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[Design](#), [Architecture](#)



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0px Helvetica}span. s1 {font: 24. 0px Helvetica}span.

s2 {color: #002486}Overview of Three Different Structures of Artificial Neural

Networks for Speech Recognitions1736880Abstract Automatic speech

recognition (ASR) is the translation, through some methodologies, of

human speech into text by machines and plays an important role nowadays.

In this research review we examine three different artificial neural

network architectures that are used in speech recognition field and we

investigate their performance in different cases. We analyze the state-of-

the-art deep neural networks (DNNs), that have evolved into complex structures

and they achieve significant results in a variety of speech benchmarks.

Afterward, we explain convolutional neural networks (CNNs) and we explore

their potential in this field. Finally, we present the recent research in highway

deep neural networks (HDNNs) that seem to be more flexible for resource

constrained platforms. Overall, we critically try to compare these methods and

show their strengths and limitations. Each method has its benefits and

applications and from them we try to draw some conclusions and give some potential future directions.

I. Introduction Machine Learning (ML) is a field of computer science that gives the computers the ability to learn through different algorithms and techniques without being programmed. ASR is closely related with ML because it uses methodologies and procedures of ML 1 , 2 , 3 .

ASR has been around for decades but it was not until recently that there was a tremendous development because of the advances in both machine learning methods and computer hardware. New ML techniques made speech recognition accurate enough to be useful outside of carefully controlled environments, so it could easily be deployed in many electronic devices nowadays (i. e. computers, smart-phones) and be used in many applications such as identifying and authenticating a user via of his/her voice. Speech is the most important mode of communication between human beings and that is why from the early part of the previous century, efforts have been made in order to make machines do what only humans could perceive.

Research has been conducted through the past five decades and the main reason was the desire of making tasks automated using machines 2 . Many motivations using different theories such as probabilistic modeling and reasoning, pattern recognition and artificial neural networks affected the researchers and helped to advance ASR. The first single advance in the history of ASR occurred in the middle of 70's with the introduction of the expectation-maximization (EM) 4 algorithm for training hidden Markov

models (HMMs). The EM technique gave the possibility to develop the first speech recognition systems using Gaussian mixture models (GMMs). Despite all the advantages of the GMMs, they are not able to model efficiently data that lie on or near a nonlinear surface in the data space (i. e. sphere).

This problem could be solved by artificial neural networks because they can capture these non-linearities in the data but the computer hardware of that era did not allow us to build complex neural networks. As a result, in the beginning most speech recognition systems were based on HMMs. Later the neural network and hidden Markov model (NN/ HMM) hybrid architecture ⁵ was used for ASR systems. After 2000s and over the last years the improvement of computer hardware and the invention of new machine learning algorithms made possible the training for DNNs. DNNs with many hidden layers have been shown to achieve comparable and sometimes much better performance than GMMs in many different databases (with speech data) and in a range of applications ⁶ .

After the huge success of DNNs, researchers try other artificial neural architectures such as recurrent neural networks with long short-term memory units (LSTM-RNNs) ⁷ , deep belief networks and CNNs, and it seems that each one of them has its benefits and weaknesses. In this literature review we present three types of artificial neural networks (DNNs, CNNs, and HDNNs). We analyze each method, we explain how they are used for training and what are their advantages and disadvantages. Finally we compare these methods in the context of ASR, identifying where each one of them is more suitable

and what are their limitations. Finally, we draw some conclusions from these comparisons and we carefully suggest some probable future directions.

II. Methods
A. Deep Neural Networks
DNNs are feed-forward artificial neural networks with more than one layer of hidden units. Each hidden layer has a number of units (or neurons) each of which Informatics Research Review (s1736880) takes all outputs of the lower layer as input and passes them through a linearity.

After that we apply a non linear activation function (i. e. sigmoid function, hyperbolic tangent function, some kind of rectified linear unit function (ReLU8, 9), or exponential linear unit function (ELU 10)) for the final transformation of our initial inputs. Sometimes, for a multi-class classification problem, the posterior probability of each class can be estimated using an output softmax layer. For the training process of DNNs we usually use the back propagation technique 11. For large training sets, it is typically more convenient to compute derivatives on a mini-batch of the training set rather than the whole training set (this is called stochastic gradient descent). As cost function we often use the cross-entropy (CE) in order to have a comparison meter between the output of the network and the actual output but the choice of the cost function actually depends on the case.

