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<th>Evaluation of a Noise-Robust Multi-Stream Speaker Verification Method Using F0 Information</th>
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<tbody>
<tr>
<td>Authors</td>
<td>Taichi Aasami, Koji Iwano, Sadaoki Furui</td>
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Evaluation of a Noise-Robust Multi-Stream Speaker Verification Method Using $F_0$ Information

Taichi ASAMI$^{*1}$, Nonmember, Koji IWANO$^1$, Member, and Sadaoki FURUI$^1$, Fellow

SUMMARY

We have previously proposed a noise-robust speaker verification method using fundamental frequency ($F_0$) extracted using the Hough transform. The method also incorporates an automatic stream-weight and decision threshold estimation technique. It has been confirmed that the proposed method is effective for white noise at various SNR conditions. This paper evaluates the proposed method in more practical in-car and elevator-hall noise conditions. The paper first describes the noise-robust $F_0$ extraction method and details of our robust speaker verification method using multi-stream HMMs for integrating the extracted $F_0$ and cepstral features. Details of the automatic stream-weight and threshold estimation method for multi-stream speaker verification framework are also explained. This method simultaneously optimizes stream-weights and a decision threshold by combining the linear discriminant analysis (LDA) and the Adaboost technique. Experiments were conducted using Japanese connected digit speech contaminated by white, in-car, or elevator-hall noise at various SNRs. Experimental results show that the $F_0$ features improve the verification performance in various noisy environments, and that our stream-weight and threshold optimization method effectively estimates control parameters so that FARs and FRRs are adjusted to achieve equal error rates (ERRs) under various noisy conditions.

key words: speaker verification, $F_0$ information, multi-stream HMMs, stream-weight and threshold optimization, Adaboost

1. Introduction

Increasing noise-robustness is one of the key issues for constructing real-world speaker verification systems. Since $F_0$ features are less sensitive to channel distortions or additive noise than spectral features, they are expected to be useful for increasing the robustness of speaker recognition. Various methods using fundamental frequency ($F_0$) in combination with spectral features have been proposed to achieve high-performance speaker recognition systems [1]-[9]. Carey et al. proposed a robust speaker recognition method using $F_0$ features to cope with the effect of handset variation on telephone speech [2]. Kyung and Lee showed that $F_0$ features increased robustness of VQ-based speaker identification against additive noise [4]. We have proposed a noise-robust speaker verification method using multi-stream HMMs for integrating $F_0$ and spectral information [10]-[12]. The $F_0$ extraction method and multi-stream HMM construction method used in the verification scheme are based on our previous work on noise-robust speech recognition [12],[13]. The Hough transform [14], a robust image processing technique, was used for reliably extracting $F_0$ values. In [10], we have confirmed that the $F_0$ features yield improvement of speaker verification performance in white noise at various SNR conditions.

In order to construct practical multi-stream speaker verification systems, optimum system parameters, such as stream-weights of multi-stream HMMs and decision thresholds, need to be estimated before the verification process [15],[16]. We have proposed an automatic stream-weights and threshold estimation method for multi-stream speaker verification [17], using the linear discriminant analysis (LDA) [18] and the Adaboost technique [19]. In this method, the stream-weights and the threshold are automatically optimized according to the noise conditions of a development set. Since the optimum threshold of the multi-stream speaker verification is variable according to the setting of stream-weights, the threshold and the stream-weights need to be simultaneously estimated. The threshold is estimated so that the false acceptance rate (FAR) is equal to the false rejection rate (FRR) in the proposed method. In [17], we have confirmed that this method can accurately optimize the stream-weights and the threshold in white noise at various SNR conditions.

In this paper, additional experiments in more practical in-car and elevator-hall noise conditions are conducted for evaluating performance of our proposed multi-stream speaker verification method incorporating an automatic stream-weights and threshold optimization technique.

This paper is organized as follows. Section 2 explains a speaker verification method using multi-stream HMMs, which integrates spectral and noise-robust $F_0$ features. In Sect. 3, an automatic stream-weight and threshold optimization method based on the LDA and the Adaboost is explained. Experimental results are presented in Sect. 4, and Sect. 5 concludes this paper.

2. Speaker Verification Using $F_0$ Information

We have proposed a noise-robust $F_0$ extraction method and an effective method for integrating spectral and $F_0$ information, and effectiveness of these methods has been confirmed for speech recognition [11],[12]. These methods have also been implemented for speaker verification [10]. Details of...
the methods are described below.

