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An Improved Deep Embedding Learning Method for Short Duration Speaker Verification

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Abstract

This paper presents an improved deep embedding learning method based on convolutional neural network (CNN) for short-duration speaker verification (SV). Existing deep learning-based SV methods generally extract frontend embeddings from a feed-forward deep neural network, in which the long-term speaker characteristics are captured via a pooling operation over the input speech. The extracted embeddings are then scored via a backend model, such as Probabilistic Linear Discriminative Analysis (PLDA). Two improvements are proposed for frontend embedding learning based on the CNN structure: (1) Motivated by the WaveNet for speech synthesis, dilated filters are designed to achieve a tradeoff between computational efficiency and receptive-filter size; and (2) A novel cross-convolutional-layer pooling method is exploited to capture 1^{st}-order statistics for modelling long-term speaker characteristics. Specifically, the activations of one convolutional layer are aggregated with the guidance of the feature maps from the successive layer. To evaluate the effectiveness of our proposed methods, extensive experiments are conducted on the modified female portion of NIST SRE 2010 evaluations, with conditions ranging from 10s-10s to 5s-4s. Excellent performance has been achieved on each evaluation condition, significantly outperforming existing SV systems using i-vector and d-vector embeddings.

Index Terms: speaker verification, convolution neural network, dilated convolution, cross-convolutional-layer pooling

1. Introduction

Speaker verification (SV) is the task of determining whether the claimed identity of a speaker matches an enrolled identity, according to their speech. A typical SV system consists of a frontend embedding learning stage and a backend modeling stage. A low-dimensional embedding that is rich in speaker information is extracted in the frontend learning stage, and the similarities between embeddings are computed for verification by the backend modelling stage.

For decades, the most successful SV systems have relied on i-vectors with a PLDA backend [1, 2, 3, 4, 5, 6], which model speaker representations and channel variability in a low-dimensional space. An i-vector is learned through a pipeline of generative modelling, as shown in Section 2. Despite excellent performance on long-duration evaluations, the effectiveness of i-vectors degrades dramatically for short-duration recordings [7, 8].

Several methods based on deep learning have recently been proposed to extract frontend embeddings for short-duration SV [9, 10, 7, 8, 11]. In [9], deep neural networks (DNNs) were employed to learn speaker embeddings, termed d-vectors. In [7, 11], deep CNNs were exploited to model long-term temporal dependencies. CNNs were shown to have better discriminative ability than DNNs[12], leading to a certain robustness to variability caused by different channels, gender and speech content. After training, d-vectors are extracted by averaging the last hidden layer activations for enrolment and test recordings. In [10, 8], aggregation was improved by incorporating variance information into speaker embeddings via statistical pooling.

However, existing deep learning methods still have several shortcomings. One drawback is the dilemma between learning efficiency and receptive-field size. To acquire a large receptive-field size, a CNN may require many filters with large kernel size, or many layers, which may be inefficient and difficult to converge. Another weakness is that average-pooling methods for aggregating frame-level features are insufficient, since only 0^{th}-order statistics are utilized. Recently, statistical pooling has shown good performance [10, 8], and hence we aim to exploit this through a better aggregation method. Our proposed improved deep embedding learning architecture is shown in Fig 1.

Its architecture is based on a fully convolutional network structure, with two main improvements. Firstly, the dilated filters are able to achieve a better tradeoff between computational efficiency and receptive-filter size. This is mainly inspired by WaveNet [13], in which a dilated causal CNN is exploited to efficiently enlarge the receptive-field size with low computation complexity. It in essence, is similar to the time-delay architecture in the 1D-CNN [14]. The major improvement comes from dilated filters that are more flexible in both the temporal and frequency domain. Secondly, a novel cross-convolutional-layer pooling method is proposed for better embedding learning. This uses the feature maps of a higher layer to guide the aggregation of activations from a convolutional layer. This successive layer can be viewed as a probabilistic discriminative model for deriving 1^{st}-order statistics.

In this paper, we are interested in short-duration text-independent SV, similar to [11]. It is worth noting that it is easy to extend from this scenario in future, to other SV tasks such as text-dependent and long-duration SV. To demonstrate the effectiveness and robustness of the proposed method, we have conducted extensive evaluations of short-duration SV with experimental conditions ranging from 10s-10s to 5s-4s. Excellent performance has been achieved in each condition when compared to state-of-the-art i-vector and other d-vector embedding methods.

The remainder of the paper is organized as follows. Section 2 describes i-vector baseline, Section 3 details the CNN-based SV system. Experimental results and discussion follow in Section 4 before Section 5 concludes the paper.

