

LEARNING LINEARLY SEPARABLE FEATURES FOR SPEECH RECOGNITION USING CONVOLUTIONAL NEURAL NETWORKS

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ABSTRACT

Automatic speech recognition systems usually rely on spectral-based features, such as MFCC or PLP. These features are extracted based on prior knowledge such as, speech perception or/and speech production. Recently, convolutional neural networks have been shown to be able to estimate phoneme conditional probabilities in a completely data-driven manner, i.e. using directly temporal raw speech signal as input. This system was shown to yield similar or better performance than HMM/ANN based system on phoneme recognition task and on large scale continuous speech recognition task, using less parameters. Motivated by these studies, we investigate the use of simple linear classifier in the CNN-based framework. Thus, the network learns linearly separable features from raw speech. We show that such system yields similar or better performance than MLP based system using cepstral-based features as input.

1 INTRODUCTION

State-of-the-art automatic speech recognition (ASR) systems typically divide the task into several sub-tasks, which are optimized in an independent manner (Bourlard & Morgan, 1994). In a first step, the data is transformed into features, usually composed of a dimensionality reduction phase and an information selection phase, based on the task-specific knowledge of the phenomena. These two phases have been carefully hand-crafted, leading to state-of-the-art features such as mel frequency cepstral coefficients (MFCCs) or perceptual linear prediction cepstral features (PLPs). In a second step, the likelihood of subword units such as, phonemes is estimated using generative models or discriminative models. In a final step, dynamic programming techniques are used to recognize the word sequence given the lexical and syntactical constraints.

Recently, in the hybrid HMM/ANN framework (Bourlard & Morgan, 1994), there has been growing interests in using “intermediate” representations, like short-term spectrum, instead of conventional features, such as cepstral-based features. Representations such as Mel filterbank output or log spectrum have been proposed in the context of deep neural networks (Hinton et al., 2012). In our recent study (Palaz et al., 2013), it was shown that it is possible to estimate phoneme class conditional probabilities by using temporal raw speech signal as input to convolutional neural networks (LeCun, 1989) (CNNs). This system yielded similar or better results on TIMIT phoneme recognition task with standard hybrid HMM/ANN systems. We also showed that this system is scalable to large vocabulary speech recognition task (Palaz et al., 2015). In this case, the CNN-based system was able to outperform the HMM/ANN system with less parameters.

In this paper, we investigate the features learning capability of the CNN based system with simple classifiers. More specifically, we replace the classification stage of the CNN based system, which was a non-linear multi-layer perceptron, by a linear single layer perceptron. Thus, the features

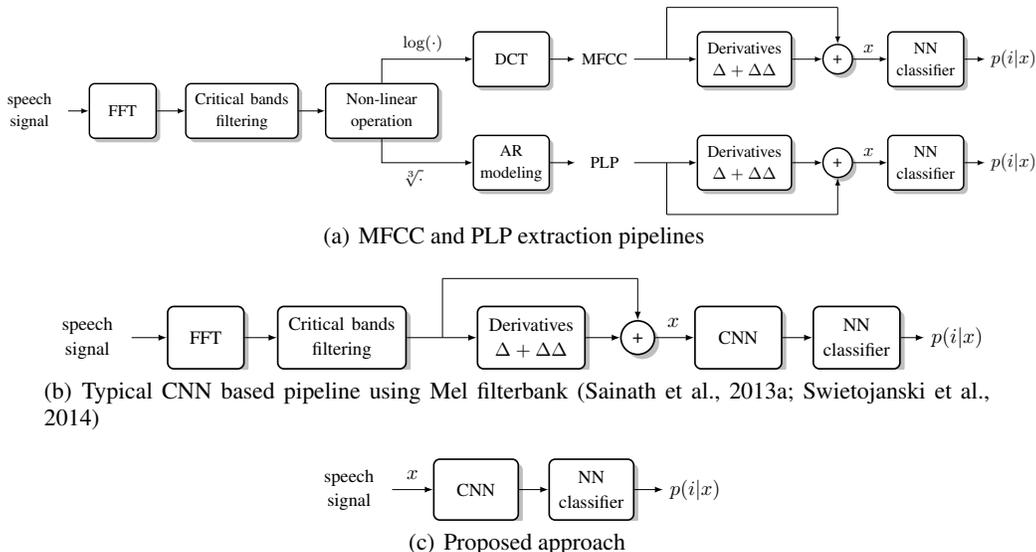


Figure 1: Illustration of several features extraction pipelines. $p(i|x)$ denotes the conditional probabilities for each input frame x , for each label i .