2.1 Noise-Robust $F_0$ Extraction Method Using Hough Transform

Cepstral peaks extracted independently for each short period of speech have been widely used to extract $F_0$ values. This method often causes errors, including half pitch, double pitch and drop outs, for noisy speech. Since $F_0$ contours have temporal continuity in voiced periods, the Hough transform [14], taking advantage of its continuity, applied to time-cepstrum images is expected to have robustness in extracting pitch in the noisy environment. The Hough transform is an image processing technique to robustly extract parametric patterns, such as lines, and ellipses, from a noisy image [20].

Speech waveforms are sampled at 16 kHz and transformed to a sequence of 256-dimensional cepstral vector. A 32 ms-long Hamming window is used to extract each frame at every 10 ms. A nine-frame moving window is applied at every frame interval to extract a time-cepstrum image used for line information detection. The time-cepstrum image is used as the pixel brightness image for the Hough transform. An $F_0$ value is obtained from a cepstrum index at the center point of the detected line. Since the moving window has nine frames, time continuity over 90 ms is taken into account in this method. More details of the $F_0$ extraction method is explained in [13].

By using this method, $F_0$ extraction error can be significantly reduced in comparison with the conventional method in which cepstral peaks are chosen at each frame independently [13]. It has also been confirmed that the Hough transform-based $F_0$ extraction method yields better performance than the ESPS get_f0 function in various noise conditions [21]. Although the Hough transform-based method requires 10 times more computational cost than the conventional method, the computational cost can be reduced by adjusting the threshold value to control the number of points to be used for the transformation.

2.2 Noise-Robust Speaker Verification Using Multi-Stream HMMs

2.2.1 Japanese Connected Digit Speech

The proposed method is evaluated using four-connected-digit speech in Japanese. In Japanese connected digit speech, two consecutive digits usually make one prosodic phrase. Figure 1 shows an example of an $F_0$ contour of four-connected-digit speech. The first two digits, “1” and “4”, make the first prosodic phrase, and the latter two digits, “3” and “8”, make the second prosodic phrase. The transition of $F_0$ is represented by CV syllabic units, and each CV syllable can be prosodically labeled as “rising” or “falling”.

![Figure 1](image)

Fig. 1 An example of $F_0$ contour of four-connected-digit speech in Japanese.

2.2.2 Integration of Segmental and Prosodic Features

Each segmental feature vector has 25 elements consisting of 12MFCC (Mel Frequency Cepstral Coefficients), their delta and the delta log energy. The window length is 25 ms and the frame interval is 10 ms. Cepstral mean subtraction (CMS) is applied to each utterance.

Simultaneously, two kinds of prosodic features are extracted: $\log F_0$ and $\Delta \log F_0$. It has been confirmed that speaker verification performance is improved by using both prosodic features in comparison with the method using either of the features [10]. A segmental-prosodic feature vector is built by combining the segmental and prosodic feature vectors at each frame.

2.2.3 Integration of Segmental and Prosodic Models

Since $F_0$ transitions are easily represented by using CV syllabic units in Japanese connected digit speech, syllabic unit HMMs are used for modeling segmental and prosodic features in this study. This method can also be applied to other languages by changing the modeling units depending on the characteristics of the $F_0$ transitions of the target language.

An integrated syllable HMM denoted by "SP-HMM (Segmental-Prosodic HMM)" is modeled by taking both the syllable context and the $F_0$ transitions into account. Each Japanese digit uttered continuously with other digits can be modeled by a concatenation of two-syllable units (morae). Even “2” (/ni/) and “5” (/go/) can be modeled by two syllables, since their final vowel is usually lengthened as /ni:/ and /go:/.

The syllable context is considered only within each digit in our experiment. Therefore, the SP-HMM can be denoted by either a left-context dependent syllable "LC-SYL, PH" or a right-context dependent syllable "SYL+RC, PH", where "PH" indicates an $F_0$ transition pattern which is either rising (U) or falling (D). For example, the first syllable /i/ of “1” (/ichi/) which has rising $F_0$ transition is denoted as “1+chi, U”. Each SP-HMM has a standard left-to-right topology with $n \times 3$ states, where $n$ is the number of

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1The ESPS get_f0 function is one of the standard robust $F_0$ extraction methods. It uses the normalized cross-correlation function and dynamic programming [22].
Table 1 The list of integrated models (SP-HMMs).