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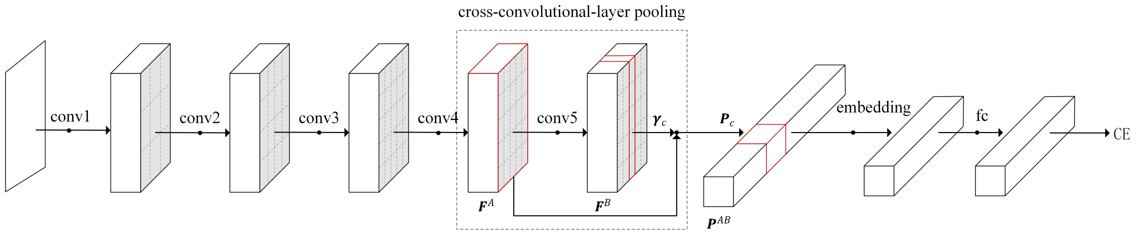

Figure 1: *Architecture of proposed CNN*

2. Review of i-vector based SV

SV systems based on i-vectors with a PLDA backend currently achieve state-of-the-art performance [4]. In these systems, the i-vector is learned via a pipeline of generative modeling including: (1) a universal background model (UBM) training, which is used to collect sufficient statistics, and (2) a large loading matrix learning, which projects high-dimensional sufficient statistics *i.e.*, supervectors to a low-dimensional space that contains speaker information and channel variability. Specifically, given an utterance, the speaker-dependent GMM supervector can be defined as:

$$\boldsymbol{M} = \boldsymbol{m} + \boldsymbol{T}\boldsymbol{w} \tag{1}$$

where $\boldsymbol{m}$ is the speaker-independent supervector, taken from the UBM. $\boldsymbol{T}$ is loading matrix with low rank, seen as the total variability matrix. And $\boldsymbol{w}$ is the speaker and channel factor with a standard normal distribution $\boldsymbol{N}(\boldsymbol{0}, \boldsymbol{I})$.

However, it is known that for short duration SV, the statistics are not sufficient for reliable i-vector learning, which leads to degraded performance [15]. Furthermore, the generative models obtained via unsupervised learning methods may be improved with discriminative models, *e.g.*, d-vectors learned by DNN or CNN.

3. An improved deep speaker embedding framework

3.1. Overview

We propose an improved CNN-based deep embedding learning method for short-duration SV. The network architecture is depicted in Fig. 1. The dilated convolution enables the network to learn long-term temporal content with low computational complexity. Frame-level features are then aggregated by cross-convolutional-layer pooling, which is designed to exploit 1^{st}-order statistics. The proposed CNN is then trained to discriminate variable-length input features between speakers in an end-to-end manner. After training, speaker embeddings are extracted, and similarity scores are calculated with a PLDA backend.

3.2. Dilated convolution

The dilated convolution was originally proposed for wavelet decomposition to extract dense features [16]. It has also been widely used in image segmentation to increase image resolution [17, 18, 19]. More recently, WaveNet utilized dilated convolution to enlarge the receptive field in speech synthesis [13].

The main idea of dilated convolution is to insert "holes" in convolutional kernels so as to enlarge the receptive field. The dilated convolution enables the convolutional kernels to filter on a larger effective area than its own size. Its receptive-field size implies a convolution with a large kernel generated from an original kernel by dilating it with zeros, and is more computationally efficient than simply increasing the original kernel size. A dilation factor, defining the size of dilation, of 1, equates to the standard convolution.

In this paper, we propose an efficient dilated CNN framework. We stack the dilated convolutional layers to obtain large receptive field with just a few layers, which is highly computationally efficient. The network can model long-term temporal dependencies through the enlarged network receptive field. Dilated convolution enables a tradeoff to be found between learning efficiency and receptive-field size. The setup of dilation is described in Section 4.2, where an intuitive exponential increase in dilation factor leads to an exponentially increased receptive-field size for each CNN layer.

3.3. Cross-convolutional-layer pooling

We exploit a novel cross-convolutional-layer pooling method to capture 1^{st}-order statistics for modelling long-term speaker characteristics. The cross-layer pooling method is motivated by statistical pooling, in which high order statistics can be used to improve performance. The cross-convolutional-layer pooling step resides within the dotted box in the centre of Fig. 1.

