Improvement of distant-talking speaker identification using bottleneck features of DNN

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Abstract

In this paper we propose bottleneck features of deep neural network for distant-talking speaker identification. The accuracy of distant-talking speaker recognition is significantly degraded under reverberant environment. Feature mapping or feature transformation has been shown efficacy in channel-mismatch speaker recognition. Bottleneck feature derived from multi-layer network, which is a nonlinear feature transformation method, has been shown efficacy in automatic speech recognition (ASR) system. In this study, bottleneck features extracted from deep neural networks (DNNs) which employ an unsupervised pre-training method are used as nonlinear feature transformation for distant-talking speech. The speaker identification experiment was performed on large-scale distant-talking speech set, with reverberant environments different to the training environments. The proposed bottleneck features achieved a relative error reduction of 46.3% compared with conventional MFCC. Moreover, a combination of likelihoods of bottleneck features and MFCC achieved a further improvement.

Index Terms: speaker recognition, bottleneck features, pre-training, deep neural network, reverberant speech

1. Introduction

Due to the effects of reverberation, the accuracy of distant-talking speaker identification is significantly reduced. Many techniques such as dereverberation method, feature warping and feature transformation have been proposed for robust distant-talking speaker recognition. The most general approach may be cepstral mean normalization (CMN) [1]. However, length of impulse response in a distant-talking environment is usually much longer than analysis window size of short-term spectral analysis. Therefore, the CMN cannot compensate for late reverberation. Some spectral subtraction-based dereverberation methods have been proposed for distant-talking speaker recognition [2-4]. They treated late reverberation as additive noise, and a noise reduction technique based on spectral subtraction was proposed to suppress the late reverberation. To construct a more robust representation of each cepstral feature distribution, feature warping method was proposed [2][5]. This method warps the distribution of a cepstral feature stream to a standardized distribution over a specified time interval. In addition, feature transformation approach was proposed for robust distant-talking speaker recognition [6]. The transformation is applied to the distorted feature before mapping them to a normal distribution, and aims to decorrelate the feature vectors, making them more amendable to diagonal covariance Gaussian mixture model (GMM). Bottleneck features extracted by a multi-layer perceptron (MLP) can be used a non-linear feature transformation and dimensionality reduction [7]. An MLP was trained by backpropagation (BP) algorithm from random initial parameters. And then the bottleneck features were extracted by dimensionality reduction of several frames of cepstral coefficients. The combination of bottleneck features and cepstral coefficient is better than the conventional MFCC. However, deep networks of MLP with many hidden layers have a high computational cost, and can’t learn much further away from the top layer. In addition, it does not lead to improved accuracy even after a long time training by the BP algorithm using the initial random value. Recently, deep belief networks (DBNs) which employ an unsupervised pretraining method using restricted Boltzmann machine (RBM) have been proposed to train better initial values of deep networks [8]. A RBM is obtained by adding constraints that there is no connection of the same layer and connected to the symmetry to the Boltzmann machine. Its parameter is determined by the Greedy layer-wise training. This training method minimize the error between the input feature and the inverse transform feature of the input. After pretraining, DBNs are obtained by discriminative trained using BP algorithm from DBNs which are multi-layer structure stacked RBMs. DBNs with pretraining have been shown better performance than the conventional MLP without pretraining on automatic speech recognition [9] and large vocabulary business search task [10] etc.

In this paper, we propose a robust distant-talking speaker identification method using the bottleneck features extracted from DNNs. We consider that the DNNs can transform the reverberant speech feature to a new feature space close to clean speech feature which has more discriminative classification ability for distant-talking speaker recognition. In addition, by using multiple contexts (frames) for input data, the bottleneck features can reduce the influence of reverberation over frames. Bottleneck features extracted from the DNNs are used for training the conventional speaker-dependent GMM. Furthermore, the likelihood of MFCC and that of the bottleneck features are combined linearly.

2. Deep Neural Networks

Although DNNs can be seen as improved MLPs, discriminative training method is the same. The advantages of DNN is pretraining which can determine the good initial value of the multi-layer networks. However, pretraining of multi-layer networks is computationally expensive. Therefore, we used RBM for efficient computation.

