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TOWARDS A NEW SPEECH EVENT DETECTION APPROACH FOR LANDMARK-BASED SPEECH RECOGNITION

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ABSTRACT

In this work, we present a new approach for the classification and detection of speech units for the use in landmark or event-based speech recognition systems. We use segmentation to model any time-variable speech unit by a fixed-dimensional observation vector, in order to train a committee of boosted decision stumps on labeled training data. Given an unknown speech signal, the presence of a desired speech unit is estimated by searching for each time frame the corresponding segment, that provides the maximum classification score. This approach improves the accuracy of a phoneme classification task by 1.7%, compared to classification using HMMs. Applying this approach to the detection of broad phonetic landmarks inside a landmark-driven HMM-based speech recognizer significantly improves speech recognition.

Index Terms—speech event detection, landmark-driven ASR

1. INTRODUCTION

In state-of-the-art hidden Markov model-based (HMM) automatic speech recognition (ASR), speech is modeled as a sequence of phone-segments, often referred to as the beads-on-a-string model of speech [1]. In contrast to that, acoustic-phonetic or event-based approaches to speech recognition model speech as a stream of asynchronous phonetic events, which have to be further processed to obtain a higher level speech representation [2, 3, 4, 5]. Many of these approaches require the detection of phonetic events as time instances, referred to as landmarks.

There are numerous studies concerning the detection of phonetic events in the speech signal, which can be divided into two general groups. The first group uses expert knowledge to derive detection rules from various signal representations (e.g., [2]). The second group uses a classification and detection approach, where labeled data is converted into the desired phonetic feature representation and classifiers are trained to map acoustic observations onto phonetic speech units (e.g., [3, 4]). Employing these classifiers sequentially on the speech signal results in detection functions indicating the presence of the desired speech units. These classifiers operate usually on a frame basis, extracting a fixed-dimensional observation vector for each frame, eventually considering a small context window. On the one hand, this fixed observation space enables the use of non-linear and non-parametric classifiers like support vector machines [3, 4] or multilayer perceptrons for classification, which possess good discrimination abilities and can be trained efficiently. On the other hand, predicting speech units by observing a fixed-size fraction of the speech signal does only partly model time-variable speech units, including phonemes and many articulatory feature units.

We are interested in the problem of extracting a fixed-dimensional observation vector from time-variable speech units, to discriminate them using a classifier that requires a fixed number of observations. To obtain this observation vector, we use a maximum-likelihood segmentation method, to force each labeled speech unit into three spectrally homogeneous parts from which the observation vector is extracted. In our work, this observation vector is used to train speech units with boosted decision trees and to predict unknown segments during testing. To obtain a detection function, indicating the locations of a speech event by its local maxima, we search for each frame the segment that maximizes the classification score at this frame. This method is applied to a recently proposed landmark-driven ASR framework, that uses phonetic landmarks to guide the search in an HMM-based ASR system [6].

The paper is organized as follows. First, we present the details of our proposed method and review boosting with an ensemble of weak learners, as well as landmark-driven ASR. Evaluation is carried out on a phoneme classification task and on landmark-driven speech recognition experiments and we conclude with an outlook on future work.

2. PROPOSED SYSTEM FOR SPEECH EVENT DETECTION

In event-based speech recognition, phonetic speech units are often trained and classified at the frame level. The classification scores of successively predicted frames correspond to a
detect parameters, with the observation space. Since we reduce each speech segment to a fixed-dimensional observation vector, we can overcome the loss in information by post-processing noisy and erroneous detection profiles. If the detection function is obtained by inaccurate modeling of the desired speech event, it is difficult to recover the loss in information by post-processing noisy and erroneous detection profiles.

In the following, we present a method that overcomes these shortcomings by extracting a fixed-dimensional observation vector \( x \) from a given time-variable sequence of parametrized speech \( Y = y_1, \ldots, y_L \) (section 2.1), corresponding to any desired speech unit. We are able to keep a fixed-dimensional observation space, since we reduce every speech segment to three subsegments, similar to a three-state HMM. This observation vector enables the use of any desired classification method, while generalizing all time-variable speech units. In our case, we apply a committee of boosted decision stumps to train a classifier \( F(x) \) (section 2.2). \( F(x) \) is then used to generate a detection function \( d_k = d_{k,1}, \ldots, d_{k,t} \) for \( k \) desired speech units, with the profile of \( d_k \) indicating the presence of the speech unit in the unknown signal. For each \( k \)-th detection function, a set of \( t_k \) time instances \( L_k = \{ t_1, \ldots, t_{t_k} \} \) is extracted, representing the exact locations of the speech event associated with speech unit \( k \) (section 2.3).

