simplest, I will implement n -gram models with varying values of n . I will also investigate the effect of add-1 smoothing [1], Katz smoothing [2], absolute discounting [3], Kneser-Ney smoothing [4] and modified Kneser-Ney smoothing [5].

Once the n -gram models along with their corresponding smoothing techniques have been implemented, I will implement some neural network-based language models, which have recently proven very effective. I will start with an RNN (recurrent neural network) based language model [6], and then I will advance to the more complex LSTM (long short-term memory) architecture [7].

Combinations of the neural network and n -gram based language models, and extensions to the aforementioned models that aim to improve performance over error-prone text, will also be implemented.

In order to implement the neural network-based language models, I will make use of TensorFlow, Google's open-source machine learning library. TensorFlow has both Python and C++ APIs, but the Python API is more complete and easier to use⁵, and so I will construct my language models in Python.

Benchmarking framework

Once a series of language models have been implemented, the next stage would be to implement a framework into which each language model can be inserted. The framework should run a series of tests and generate evaluation for:

- **Accuracy:** Perplexity, cross-entropy and guessing entropy.
- **Speed and resource usage:** Time and memory usage at training and inference.

Given that I will be evaluating the language models in the context of text prediction, I will also include some metrics for how useful the predictions are for users, such as the number of letters the user could avoid typing as the result of a correct prediction.

⁵https://www.tensorflow.org/versions/r0.11/api_docs/cc/index.html

Demonstration application

In order to demonstrate the language models in a practical context, I aim to integrate them into a simple console application that predicts the next word in the user's sentence.

Success criteria

This project will be deemed successful if the following criteria are met:

- Language models (LMs) using the following techniques are implemented:
 - N -gram LMs with various smoothing techniques.
 - A vanilla RNN-based LM.
 - An LSTM-based LM.
- Comprehensible and reliable comparisons between the various LM implementations and their combinations are made regarding their accuracy, speed and resource consumption during both training and inference.
- A simple console application is developed to demonstrate the capability of the aforementioned language models in the context of next-word prediction.

Possible extensions

Mobile Keyboard

One possible extension would be to integrate the language model implementations into a system-wide mobile keyboard on iOS - made possible as of iOS 8.⁶ This would involve constructing a simple keyboard interface which displays suggestions for the next word in the user's sentence, by querying the language model(s) and filtering the results as the next word is typed. TensorFlow supports exportation of models into a language-independent format (using Google's protocol buffers), which means I could train and export my language models in Python and then load them into iOS using TensorFlow's C++ API.

⁶<https://developer.apple.com/library/content/documentation/General/Conceptual/ExtensibilityPG/CustomKeyboard.html>

Timetable

Michaelmas term

06/10 - 19/10	<ul style="list-style-type: none">• Refine project idea and write project proposal.• Background reading on n-gram language models and smoothing techniques.
Milestone	<i>Project proposal finished.</i>
20/10 - 26/10	<ul style="list-style-type: none">• Implement a simple bigram language model, without any smoothing.• Background reading on recurrent neural networks and long short-term memory in particular.
Milestone	<i>A working implementation of a basic bigram language model completed.</i>
27/10 - 09/11	<ul style="list-style-type: none">• Implement a test framework for generating evaluation metrics for a given language model.
Milestone	<i>Test framework complete.</i>
10/11 - 16/11	<ul style="list-style-type: none">• Implement the various smoothing techniques outlined in ‘Work to be done’.
Milestone	<i>A working implementation of n-gram language models with various smoothing techniques completed.</i>
17/11 - 23/11	<ul style="list-style-type: none">• Implement vanilla RNN language model.
Milestone	<i>A working implementation of RNN-based language model.</i>
24/11 - 30/11	<ul style="list-style-type: none">• Implement LSTM-based language model.
Milestone	<i>A working implementation of LSTM-based language model.</i>

Christmas vacation

01/12 - 14/12	<ul style="list-style-type: none">• Experiment with extending the existing language models to improve performance on error-prone text.
05/01 - 18/01	<ul style="list-style-type: none">• Benchmark and evaluate the various language models.• Start implementing an application to demonstrate the capability of the language models in the context of next-word prediction.
Milestone	<i>Language models benchmarked.</i>

Lent term

19/01 - 01/02	<ul style="list-style-type: none">• Write progress report.• Complete demonstration application.
Milestone	<i>Progress report and demonstration application complete.</i>
02/02 - 22/02	<ul style="list-style-type: none">• Rigorous testing and bug-fixing of project.• If time allows, implement project extensions.• Present project to overseers.
Milestone	<i>Project presentation given and implementation complete.</i>
23/02 - 15/03	<ul style="list-style-type: none">• Start writing dissertation.

Easter vacation

16/03 - 19/04	<ul style="list-style-type: none">• Finish writing dissertation and proof-read it.
Milestone	<i>Final dissertation draft submitted to supervisor and Director of Studies.</i>

Submission deadline on 19/05/2017.

References

- [1] W.E. Johnson. Probability: deductive and inductive problems *Mind*, Vol. 41, No. 164, pp. 409-423, 1932.
- [2] S.M. Katz. Estimation of probabilities from sparse data for the language model component of a speech recognizer. *IEEE Transactions on Acoustics, Speech and Signal Processing*, Vol. 35, No. 3, pp. 400-401, 1987.
- [3] H. Ney, U. Essen and R. Kneser. On structuring probabilistic dependences in stochastic language modeling. *Computer, Speech, and Language*, Vol. 8, No. 1, pp. 1-38, 1994.
- [4] R. Kneser and H. Ney. Improved backing-off for m-gram language modeling. In *Proceedings of the IEEE International Conference on Acoustics, Speech and Signal Processing*, Vol. 1, pp. 181-184, 1995.
- [5] S.F. Chen and J. Goodman. An empirical study of smoothing techniques for language modeling. *Computer Speech & Language*, Vol. 13, No. 4, pp. 359-394, 1999.
- [6] T. Mikolov, M. Karafiát, L. Burget, J. Cernocký, and S. Khudanpur. Recurrent neural network based language model. In *Interspeech*, Vol. 2, pp. 3, 2010.
- [7] S. Hochreiter and J. Schmidhuber. Long short-term memory. *Neural Computation*, Vol. 9, No. 8, pp. 1735-1780, 1997.