The insight to perform cross-convolutional-layer pooling is that it can aggregate features across different layers [20], with formation of cross-convolutional-layer pooling defined as:

$$\boldsymbol{P}^{A,B} = P(\boldsymbol{F}^A, \boldsymbol{F}^B) \tag{2}$$

where $\boldsymbol{F}^A$ and $\boldsymbol{F}^B$ are feature maps derived from different layers in a hierarchical architecture (written as $\boldsymbol{F}^A$ layer and $\boldsymbol{F}^B$ layer respectively). The shape of $\boldsymbol{F}^A$ and $\boldsymbol{F}^B$ are $N_A \times C_A$ and $N_B \times C_B$ respectively (reshaped from $H_A \times W_A \times C_A$ and $H_B \times W_B \times C_B$ respectively). $\boldsymbol{P}^{A,B}$ is the output of cross-convolutional-layer pooling with shape $1 \times (C_A \times C_B)$. Furthermore, $\boldsymbol{P}^{A,B}$ can be viewed as the pooled features by concatenating the pooled features of each channel:

$$\boldsymbol{P}^{A,B} = [\boldsymbol{P}_1, \boldsymbol{P}_2, ..., \boldsymbol{P}_c, ..., \boldsymbol{P}_{C_B}] \tag{3}$$

where, $c = 1, 2, ..., C_B$, the shape of $\boldsymbol{P}_c$ is $1 \times C_A$, and $\boldsymbol{P}_c$ is defined as:

$$\boldsymbol{P}_c = \sum_{t=1}^{N_A} F_{t,c}^B \boldsymbol{F}_t^A \tag{4}$$

The shape of $\boldsymbol{F}_t^A$ is $1 \times C_A$, and can be considered as the t^{th} feature of $\boldsymbol{F}^A$. The $F_{t,c}^B$ is the t^{th} feature on the c^{th} channel of $\boldsymbol{F}^B$. The correspondence between $F_{t,c}^B$ and $\boldsymbol{F}_t^A$ is defined as:

$$F_{t,c}^B = f_c(\boldsymbol{F}_t^A) \tag{5}$$

where, f_c is the nonlinear mapping function of $\boldsymbol{F}^B$ layer. If we review $\boldsymbol{F}^B$ layer is a phonetic recognition extractor, and set the

Table 1: *Details of CNN architecture*

conv1	conv2	conv3	conv4	conv5	embedding	fc
$512@23 \times 5$	$512@1 \times 3$	$512@1 \times 3$	$512@1 \times 1$	$512@1 \times 1$	512	300

posterior $\gamma_{t,c} = F^B_{t,c}/N_A (t = 1, 2, ..., N_A, c = 1, 2, ..., C_B)$, $\boldsymbol{\gamma}_c^T = [\gamma_{1,c}, \gamma_{2,c}, ..., \gamma_{N_A,c}]$, then (4) could be rewritten as:

$$\boldsymbol{P}_c = \frac{1}{N_A} \sum_{t=1}^{N_A} \gamma_{t,c} \boldsymbol{F}_t^A \tag{6}$$

which in turn can be viewed as 1^{st}-order Baum-Welch statistics ($\boldsymbol{F}^A$ and $\boldsymbol{F}^B$ are all mean-normalized). f_c can then be implicitly reinterpreted as a probabilistic discriminative model for deriving 1^{st}-order statistics. In summary, the $\boldsymbol{F}^B$ layer is used as a guide for statistical aggregation of layer $\boldsymbol{F}^A$.

Before being fed into the subsequent layer, the resulting cross-convolutional-layer vectors are passed through a signed square root step, followed by l_2 normalization.

3.4. Architecture of proposed CNN

The architecture of the proposed CNN is depicted in Fig. 1. The network inputs are raw MFCC features and there are five convolution layers followed by cross-convolutional-layer pooling. A fully connected layer named the embedding layer is inserted to extract deep speaker embeddings. The last layer is a fully connected layer, fed into the softmax output layer. The nonlinear function is ReLu, and BN [21] is applied to every layer.

The network is trained to discriminate between training speakers with cross-entropy loss. After training, speaker embeddings are extracted from embedding layers. The details of the CNN architecture are summarized in Table 1 where a convolution of 512@23×5, means that the number of kernels is 512, and the kernel size is 23×5. The padding and stride of each convolution are 0 and 1 respectively.