### 2.2. Boosting for speech unit classification

Having extracted a fixed-dimensional observation vector for each variable-length speech unit, speech unit models can be trained using the desired classifier. In this work, we boost decision stumps with the multiclass AdaBoost.MH algorithm [8].

Boosting is an ensemble learning method that iteratively learns weak learners \( h_m(x) \) to classify instances, that were for the most part misclassified by the previous learner. The error of learner \( h_m(x) \) determines its weight \( \alpha_m \). The outputs of \( m \) weak learners \( h(x) \) are combined to a strong classifier \( H(x) \), providing a score for every class \( k \) with \( H(x) = \sum_m \alpha_m h_m(x) \).

A set of weak learners provides a very flexible framework during prediction, since the trade-off between computation time and precision can be adapted by changing the number of weak learners. Furthermore, the use of decision trees allows to insert nominal attributes in the future, like information about gender or bandwidth.

The training data used in this work includes over 200 hours of speech, or about 8.5 million labeled speech units. Training on the entire data set is usually not feasible, since most algorithms need the data to fit into memory and training time increases non-linearly. We follow the approach proposed in [9], which is to divide our training set into \( J \) partitions and
to train a classifier \( H_j(x) \) on each partition. The final classification consists in averaging the output scores of the full committee of classifiers:

\[
F(x) = \frac{\sum_{j=1}^{J} H_j(x)}{J}.
\]

(2)

2.3. From classification to detection

As stated in the introduction of section 2, \( F(x) \) is used to obtain a detection function \( d_k = d_{k,1}, \ldots, d_{k,t} \). If \( F(x) \) is trained to predict single frames \( y_t \) of parametrized speech, \( d_{k,t} \) directly corresponds to the prediction score of class \( k \) at frame \( t \). Since our classifier provides segment and not frame-based prediction scores, we provide a new method to generate detection profiles from a collection of predicted segments.

Intuitively, in order to obtain \( d_{k,t} \), one has to search for the segment containing \( t \), that maximizes the likelihood of the given speech unit at frame \( t \). Given a collection of overlapping segments, with each segment corresponding to a hypothetical speech unit, we can directly compare the score of variable-length segments, using our proposed observation-vector. A collection of segments does not correspond to a graph, so that single segments might not have a connection to preceding or subsequent segments (see simplified example in Figure 2). With this work being a preliminary study, we do not consider the task of finding a collection of suitable prior segments, which is desirable to reduce the computational costs for generating \( d_{k,t} \). Instead, we use an exhaustive collection of segments by predicting the observations extracted from all possible segments inside the speech signal, up to a maximum segment length of 300ms. Using fast classification like boosting, and an effective implementation for the solution of equation 1, this stays computationally feasible.

The value of the detection function \( d_{k,t} \) at \( t \) is calculated as follows:

\[
d_{k,t} = \max \{ F_k(x_{i,s,e}) \}; \quad s \leq t \leq e, e - s \geq i - 1.
\]

(3)

\( F_k(x) \) is the score of the \( k \)-th detection function given the observation \( x \). \( s \) and \( e \) correspond to the first and last frame of a segment and \( x_{i,s,e} \) to the observation vector extracted from this segment. The minimum number of frames in a segment is \( i = 3 \).

The obtained detection function indicates the positions of the \( k \)-th speech event in the speech signal by its local maxima. Since we put most effort into the accurate modeling of the speech units, we expect clear indications of the speech events, without requiring sophisticated post-processing. The following simple peak picking algorithm can be applied to a system of discrete speech units, like phonemes or broad phonetic classes (BPCs). Given the detection functions for \( k \) speech units, we extract landmarks in three steps:

1. We first collect the time instances corresponding to local maxima for each detection function \( d_k \) and associate each local maximum \( t \) with its magnitude \( d_{k,t} \). Since a local maximum corresponds to a segment of at least three frames, the central frame \( t \) of a segment is always selected as the time instance \( t \) of the local maximum.

2. If several units \( k \) share the same local maximum \( t \), only the landmark with the highest magnitude \( d_{k,t} \) is kept at \( t \).