2.1. Restricted Boltzmann Machine

RBM is a bipartite graph shown in Figure 1. It has visible and
hidden layer in which visible units that represent observations are connected to hidden units that learn to represent features using weighted connection. An RBM is restricted that there are no visible-visible or hidden-hidden connections. Different types of RBM is used in the case of binary or real-valued input. Bernoulli-Bernoulli RBMs used to convert binary stochastic variables to binary stochastic variables. Gaussian-Bernoulli RBMs is used to convert real-valued stochastic variables to binary stochastic variables.

In a Bernoulli-Bernoulli RBMs, the weights on the connections and the biases of the individual units define a probability distribution over the joint states of the visible and hidden units via an energy function. The energy of a joint configuration is:

\[ E(v, h|\theta) = -\sum_{i=1}^{V} \sum_{j=1}^{H} w_{ij}v_i h_j - \sum_{i=1}^{V} a_i v_i - \sum_{j=1}^{H} b_j h_j \]  

where \(\theta = (w, a, b)\) and \(w_{ij}\) represents the symmetric interaction term between visible unit \(i\) and hidden unit \(j\) while \(a_i\) and \(b_j\) are their bias term. \(V\) and \(H\) are the numbers of visible and hidden units.

The probability that an RBM assigns to a visible vector \(v\) is:

\[ p(v|\theta) = \frac{\exp(-E(v, h|\theta))}{\sum_v \sum_h \exp(-E(v, h|\theta))} \]  

Since there are no hidden-hidden connections, the conditional distribution \(p(h|v, \theta)\) is factorial and is given by:

\[ p(h_j = 1|v, \theta) = \sigma(b_j + \sum_{i=1}^{V} w_{ij} v_i) \]  

where \(\sigma(x) = (1 + \exp(-x))^{-1}\). Similarly, since there are no visible-visible connections, the conditional distribution \(p(v|h, \theta)\) is factorial and is given by:

\[ p(v_i = 1|h, \theta) = \sigma(a_i + \sum_{j=1}^{H} w_{ij} h_j) \]  

In a Gaussian-Bernoulli RBMs, the energy of a joint configuration is:

\[ E(v, h|\theta) = \sum_{i=1}^{V} \frac{(v_i - a_i)^2}{2} - \sum_{i=1}^{V} \sum_{j=1}^{H} w_{ij} v_i h_j - \sum_{j=1}^{H} b_j h_j \]  

The conditional distribution \(p(h|v, \theta)\) is factorial and is given by:

\[ p(h_i = 1|h, \theta) = \mathcal{N}(v_i; a_i + \sum_{j=1}^{H} w_{ij} h_j, 1) \]  

where \(\mathcal{N}(\mu, V)\) is a Gaussian with mean \(\mu\) and variance \(V\).
both MLP without pretraining and DBNs with pretraining. Finally, the bottleneck features extracted from the bottleneck layer of DNNs are used to train speaker-dependent model.

4. Experiments

4.1. Experimental Setup

In this study, the bottleneck features for distant-talking speaker identification were extracted from MFCC features. 1-frame of 25-dimensional MFCC features was used as the input of the DNNs. There are 25 hidden units in the bottleneck layer and 500 hidden units in non-bottleneck hidden layer. The number of hidden-layer of DNNs is 5. MFCC features were normalized with mean of the entire training data. The DNN training was carried out using stochastic mini-batch gradient descent with a minibatch size of 100 samples. 50 epochs with a learning rate of 0.1 were used for all layers during pretraining and 1000 epochs with a learning rate of 0.1 were used for all layers during finetuning.

Distant-talking speaker identification was evaluated in artificial reverberant speech for the sake of convenience. Five kinds of multi-channel impulse responses were selected from Real World Computing Partnership (RWCP) sound scene database for test [11] and the CENSREC-4 database for training [12], which were convoluted with clean speech to create artificial reverberant speech. A large scale database, Japanese Newspaper Article Sentence (JNAS) [13] corpus, was used as clean speech. The utterances of training set is composed of 50 male speakers with 10 utterances taken from each speaker. 20 utterances of each speaker were used as test data. The mean length of utterance of the training set was 3.90 sec, and that of the test set was 5.63 sec.

Table 1 lists the impulse responses for train set and test set. Impulse responses were measured at position 2.0 m for the RWCP database and 0.5 m for the CENSREC-4 database from the microphone, respectively.