learned by the CNNs are trained to be linearly separable. We evaluate the proposed approach on phoneme recognition task on the TIMIT corpus and on large vocabulary continuous speech recognition on the WSJ corpus. We compare our approach with conventional HMM/ANN system using cepstral-based features. Our studies show that the CNN-based system using a linear classifier yields similar or better performance than the ANN-based approach using MFCC features, with fewer parameters.

The remainder of the paper is organized as follows. Section 2 presents the motivation of this work. Section 3 presents the architecture of the proposed system. Section 4 presents the experimental setup and Section 5 presents the results. Section 6 presents the discussion and conclude the paper.

2 MOTIVATION

In speech recognition, designing relevant features is not a trivial task, mainly due to the fact that the speech signal is non-stationary and that relevant information is present at different level, namely spectral level and temporal level. Inspired by speech coding studies, feature extraction typically involves modeling the envelop of the short-term spectrum. The two most common features along that line are Mel frequency cepstral coefficient (MFCC) (Davis & Mermelstein, 1980) and perceptual linear prediction cepstral coefficient (PLP) (Hermansky, 1990). These features are both based on obtaining a good representation of the short-term power spectrum. They are computed following a series of steps, as presented in Figure 1(a). The extraction process consists of (1) transforming the temporal data in the frequency domain, (2) filtering the spectrum based on critical bands analysis, which is derived from speech perception knowledge, (3) applying a non-linear operation and (4) applying a transformation to get reduced dimension decorrelated features. This process only models the local spectral level information, on a short time window. To model the temporal variation intrinsic in the speech signal, dynamic features are computed by taking the first and second derivative of the static features on the longer time window, and concatenate them together. These resulting features are then fed to the acoustic modeling part of the speech recognition system, which can be based on Gaussian mixture model (GMM) or artificial neural networks (ANN). In the case of neural networks, the classifier outputs the conditional probabilities $p(i|x)$, with x denoting the input feature and i the class.

In recent years, deep neural network (DNN) based and deep belief network (DBN) based approaches have been proposed (Hinton et al., 2006), which yield state-of-the-art results in speech recognition using neural networks composed of many hidden layers. In the case of DBN, the networks are

initialized in an unsupervised manner. While this original work relied on MFCC features, several approaches have been proposed to use ‘‘intermediate’’ representations (standing between raw signal and ‘‘classical’’ features such as cepstral-based features) as input. In other words, these are approaches that discard several operations in the extraction pipeline of the conventional features (see Figure 1(b)). For instance, Mel filterbank energies were used as input of convolutional neural networks based systems (Abdel-Hamid et al., 2012; Sainath et al., 2013a; Swietojanski et al., 2014). Deep neural network based systems using spectrum as input has also been proposed (Mohamed et al., 2012; Lee et al., 2009; Sainath et al., 2013b). Combination of different features has also been investigated (Bocchieri & Dimitriadis, 2013).

Learning features directly from the raw speech signal using neural networks-based systems has been investigated. In Jaitly & Hinton (2011), the learned features by a DBN are post-processed by adding their temporal derivatives and used as input for another neural network. A recent study investigated acoustic modeling using raw speech as input to a DNN Tüske et al. (2014). The study showed that raw speech based system is outperformed by spectral feature based system. In our recent studies (Palaz et al., 2013; 2015), we showed that it is possible to estimate phoneme class conditional probabilities by using temporal raw speech signal as input to convolutional neural networks (see Figure 1(c)). This system is composed of several filter stages, which performs the features learning step and which are implemented by convolution and max-pooling layers, and of a classification stage, implemented by a multi-layer perceptron. Both stages are trained jointly. On phoneme recognition and on large vocabulary continuous speech recognition task, we showed that the system is able to learn features from the raw speech signal, and yielded performance similar or better than conventional ANN based system that takes cepstral features as input. The proposed system needed less parameters to yield similar performance with conventional systems, suggesting that the learned features seems to be somehow more efficient than cepstral-based features.