4. Experiments

4.1. Dataset

This paper focuses on short-duration SV, where both sides of the verification trials are short-duration recordings. We evaluate the performance of the female portion of the NIST 2010 SRE evaluation [22], which is the same as [11]. To be comparable with state-of-the-art approaches [8, 11], our enrolment sets are cut into two different durations, *e.g.*, 10s and 5s, where we select the first 10s and 5s respectively from original recordings, as determined by VAD. Similar to enrolment, the test recordings are truncated to the first $T \in \{10, 5, 4\}$s of speech. In this paper, we term 10s enrolment and 10s test condition as 10s-10s, following notation in [11], and apply other testing conditions similarly. Training datasets are from NIST04-08 plus a portion of Switchboard including male and female speakers. We construct the training dataset by discarding any recording that is less than 5 seconds long and any speaker with fewer than 8 recordings. After that, there are 34446 recordings of 2253 speakers remaining.

4.2. Experiment setup

In order to evaluate the performance of our proposed method, we compare several state-of-the-art SV systems. The training dataset for all systems is the same. For embedding systems based on neural networks, the input features are raw MFCCs of dimension 23, and are all mean-normalized. An energy-based VAD is applied to filter out silent frames. Before being input to the network, input features are sliced into short durations ranging from 2s to 4s, generating 3400 features per speaker. We utilize SGD to optimize the network with a momentum rate of 0.9. The batch size is 64, the initial learning rate is 0.1, and this is multiplied by 0.1 with every epoch. All of the networks are trained within 5 epochs to converge. After training, the LDA and PLDA backends are employed to calculate scores. The LDA dimension is 100 and scores are not normalized. We trained the CNN by using Pytorch [23]. The six comparison systems are;

I-vector: The training dataset for the i-vector system is the same as the embedding system based on the neural network. Feature vectors are extended with delta and delta-delta to become 69 dimensional, which is a standard procedure. The UBM is a 2048 component full-covariance GMM and the i-vector dimension is 400. The LDA and PLDA backend is the same as in other systems and the system is implemented using Kaldi [24].

CNN-G-D1/D2: This is the first baseline. It follows Fig. 1 except that the cross-convolutional-layer pooling is replaced with basic single-layer average pooling. In CNN-G-D1, the dilation factors of convolutional layers are all 1. In CNN-G-D2, the dilation factors of convolutional layers are set to 1, 2, 4, 1 and 1 respectively.

CNN-S-D1/D2: This is the second baseline. In this case we replace the cross-convolutional-layer pooling with single-layer statistical pooling (*i.e.* this also has no cross-layer connection). In CNN-S-D1, the dilation factors of convolutional layers are all 1. In CNN-S-D2, the dilation factors of convolutional layers are 1, 2, 4, 1 and 1 respectively.

CNN-C-D1/D2: This is our proposed method, as depicted in Fig. 1. In CNN-C-D1, the dilation factors of convolutional layers are all 1. In CNN-C-D2, the dilation factors of convolutional layers are 1, 2, 4, 1 and 1 respectively.

TDNN: Snyder et al. [8] obtained state-of-the-art performance in short-duration SV evaluation. The code[1] was released by the author. We retrained their model on our dataset using Kaldi, and used the same setup described in their paper [8].

VGGnet: Bhattacharya et al. [11] demonstrated the superiority of deep CNNs over i-vectors for short-duration testing trials. Since they did not release their code, we reproduced the results that their paper reported for identical testing conditions. However it should be noted that their training dataset was twice as large as ours, giving their system a potential advantage.

4.3. Results for the 10s enrolment condition

In this section, we evaluate performance for 10s enrolment conditions, described in Table 2. We do not show the performances of CNN-G-D1 and CNN-S-D1 for the sake of clarity, since their performance is inferior in each case to the -D2 system variants.

From Table 2, we can see that when we aggregate features by average pooling, performance dramatically degrades compared to the i-vector system. This is consistent with [9].

[1] https://david-ryan-snyder.github.io/2017/10/04/model_sre16_v2.html

Table 2: *EER(%) of SV in 10s enrolment condition*

System \ Condition	10s – 10s	10s – 5s	10s – 4s
I-vector	17.95	20.77	22.18
CNN-G-D2	28.38	32.23	34.06
CNN-S-D2	21.47	26.05	27.81
CNN-C-D1	17.60	20.77	22.88
CNN-C-D2	16.54	20.65	21.83
CNN-C-Fusion	**14.78**	**17.60**	**19.36**
TDNN	16.9	20.42	21.12
VGGnet	17.51	–	–

The CNN-S-D2 system achieves a large 18%$\sim$24% relative improvement over CNN-G-D2. Clearly, statistical pooling, which effectively incorporates variance information, enables an improvement in performance. Compared with CNN-S-D2, the CNN-C-D2 system then obtains a further 22%, 20% and 21% relative improvements in 10s-10s, 10s-5s and 10s-4s evaluation respectively. In fact, CNN-C-D2 outperforms the i-vector system for each evaluation condition.