3. Some of the remaining landmarks might correspond to a local maximum at a very low magnitude. Therefore, at each landmark, all units \( k \) with a score \( d_{k,t} \) bigger than the landmark magnitude are also activated at that time instance. Afterwards, each set \( L_k = \{ t_1, \ldots, t_{l_k} \} \) contains the final collection of \( l_k \) discrete time instances signaling the presence of the \( k \)-th speech event.

If a local maximum \( t \) of speech unit \( k \) already corresponds to the highest score \( d_{k,t} \) at that time frame, step 3 will have no effect. If not, this step will merge several speech units into broader units, effectively signaling the possible presence of several units at the same time.

3. LANDMARK-DRIVEN HMM-BASED ASR

We use the presented speech event detection approach to extract broad phonetic landmarks for a recently proposed landmark-driven HMM-based ASR framework [6]. In this framework, broad phonetic landmarks are used to guide the Viterbi decoding of the first pass of a HMM-based speech recognizer, which we briefly summarize in the following.

The Viterbi algorithm searches the best path reaching state \((j, t)\) by computing a score

\[
S(j, t) = \max_i S(i, t-1) + \log(a_{ij}) + \log(p((y)_i(j))) + R(j, t),
\]

(4)

where \( \log(a_{ij}) \) is the transition probability and \( \log(p(y_i(j))) \) corresponds to the acoustic likelihood. \( R(j, t) \) enhances paths through states \( j \) that are compatible with phonetic landmarks by \( R(j, t) = \lambda_{j,t} \cdot R_{max} \). With states corresponding
to triphones, checking the compatibility is trivial, since each state can be directly linked to one BPC. \( \lambda_{j,t} \) is a binary indicator, that becomes 1 if state \( j \) is compatible with the landmark at time \( t \) and 0 if there is no landmark at all. \( R_{\text{max}} \) limits the influence of landmarks onto the overall score and has to be determined experimentally.

4. EXPERIMENTS

Our experiments focus on two objectives. First, we are interested in evaluating whether our proposed approach can capture the acoustic information of speech units and compare the classification performance to existing approaches. Second, the proposed approach will be used to extract broad phonetic landmarks to guide the search of an HMM-ASR system as described in section 3.

4.1. Experimental setup

The speech corpus used in the experiments corresponds to radio broadcast news in the French language from the ESTER2 broadcast transcription evaluation campaign [10]. Our training set consists of over 200 hours of speech, the development set of 3 hours and the test set of 4.5 hours from 4 broadcast shows (radio Africa1, radio Inter, radio RFI, radio TVME). The test set is divided into \( J = 12 \) non-overlapping partitions for a committee of 12 classifiers. Classification experiments are run on the development set, while speech recognition is carried out on the test set. Speech labels are obtained by forced-alignment.

The speech recognizer used for recognition experiments is a two-pass system, with the first pass generating a word graph and the second pass rescoring the previously obtained graph using more complex models. We use the tool bonzaiboost\(^1\) for the training of the ensemble of weak classifiers.

4.2. Phoneme classification

While landmark detection is primarily used for the detection of phonetic events, our proposed method is designed to learn any time-variable speech unit. This includes phonetically motivated units, as well as phonemes. We decided to do classification experiments on a phoneme classification task, since this is a challenging task, considering the fine acoustic distinctions between phonemes. Our French phoneme alphabet consists of 40 phonemes, including 5 non-speech events.

HMM-monophone models serve as the baseline phoneme classifier, because of their warping ability, which makes them perfectly suitable for discriminating time varying speech units. The models employed correspond to monophone 3-state left-to-right HMMs with 64 diagonal-covariance Gaussian components per state. HMMs are trained on the full

\(^1\)developed by Christian Raymond, available at http://bonzaiboost.gforge.inria.fr

\begin{table}[h]
\centering
\begin{tabular}{|c|c|}
\hline
# & classifier & accuracy \\
\hline
1 & boosting, concatenated frames (depth=2) & 55.7 \\
2 & boosting, proposed (depth=1) & 57.8 \\
3 & boosting, proposed (depth=2) & 63.0 \\
4 & boosting, proposed (depth=3) & 62.8 \\
\hline
\end{tabular}
\caption{Classifiers of the phoneme classification task. The classifier is described by its classification method (e.g. boosting) and observation vector (e.g. proposed).}
\end{table}

The training set using the same speech parametrization as the boosted ensembles, which are 39-dimensional MFCC vectors, composed of 13 MFCC coefficients with first and second order derivations. To compare the abilities of the classifiers to capture the acoustic properties of each phoneme, we evaluate on a pure classification task, i.e. all observation vectors were extracted using the known phoneme borders obtained by forced alignment, either for training or predicting the associated phoneme. HMM-classification was performed by force-aligning each model to the known phoneme borders and comparing the obtained likelihoods.