The difficulty to optimize DNNs with many hidden layers along with overfitting problem force us to use pretraining methods. One such a popular method is to use the restricted Boltzmann machines (RBMs) 12. If we use a stack of RBMs then we can construct a deep belief network (DBN) (you should not be confused with dynamic Bayesian network). The purpose of this is to add an

initial stage of generative pretraining. The pretraining is very important for DNNs because it reduces overfitting and it also reduces the time required for discriminative fine-tuning with propagation. DNNs in the context of ASR play a major role. Many architectures have been used by different research groups in order to gain better and better accuracy in acoustic models.

You can see some methodologies in the article 6 that it presents some significant results and shows that DNNs in general achieve higher speech recognition accuracy than GMMs on a variety of speech recognition benchmarks such as TIMIT and some other large vocabulary environments. The main reason is that they take advantage from the fact that they can handle the non-linearities in the data and so they can learn much better models comparing to GMMs. However, we have to mention that they use many model parameters in order to achieve a good enough speech accuracy and this is sometimes a drawback.

Furthermore, they are complex enough and need many computational resources. Finally, they have been criticized because they do not preserve some specific structure (we can use different structures until we achieve a significant speech accuracy), they are difficult to be interpreted (because they have not some specific structure) and they possess limited adaptability (we use different approaches for different cases). Besides all of these disadvantages they remain the state-of-the-art for speech recognition the last few years and they have given us the most reliable and consistent results overall. B.

Convolutional Neural Networks Convolutional neural networks (CNNs) can be regarded as DNNs with the main difference that instead of using fully connected hidden layers (as it happens in DNNs; full connection with all the possible combinations among the hidden layers) they use a special network structure, which consists of convolution and pooling layers 13 , 14 , 15 .

Basic rule is that the data have to be organized as a number of feature maps in order to be passed in each convolutional layer. One significant problem we have when we want to transform our speech data in feature maps concerns frequency because we are not able to use the conventional mel-frequency cepstral coefficient (MFCC) technique 16 .

The reason is that this technique does not preserve the locality of our data (in the case of CNNs), although we want to preserve locality in both frequency and time. Hence, a solution is the use of mel-frequency spectral coefficients (MFSC features) 15 . Our purpose with MFSC technique is to form the input feature maps without losing the property of locality in our data. Then we can apply the convolution and pooling layers with their respective operations to generate the activations of the units in those layers. We should mention that each input feature map is connected to many feature maps and the feature maps share the weights. Thus, firstly, we use the convolution operation to construct our convolutional layers and afterwards, we apply the pooling layer in order to reduce the resolution of the feature maps.

This process continues depending on how deep we want to be our network (maybe we could achieve higher speech accuracy with more layers on this structure or maybe not). You can see the whole process and the usage

of convolution and pooling layers in the paper 15 . Moreover, as it happens for DNNs with RBMs, there is a respective procedure CRBM 17 for CNNs that allow us pretraining our data in order to gain in speech accuracy and reduce the overfitting effect.

In the paper 15 , the authors also examine the case of a CNN with limited weight sharing for ASR (LWS model) and they propose to pretrain it modifying the CRBM model. CNNs have three major properties: locality, weight sharing, and pooling. Each one of them has the potential to improve speech recognition performance. These properties can reduce the overfitting problem and they can add robustness against non-white noise. In addition, they can reduce the number of network weights to be learned.

Both locality and weight sharing are significant factors for the property of pooling which is very helpful in handling small frequency shifts that are common in speech signals. These shifts may occur from differences in vocal tract lengths among different speakers 15 . In general, CNNs seem to have a relative better performance in ASR taking advantage from their special network structure. C.

Highway Deep Neural Networks HDNNs are depth-gated feed-forward neural networks 18 . They are distinguished from the conventional DNNs for two main reasons. Firstly they use much less model parameters and secondly they use two types of gate functions to facilitate the information flow through the hidden layers. Informatics Research Review (s1736880) HDNNs are multi-layer networks with many hidden layers.