Motivated by these studies, the goal of the present paper is to ascertain the capability of the convolutional neural network based system to learn linearly separable features in a data-driven manner. To this aim, we replace the classifier stage of the CNN-based system, which was a non-linear multi-layer perceptron, by a linear single layer perceptron. Our objective is not to show that the proposed approach yields state-of-the-art performance, rather show that learning features in a data-driven manner together with the classifier leads to flexible features. Using these features as input for a linear classifier yields better performance than SLP-based baseline system and almost reach the performance of MLP-based system.

3 CONVOLUTIONAL NEURAL NETWORKS

This section presents the architecture used in the paper. It is similar to the one presented in (Palaz et al., 2013), and is presented here for the sake of clarity.

3.1 ARCHITECTURE

Our network (see Figure 2) is given a sequence of raw input signal, split into frames, and outputs a score for each classes, for each frame. The network architecture is composed of several filter stages, followed by a classification stage. A filter stage involves a convolutional layer, followed by a temporal pooling layer and a non-linearity ($\tanh(\cdot)$). Processed signal coming out of these stages are fed to a classification stage, which in our case can be either a multi-layer perceptron (MLP) or a linear single layer perceptron (SLP). It outputs the conditional probabilities $p(i|x)$ for each class i , for each frame x .

3.2 CONVOLUTIONAL LAYER

While ‘‘classical’’ linear layers in standard MLPs accept a fixed-size input vector, a convolution layer is assumed to be fed with a sequence of T vectors/frames: $X = \{x^1 \ x^2 \ \dots \ x^T\}$. A convolutional layer applies the same linear transformation over each successive (or interspaced by dW frames) windows of kW frames. For example, the transformation at frame t is formally written

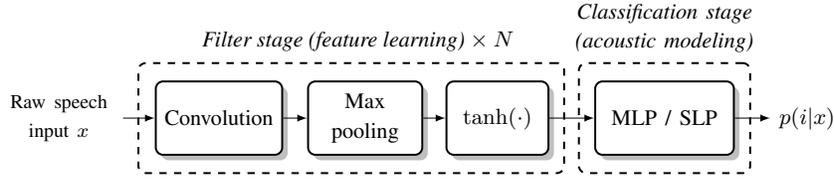


Figure 2: Convolutional neural network based architecture, which estimates the conditional probabilities $p(i|x)$ for each class i , for each frame x . Several stages of convolution/pooling/tanh might be considered. The classification stage can be a multi-layer perceptron or a single layer perceptron.

as:

$$M \begin{pmatrix} x^{t-(kW-1)/2} \\ \vdots \\ x^{t+(kW-1)/2} \end{pmatrix}, \quad (1)$$

where M is a $d_{out} \times d_{in}$ matrix of parameters. In other words, d_{out} filters (rows of the matrix M) are applied to the input sequence.

3.3 MAX-POOLING LAYER

These kind of layers perform local temporal max operations over an input sequence. More formally, the transformation at frame t is written as:

$$\max_{t-(kW-1)/2 \leq s \leq t+(kW-1)/2} x_s^d \quad \forall d \quad (2)$$

with x being the input, kW the kernel width and d the dimension. These layers increase the robustness of the network to minor temporal distortions in the input.

3.4 SOFTMAX LAYER

The *Softmax* (Bridle, 1990) layer interprets network output scores $f_i(x)$ as conditional probabilities, for each class label i :

$$p(i|x) = \frac{e^{f_i(x)}}{\sum_j e^{f_j(x)}} \quad (3)$$

3.5 NETWORK TRAINING

The network parameters θ are learned by maximizing the log-likelihood L , given by:

$$L(\theta) = \sum_{n=1}^N \log(p(i_n|x_n, \theta)) \quad (4)$$

for each input x and label i , over the whole training set (composed of N examples), with respect to the parameters of each layer of the network. Defining the `logsumexp` operation as: $\text{logsumexp}_i(z_i) = \log(\sum_i e^{z_i})$, the likelihood can be expressed as:

$$L = \log(p(i|x)) = f_i(x) - \text{logsumexp}_j(f_j(x)) \quad (5)$$

where $f_i(x)$ described the network score of input x and class i . The log-likelihood is maximized using the stochastic gradient ascent algorithm (Bottou, 1991).