When we compare CNN-C-D2 with CNN-C-D1, the EER improves as expected; this demonstrates that dilation enlarges the receptive field to yield benefit for the SV task.

CNN-C-D2 obtains state-of-the-art performance in the 10s-10s evaluation, and achieves comparable performance to TDNN in the 10s-5s and 10s-4s evaluations as well. However when we fuse CNN-C-D1 and CNN-C-D2, termed CNN-C-Fusion, we obtain better performance for each evaluation. In fact CNN-C-Fusion system improves EER compared to the i-vector system by 17%, 15% and 12% in 10s-10s, 10s-5s and 10s-4s evaluations respectively.

Table 3: *EER(%) of SV in 5s enrolment condition*

System \ Condition	5s – 10s	5s – 5s	5s – 4s
I-vector	20.77	23.23	25.00
CNN-C-D1	20.07	25.35	25.75
CNN-C-D2	19.01	22.53	23.59
CNN-C-Fusion	**17.60**	**20.42**	**21.83**
TDNN	19.01	23.23	25.70
VGGnet	–	23.16	–

4.4. Results for the 5s enrolment condition

In this section, we evaluate performance for 5s enrolment conditions, described in Table 3. We omit the performance of the CNN-G-D1/D2 and CNN-S-D1/D2 for the sake of clarity, since they are similar to 10s enrolment in Section 4.3.

Table 3 shows that the proposed CNN-C-D2 system obtains state-of-the-art performance in each evaluation. In Section 4.3, TDNN obtained similarly good performance in the 10s-5s and 10s-4s evaluation, however, for this shorter duration enrolment, TDNN is less robust. TDNN performance is slightly worser than the i-vector system in the 5s-4s evaluation, and CNN-C-D2 performs better than all tested systems for all experimental conditions.

VGGnet reported state-of-the-art performance in 5s-5s evaluation in [11], however the CNN-C-D2 system is shown to perform better for this evaluation.

When we then fuse the CNN-C-D1 and CNN-C-D2 systems, termed CNN-C-Fusion, we gain 15%, 12% and 12% relative improvements for the 5s-10s, 5s-5s and 5s-4s evaluations respectively.

4.5. Discussion

CNN-C-D2 demonstrates consistent improvements over CNN-C-D1 since it has a larger receptive field. The dilated convolution enlarges the filter size by dilating the filter with zeros. However, it is possible that the two networks focus on different patterns. The filters enlarged by dilation tend to learn global features or patterns, while filters with no dilation are more prone to learn local features. We infer that the two networks are thus learning some information that may be complementary. This appears to be confirmed by the excellent performance achieved by the fusion system (termed CNN-C-Fusion).

We have compared the performance of average and statistical pooling. If we see from the Baum-Welch statistics point of view, average pooling can be viewed as 0^{th}-order statistics, whereas statistical pooling incorporates variance which can be viewed as being 2^{nd}-order statistics. The results conform that high-order statistics are significant for short-duration SV. However, it does not make full use of covariance, since it assumes that the covariance is diagonal. We thus proposed cross-convolutional-layer pooling to capture 1^{st}-order statistics for modelling long-term speaker characteristics. Specifically, the activations of one convolutional layer are aggregated with the guidance of its successive layer. This technique achieved state-of-the-art performance in each evaluation condition. The results clearly demonstrate that cross-convolutional-layer pooling is a more efficient method for the aggregation of frame-level features.

5. Conclusions

In this paper, we present an improved deep embedding learning method based on CNN for short-duration SV task. Two main improvments have been proposed based on a fully convolutional network structure. Firstly, the dilated filters are designed, which enable the convolution layers to learn long-term temporal context information with relative low computation complexity. Secondly, a coss-convolutional-layer pooling method is proposed to aggregate the frame-level convolutional features, which effectively derives 1^{st}-order statistics for better embedding learning. The proposed CNNs are trained to discriminate variable-length input features between speakers in an end-to-end manner.

To evaluate the effectivness of learned embeddings, extensive experiments have been conducted on the modified female part of the NIST 2010 SRE evaluation, consisting of 10s and 5s enrolment conditions. The state-of-the-art performance has been achieved in each evaluation condition. Specially, results show a 17% and 12% relative improvement for 10s-10s and 5s-5s evaluations respectively compared with the i-vector system. In future work, we aim to extend the proposed model to a larger number of speakers, to further investigate the data dependency of the embedding learning approaches.

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7. References

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