Table 2 gives the conditions for speaker recognition. 25-dimensional MFCCs and GMMs with 128 mixtures were used. GMMs were trained using one kind of reverberant speech corresponding to one kind of impulse response.

Three kinds of methods were compared in this study. Method 1 is to train the GMM using MFCC and used as a baseline (denoted as ‘MFCC’). Method 2 uses the bottleneck features extracted from the MLP without pretraining (denoted as ‘BF-MLP’). Method 3 uses the bottleneck features extracted from the DNNs with pretraining (denoted as ‘BF-DNNs’).

4.2. Experimental Results

Table 3 show the result of distant-talking speaker identification. In Table 3, BF-DNNs outperformed the MFCC for all conditions, and the average relative error reduction rate was 46.3%. Although there are large variations in the performance of the MFCC in terms of the training set, BF-DNNs based method has relative small variance and it is robust in all reverberant environments. The result of MFCC using training data with 0.60 s reverberation time is not good. The reason may be that the Japanese style bath is a special reverberant environment. On the other hand, the results of BF-DNNs trained by Japanese style bath environment are close to that of other conditions. This indicates that the bottleneck features can potentially transform the feature space of reverberant speech to the feature space of the relative clean speech. Comparing with the BF-DNNs and BF-MLP, BF-DNNs based method is better than BF-MLP based method in average. However, BF-MLP is better than BF-DNNs when reverberant speech with RT 60 of 0.60 s or 0.75 s is using as training data. Therefore, it is necessary to deal with the reverberant speech with long reverberation time for effective pretraining. It is considered that applying dereverberation before bottleneck feature extraction-based transformation may improve the overall performance.

A linear combination of the likelihood of MFCC and BF-DNNs was also evaluated. The weights of MFCC likelihood and BF-DNNs are 0.2 and 0.8, respectively, which are determined empirically. The results are shown in Table 4. In Table 4, the combination method was better than all of the individual methods.

5. Conclusions and Future Work

In this paper, we proposed a robust distant-talking speaker identification method by using bottleneck features. Bottleneck features extracted from MLP and DNNs were used to train the GMM for speaker identification. The proposed bottleneck features of DNNs achieved a relative error reduction of 46.3% compared with conventional MFCC. Comparing with
Table 4: Distant-talking speaker identification rates using combined score of MFCC and BF-DNNs (%).

<table>
<thead>
<tr>
<th>RT60 for training (s)</th>
<th>MFCC</th>
<th>BF-DNNs</th>
<th>Combination</th>
</tr>
</thead>
<tbody>
<tr>
<td>(a) RT 60 for test = 0.47 s</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>0.40</td>
<td>90.4</td>
<td>92.9</td>
<td>93.7</td>
</tr>
<tr>
<td>0.60</td>
<td>76.5</td>
<td>91.1</td>
<td>91.3</td>
</tr>
<tr>
<td>0.75</td>
<td>87.3</td>
<td>92.8</td>
<td>93.5</td>
</tr>
<tr>
<td>Ave.</td>
<td>84.7</td>
<td>92.3</td>
<td>92.8</td>
</tr>
<tr>
<td>(b) RT 60 for test = 1.30 s</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>0.40</td>
<td>89.3</td>
<td>90.6</td>
<td>92.9</td>
</tr>
<tr>
<td>0.60</td>
<td>81.6</td>
<td>90.6</td>
<td>91.4</td>
</tr>
<tr>
<td>0.75</td>
<td>85.1</td>
<td>93.3</td>
<td>94.7</td>
</tr>
<tr>
<td>Ave.</td>
<td>85.3</td>
<td>91.6</td>
<td>93.0</td>
</tr>
</tbody>
</table>

MLP without pretraining and DNNs with pretraining, the results showed that the pretraining was effective for distant-talking speaker identification. Moreover, the speaker recognition performance was furthermore improved by combining the likelihoods of bottleneck features and MFCC.

In this study, we did not use the benefits of DNNs ability to deal with context of multiple frames which is expected to reduce the influence of the reverberation caused over multiple frames. In the future, we will use multiple frames of features as input data of multi-layer network. In addition, so far we only used a single reverberant environment in the training set to train speaker model. We will try to train a reverberant model with multiple reverberant environments in the training set.

6. References