4 EXPERIMENTAL SETUP

In this paper, we investigate using the CNN-based approach on a phoneme recognition task and on a large vocabulary continuous speech recognition task. In this section, we present the two tasks, the databases, the baselines and the hyper-parameters of the networks.

4.1 TASKS

4.1.1 PHONEME RECOGNITION

As a first experiment, we propose a phoneme recognition study, where the CNN-based system is used to estimate phoneme class conditional probabilities. The decoder is a standard HMM decoder, with constrained duration of 3 states, and considering all phoneme equally probable.

4.1.2 LARGE VOCABULARY SPEECH RECOGNITION

We evaluate the scalability of the proposed system on a large vocabulary speech recognition task on the WSJ corpus. The CNN-based system is used to compute the posterior probabilities of context-dependent phonemes. The decoder is an HMM. The scaled likelihoods are estimated by dividing the posterior probability by the prior probability of each class, estimated by counting on the training set. The hyper parameters such as, language scaling factor and the word insertion penalty are determined on the validation set.

4.2 DATABASES

For the phoneme recognition task, we use the TIMIT acoustic-phonetic corpus. It consists of 3,696 training utterances (sampled at 16kHz) from 462 speakers, excluding the SA sentences. The cross-validation set consists of 400 utterances from 50 speakers. The core test set was used to report the results. It contains 192 utterances from 24 speakers, excluding the validation set. The 61 hand labeled phonetic symbols are mapped to 39 phonemes with an additional garbage class, as presented in (Lee & Hon, 1989).

The large vocabulary speech recognition task, we use the the SI-284 set of the Wall Street Journal (WSJ) corpus (Woodland et al., 1994). It is formed by combining data from WSJ0 and WSJ1 databases, sampled at 16 kHz. The set contains 36416 sequences, representing around 80 hours of speech. Ten percent of the set was taken as validation set. The Nov'92 set was selected as test set. It contains 330 sequences from 10 speakers. The dictionary was based on the CMU phoneme set, 40 context-independent phonemes. 2776 tied-states were used in the experiment. They were derived by clustering context-dependent phones in HMM/GMM framework using decision tree state tying. The dictionary and the bigram language model provided by the corpus were used. The vocabulary contains 5000 words.

4.3 FEATURE INPUT

For the CNN-based system, we use raw features as input. They are simply composed of a window of the temporal speech signal (hence, $d_{in} = 1$ for the first convolutional layer). The speech samples in the window are normalized to have zero mean and unit variance.

We also performed several baseline experiments, with MFCC as input features. They were computed (with HTK (Young et al., 2002)) using a 25 ms Hamming window on the speech signal, with a shift of 10 ms. The signal is represented using 13th-order coefficients along with their first and second derivatives, computed on a 9 frames context.

4.4 BASELINE SYSTEMS

We compare our approach with the standard HMM/ANN system using cepstral features. We train a multi-layer perceptron with one hidden layer, referred to as *MLP*, and a linear single layer perceptron, referred to as *SLP*. The system inputs are MFCC with several frames of preceding and following context. We do not pre-train the network. The MLP baseline performance is consistent with other works (Fosler & Morris, 2008).

4.5 NETWORKS HYPER-PARAMETERS

The hyper-parameters of the network are: the input window size w_{in} , corresponding to the context taken along with each example, the kernel width kW_n , the shift dW_n and the number of filters d_n of the n^{th} convolution layer, and the pooling width kW_{mp} . We train the CNN based system with

several filter stages (composed of convolution and max-pooling layers). We use between one and five filter stages. In the case of linear classifier, the capacity of the system cannot be tuned directly. It depends on the size of the input of the classifier, which can be adjusted by manually tuning the hyper-parameters of the filter stages. The hyper-parameters were tuned by early-stopping on the frame level classification accuracy on the validation set. Ranges which were considered for the grid search are reported in Table 1. A fixed learning rate of 10^{-4} was used. Each example has a duration of 10 ms. The experiments were implemented using the *torch7* toolbox (Collobert et al., 2011).