First, we trained four classifiers on only one partition (i.e. 1/12-th) of the training data for comparison, before creating the full committee of 12 classifiers. The first classifier uses the three concatenated frames at the temporal center of each phoneme as observation vector \( \mathbf{x} \) for training and testing (see Figure 4). Training and predicting only the center of each phoneme is supposed to minimize errors due to coarticulation effects. We compare this approach to our proposed method, which we run using weak learners of different depth (classi-
Fig. 3. Detection profiles of the four broad phonetic units vowels, glides, plosives (voiced) and fricatives (voiced) for the French word *aujourd'hui* (uttered during a broadcast news show) and its corresponding broad phonetic annotation. On the left, every frame of the detection function was predicted using classifier 1 (see Table 1). On the right, the proposed method was used to generate the detection score at each frame (using classifier 3). While vowels, plosive and fricative are well indicated for both classifiers, the detection function for classifier 1 is more noisy, especially for the vowels. Glides seem to be more clearly represented on the right side, which can be due to the fact, that glides correspond to slow articulatory movements, which can be difficult to capture by concatenated frames.

For all experiments the number of boosting rounds was limited to 3000. Table 1 displays the classification accuracy on the development set as the percentage of correctly classified phonemes. Using the proposed observation vector for boosting increases the performance by 7.3% (using decision trees with depth=2) compared to training on concatenated frames. While boosting on one partition of the training set does not outperform the accuracy of HMMs, applying the committee of 12 boosted weak ensembles improves by 1.7% compared to HMMs.

### 4.3. Landmark-driven ASR

To use our phoneme classifier for broad phonetic landmark detection, we simply derived a collection of detection functions $d_j$ for $j = 7$ BPCs vowels, nasals, glides, fricatives (voiced and unvoiced) and plosives (voiced and unvoiced) by scoring every frame $d_{j,t}$ with the score $d_{k,t}$ of the phoneme $k$, that has the maximum score among all phonemes of this BPC at frame $t$.

We extracted landmarks as proposed in section 2.3 for classifiers 3 and 6 from Table 1. For classifier 1 we equally extracted landmarks, but using the detection function obtained by predicting each individual frame. Figure 3 compares de-

<table>
<thead>
<tr>
<th></th>
<th>classifier</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>1</td>
</tr>
<tr>
<td>number of landmarks</td>
<td>1,352k</td>
</tr>
<tr>
<td>phoneme errors [%]</td>
<td>67.7</td>
</tr>
<tr>
<td>missed phonemes [%]</td>
<td>4.6</td>
</tr>
<tr>
<td>AVG BPC size [phonemes]</td>
<td>13</td>
</tr>
</tbody>
</table>

Table 2. Statistics of extracted landmarks on the whole development set for three classifiers. AVG BPC size corresponds to the average number of phonemes provided at each landmark.
### Table 3. Speech recognition performance driven by broad phonetic landmarks. Landmarks are extracted using classifier 6.

<table>
<thead>
<tr>
<th>broadcast</th>
<th>WER baseline</th>
<th>WER landmark-driven</th>
</tr>
</thead>
<tbody>
<tr>
<td>Inter</td>
<td>18.7</td>
<td>18.6</td>
</tr>
<tr>
<td>RFI</td>
<td>17.6</td>
<td>17.3</td>
</tr>
<tr>
<td>TVME</td>
<td>24.2</td>
<td>23.8</td>
</tr>
<tr>
<td>Africa1</td>
<td>31.5</td>
<td>30.8</td>
</tr>
</tbody>
</table>

The improvement in word-error-rate (WER) varies from 0.1 (radio Inter) to 0.7 (radio Africa1). The overall WER of the test set was 23.1%, compared to a 23.5% baseline, and a 23.8% WER of the BPC of every frame, no broadcast show of the test set was performed well on the presented classification task, applying other classification techniques might further improve speech unit classification and detection.

### 6. REFERENCES