In each layer we have the transformation of the initial input or of the previous hidden layer with the corresponding parameter of the current layer (they are combined in a linear way) followed by a non-linear activation function (i. e. sigmoid function). The output layer is parameterized with the parameter and we usually use the softmax function as the output function in order to obtain the posterior probability of each class given our initial inputs. Afterwards, given the target outputs, the network is usually trained by gradient descent to minimize a loss function such as cross entropy (CE function). So, we can see that the architecture and the process are the same as in DNNs that we described in subsection of DNNs. The difference from the standard DNNs is that highway deep neural networks (HDNNs) were proposed to enable every deep networks to be trained by augmenting the hidden layers with gate functions 19.

This augmentation happens through the transform and carry gate functions. The first scales the original hidden activations and the latter scales the input before passing it directly to the next hidden layer 18. Three main methods are presented for training, thesequence training, the adaptation technique and the teacher student training in the papers 18, 20, 21. Combining these methodologies with the two gates it is demonstrated how important role the carry and the transform gate play in the training. The main reason is that the gates are responsible to control the flow of the information among the hidden layers.

They allow us to achieve comparable speech recognition accuracy to the classic DNNs but with much less model parameters because we have the

ability to handle the whole network through the parameters of the gate functions (which are much less comparing to the parameters of the whole network). This outcome is crucial for platforms such as mobile devices (i. e. voice recognition on mobiles) due to the fact that we have not many disposal resources in these devices. D. Comparison of the Methods These methods, that we presented, have their benefits and limitations. In general, DNNs behave very well and in many cases they have enough better performance compared to GMMs on a range of applications. The main reason is that they take advantage from the fact that they can handle much better the non-linearities in the data space.

On the other hand, their biggest drawback compared with GMMs is that it is much harder to make good use of large cluster machines to train them on massive data [6]. As far as the CNNs are concerned, they can handle frequency shifts which are difficult to be handled within other models such as GMMs and DNNs. Furthermore, it is also difficult to learn such an operation as max-pooling in standard artificial neural networks. Moreover, CNNs can handle the temporal variability in the speech features as well [15]. On the other hand, the fine-tuning of the pooling size (careful selection of pooling size) is very important because otherwise we may cause phonetic confusion, especially at segment boundaries.

Despite the fact that CNNs seem to have better accuracy than DNNs with less parameters, computationally are more expensive because of the complexity of the convolution operation. HDNNs are considered to be more compact than regular DNNs due to the fact that they can achieve similar recognition

accuracy with many fewer model parameters. Furthermore, they are more controllable than DNNs and this is because through the gate functions we can control the behavior of the whole network using a very small number of model parameters (the parameters of the gates). Moreover, HDNNs are more controllable because the authors in paper 18 show that simply updating the gate functions using adaptation data they can gain considerably in speech recognition accuracy. We cannot conclude much for their general performance because they are a recent proposal and it is needed more research to see their overall benefits and limitations.

However, the main idea is to use them in order to have comparable ASR accuracy with DNNs and simultaneously to reduce the model parameters. III. Conclusions Overall, we can say that DNNs are the state-of-the-art today because they behave very well on a range of speech recognition benchmarks. However, other architectures of artificial neural networks such as CNNs have achieved comparable performance in the context of ASR. Besides that, research continues to be conducted in this field in order to find new methods, learning techniques and architectures that will allow us to train our data sets more efficiently. This means less parameters, less computational power, less complex models, more structured models. Ideally we would like to have a whole general model that covers a lot of cases and not many different models that applied in different circumstances.

On the other hand this is probably difficult, so just distinct methodologies and techniques for different cases may be our temporary or unique solution. In this direction, HDNNs or other methods may be used to deal with specific cases.

Many future directions have been suggested the last few years for research in order to advance ASR. Some probable suggestions are to use unsupervised learning or reinforcement learning for acoustic models. Another potential direction is to search for new architectures or special structures in artificial neural networks or inventing new learning techniques and at the same time improving our current algorithms.