On the TIMIT corpus, using 2 filter stages, the best performance was found with: 310 ms of context, 30 samples width for the first convolution, 7 frames kernel width for the second convolution, 80 and 60 filters and 3 pooling width. Using 3 filter stages, the best performance was found with: 310 ms of context, 30 samples width for the first convolution, 7 and 7 frames kernel width for the other convolutions, 80, 60 and 60 filters and 3 pooling width. Using 4 filter stages, the best performance was found with: 310 ms of context, 30 samples width for the first convolution, 7, 7 and 7 frames kernel width for the other convolutions, 80, 60, 60 and 60 filters and 3 pooling width. We also set the hyper-parameters to have a fixed classifier input. They are presented in Table 2. For the baselines, the *MLP* uses 500 nodes for the hidden layer and 9 frames as context. The *SLP* based system uses 9 frames as context.

On the WSJ corpus, using 1 filter stage, the best performance was found with: 210 ms of context, 30 samples width for the first convolution, 80 filters and 50 pooling width. Using 2 filter stages, the best performance was found with: 310 ms of context, 30 samples width for the first convolution, 7 frames kernel width for the other convolutions, 80 and 40 filters and 7 pooling width. Using 3 filter stages, the best performance was found with: 310 ms of context, 30 samples width for the first convolution, 7 and 7 frames kernel width for the other convolutions, 80, 60 and 60 filters and 3 pooling width. We also ran experiments using hyper-parameters outside the ranges considered previously using 4 filter stages. This experiment has the following hyper-parameters: 310 ms of context, 30 samples width for the first convolution, 25, 25 and 25 frames kernel width for the other convolutions, 80, 60 and 39 filters and 2 pooling width. For the baselines, the *MLP* uses 1000 nodes for the hidden layer and 9 frames as context. The *SLP* based system uses 9 frames as context.

Table 1: Network hyper-parameters ranges considered for tuning on the validation set.

Hyper-parameter	Units	Range
Input window size (w_{in})	ms	100-700
Kernel width of the first conv. (kW_1)	samples	10-90
Kernel width of the n^{th} conv. (kW_n)	frames	1-11
Number of filters per kernel (d_{out})	filters	20-100
Max-pooling kernel width (kW_{mp})	frames	2-6

Table 2: Network hyper-parameters for a fixed output size

# conv. layer	w_{in}	kW_1	kW_2	kW_3	kW_4	kW_5	d_n	kW_{mp}	# output
1	310	3	na	na	na	na	39	50	351
2	310	3	7	na	na	na	39	7	351
3	430	3	5	5	na	na	39	4	351
4	510	3	5	3	3	na	39	3	351
5	310	3	5	7	7	7	39	2	351

5 RESULTS

The results for the phoneme recognition task on the TIMIT corpus are presented in Table 3. The performance is expressed in terms of phone error rate (PER). The number of parameters in the classifier and in the filter stages are also presented. Using a linear classifier, the proposed CNN-based system outperforms the MLP based baseline with three or more filter stages. It can be observed that

the performance of the CNN-based system improves with increase in number of convolution layers and almost approaches the case where a MLP (with 60 more parameters) is used in the classification stages. Furthermore, it can be observed that the complexity of the classification stage decreases drastically with the increase in the number of convolution layers. The results for the proposed system with a fixed output size is presented in Table 4, along with the baseline performance and the number of the parameters in the classifier and filter stages. The proposed CNN based system outperforms the SLP based baseline with the same number of parameters in the classifier. Fixing the output size seems to degrade the performance compared to Table 3. This indicate that it is better to treat the feature size also as a hyper-parameter and learn it on the data.

Table 3: Results on the TIMIT core testset

Features	# conv. layers	# conv. param.	Classifier	# classifier param.	PER
MFCC	na	na	MLP	200k	33.3 %
RAW	3	61k	MLP	470k	29.6 %
MFCC	na	na	SLP	14k	51.5 %
RAW	2	36k	SLP	124k	38.0 %
RAW	3	61k	SLP	36k	31.5 %
RAW	4	85k	SLP	7k	30.2 %

Table 4: Results for a fixed output on the TIMIT core testset

Features	# conv. layers	# conv. param.	Classifier	# classifier param.	PER
MFCC	na	na	SLP	14k	51.5 %
RAW	1	1.2k	SLP	14k	49.3%
RAW	2	24k	SLP	14k	38.0 %
RAW	3	152k	SLP	14k	33.4 %
RAW	4	270k	SLP	14k	34.6 %
RAW	5	520k	SLP	14k	33.1 %

The results for the large vocabulary continuous speech recognition task on the WSJ corpus are presented in Table 5. The performance is expressed in term of word error rate (WER). We observe a similar trend to the TIMIT results, i.e. with the increase in number of convolution layers the performance of the system improves. More specifically, it can be observed that with only two convolution layers the proposed system is able to achieve performance comparable to SLP-based system with MFCC as input. With three convolution layers the proposed system is approaching the MLP-based systems. With four convolution layers, the system is able to yield similar performance with the MLP baseline using MFCC as input.

Overall, it can be observed that the CNN-based approach can lead to systems with simple classifiers, i.e. with a small number of parameters, thus shifting the system capacity to the feature learning stage of the system. On the phoneme recognition study (see Table 3), the proposed approach even leads to a system where most parameters lie in the feature learning stage rather than in the classification stage. This system yields performance similar to or better than baselines system. On the continuous speech recognition study, it can be observed that the four convolution layers experiment has five times less parameters in the classifier than the three layers experiment and still yields better performance. This four layers experiment is also able to yield similar performance to the MLP-based baseline with two times less parameters.

6 DISCUSSION AND CONCLUSION

Traditionally in speech recognition systems, feature extraction and acoustic modeling (classifier training) are dealt in two separate steps, where feature extraction is knowledge-driven, and classifier

Table 5: Results on the Nov’92 testset of the WSJ corpus.

Features	# conv. layers	# conv. param.	Classifier	# classifier param.	WER
MFCC	na	na	MLP	3M	7.0 %
RAW	3	55k	MLP	3M	6.7 %
MFCC	na	na	SLP	1M	10.9 %
RAW	1	5k	SLP	1.3M	15.5 %
RAW	2	27k	SLP	1M	10.5 %
RAW	3	64k	SLP	2.4M	7.6 %
RAW	4	180k	SLP	410k	6.9 %

training in data-driven. In the CNN-based approach with raw speech signal as input, both feature extraction and classifier training is data-driven. Such an approach allows the features to be flexible as they are learned along with the classifier. It also allows to shift the system capacity from the classifier stage to the feature extraction stage of the system. Our studies indicate that these empirically learned features can be linearly separable and could yield systems that perform similar to or better than standard spectral-based systems. This can have potential implication for low resource speech recognition. This is part of our future investigation.

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REFERENCES

- Abdel-Hamid, O., Mohamed, A., Jiang, H., and Penn, G. Applying convolutional neural networks concepts to hybrid NN-HMM model for speech recognition. In *Proc. of ICASSP*, pp. 4277–4280, 2012.
- Bocchieri, E. and Dimitriadis, D. Investigating deep neural network based transforms of robust audio features for lvcsr. In *Proc. of ICASSP*, pp. 6709–6713, 2013.
- Bottou, L. Stochastic gradient learning in neural networks. In *Proceedings of Neuro-Nmes 91*, Nimes, France, 1991. EC2.
- Bourlard, H. and Morgan, N. *Connectionist speech recognition: a hybrid approach*, volume 247. Springer, 1994.
- Bridle, J.S. Probabilistic interpretation of feedforward classification network outputs, with relationships to statistical pattern recognition. In *Neuro-computing: Algorithms, Architectures and Applications*, pp. 227–236. 1990.
- Collobert, R., Kavukcuoglu, K., and Farabet, C. Torch7: A matlab-like environment for machine learning. In *BigLearn, NIPS Workshop*, 2011.
- Davis, S. and Mermelstein, P. Comparison of parametric representations for monosyllabic word recognition in continuously spoken sentences. *IEEE Transactions on Acoustics, Speech and Signal Processing*, 28(4):357–366, 1980.
- Fosler, E.L. and Morris, J. Crandem systems: Conditional random field acoustic models for hidden markov models. In *Acoustics, Speech and Signal Processing, 2008. ICASSP 2008. IEEE International Conference on*, pp. 4049–4052, April 2008.
- Hermansky, H. Perceptual linear predictive (plp) analysis of speech. *The Journal of the Acoustical Society of America*, 87:1738, 1990.

- Hinton, G., Deng, L., Yu, D., Dahl, G. E., Mohamed, A., Jaitly, N., Senior, A., Vanhoucke, V., Nguyen, P., and Sainath, T. N. Deep neural networks for acoustic modeling in speech recognition: the shared views of four research groups. *Signal Processing Magazine, IEEE*, 29(6):8297, 2012.
- Hinton, G. E., Osindero, S., and Teh, Y. W. A fast learning algorithm for deep belief nets. *Neural computation*, 18(7):1527–1554, 2006.
- Jaitly, N. and Hinton, G. Learning a better representation of speech soundwaves using restricted boltzmann machines. In *Proc. of ICASSP*, pp. 5884–5887, 2011.
- LeCun, Y. Generalization and network design strategies. In Pfeifer, R., Schreter, Z., Fogelman, F., and Steels, L. (eds.), *Connectionism in Perspective*, Zurich, Switzerland, 1989. Elsevier.
- Lee, H., Pham, P., Largman, Y., and Ng, A. Y. Unsupervised feature learning for audio classification using convolutional deep belief networks. In *Advances in Neural Information Processing Systems 22*, pp. 1096–1104, 2009.
- Lee, K. F and Hon, H. W. Speaker-independent phone recognition using hidden markov models. *IEEE Transactions on Acoustics, Speech and Signal Processing*, 37(11):1641–1648, 1989.
- Mohamed, A., Dahl, G.E., and Hinton, G. Acoustic modeling using deep belief networks. *IEEE Transactions on Audio, Speech, and Language Processing*, 20(1):14–22, jan. 2012.
- Palaz, D., Collobert, R., and Magimai.-Doss, M. Estimating phoneme class conditional probabilities from raw speech signal using convolutional neural networks. In *Proc. of Interspeech*, 2013.
- Palaz, D., Magimai.-Doss, M., and Collobert, R. Convolutional neural networks-based continuous speech recognition using raw speech signal. In *Proc. of ICASSP, preprint*, April 2015.
- Sainath, T. N., Mohamed, A., Kingsbury, B., and Ramabhadran, B. Deep convolutional neural networks for lvcsr. In *Proc. of ICASSP*, pp. 8614–8618, 2013a.
- Sainath, T.N., Kingsbury, B., Mohamed, A.-R., and Ramabhadran, B. Learning filter banks within a deep neural network framework. In *Proc. of ASRU*, pp. 297–302, December 2013b.
- Swietojanski, P., Ghoshal, A., and Renals, S. Convolutional neural networks for distant speech recognition. *Signal Processing Letters, IEEE*, 21(9):1120–1124, September 2014.
- Tüske, Z., Golik, P., Schlüter, R., and Ney, H. Acoustic modeling with deep neural networks using raw time signal for lvcsr. In *Interspeech*, pp. 890–894, Singapore, September 2014.
- Woodland, P.C., Odell, J.J., Valtchev, V., and Young, S.J. Large vocabulary continuous speech recognition using htk. In *Proc. of ICASSP*, volume ii, pp. II/125–II/128 vol.2, apr 1994.
- Young, S., Evermann, G., Kershaw, D., Moore, G., Odell, J., Ollason, D., Valtchev, V., and Woodland, P. The htk book. *Cambridge University Engineering Department*, 3, 2002.