<table>
<thead>
<tr>
<th>Digit</th>
<th>Model</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>ze+ro, U ze+ro, D</td>
</tr>
<tr>
<td>/zero/</td>
<td>ze- ro, U ze- ro, D</td>
</tr>
<tr>
<td>1</td>
<td>i+chi, U i+chi, D</td>
</tr>
<tr>
<td>/chi/</td>
<td>i- chi, U i- chi, D</td>
</tr>
<tr>
<td>2</td>
<td>ni+i, U ni+i, D</td>
</tr>
<tr>
<td>/ni/</td>
<td>ni- i, U ni- i, D</td>
</tr>
<tr>
<td>3</td>
<td>sa+K, U sa+K, D</td>
</tr>
<tr>
<td>/sa/N</td>
<td>sa- K, U sa- K, D</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Digit</th>
<th>Model</th>
</tr>
</thead>
<tbody>
<tr>
<td>4</td>
<td>yo+K, U yo+K, D</td>
</tr>
<tr>
<td>/yoN/</td>
<td>yo- K, U yo- K, D</td>
</tr>
<tr>
<td>5</td>
<td>go+o, U go+o, D</td>
</tr>
<tr>
<td>/go/</td>
<td>go- o, U go- o, D</td>
</tr>
<tr>
<td>6</td>
<td>ro+ku, U ro+ku, D</td>
</tr>
<tr>
<td>/ruku/</td>
<td>ro- ku, U ro- ku, D</td>
</tr>
<tr>
<td>7</td>
<td>na+na, U na+na, D</td>
</tr>
<tr>
<td>/na/N</td>
<td>na- na, U na- na, D</td>
</tr>
<tr>
<td>8</td>
<td>ha+chi, U ha+chi, D</td>
</tr>
<tr>
<td>/hachi/</td>
<td>ha- chi, U ha- chi, D</td>
</tr>
<tr>
<td>9</td>
<td>kyu+u, U kyu+u, D</td>
</tr>
<tr>
<td>/kyu/</td>
<td>kyu- u, U kyu- u, D</td>
</tr>
</tbody>
</table>

Fig. 2 Building SP-HMMs using a tied-mixture technique. S-HMMs and P-HMMs are trained using segmental and prosodic features, respectively.

phonemes in the syllable. Table 1 shows a list of integrated models. “sil” denotes a long pause that appears at the beginning and end of an utterance, and “sp” denotes a short pause that appears between digits. The “sil” model has 3 states, and the “sp” model has 1 state.

SP-HMMs are modeled as multi-stream HMMs. In recognition, the log-probability $b_j(O_{sp}^t)$ of generating the $t$-th frame segmental-prosodic observation $O_{sp}^t$ at state $j$ is calculated by:

$$b_j(O_{sp}^t) = \lambda_s b_j(O_s^t) + \lambda_p b_j(O_p^t)$$

(1)

where $b_j(O_s^t)$ is the log-probability of generating a segmental feature vector $O_s^t$, and $b_j(O_p^t)$ is the log-probability of generating a prosodic feature vector $O_p^t$. $\lambda_s$ and $\lambda_p$ are weighting factors for the segmental and prosodic streams respectively.

Syllable HMMs for segmental and prosodic feature vectors are separately made and combined to build SP-HMMs using a tied-mixture technique as follows:

i. “S-HMMs (Segmental HMMs)” are trained by segmental features only. They can be denoted by either “LC-SYL,” “*" or "SYL+RC," “*". Here, “*” (wild card) means that HMMs are built without considering the $F_0$ transitions, “U” and “D”. The total number of S-HMM states is the same as the number of SP-HMM states.

ii. Training utterances are segmented into syllables by the forced-alignment technique using the S-HMMs, and one of the $F_0$ transition labels, “U” or “D”, is given to each segment according to the actual $F_0$ pattern.

iii. “P-HMMs (Prosodic HMMs)” are trained by prosodic feature vectors within these segments, according to the $F_0$ transition label. Six separate models, “*", “U", “*", “D", “*", “D", “sil", and “sp” are made. Each P-HMM has a single state.

iv. The S-HMMs and P-HMMs are combined to make SP-HMMs. Gaussian mixtures in the segmental stream of SP-HMMs are tied with corresponding S-HMM mixtures, while the mixtures in the prosodic stream are tied with corresponding P-HMM mixtures. Figure 2 shows the integration process. In this example, the mixtures of SP-HMM “i+chi, U" are tied with S-HMM “i+chi, *" and P-HMM “*", “U".

2.2.4 Verification Score

A verification score after observing a feature set $O$ is denoted by $q(O)$, which is calculated as

$$q(O) = l(O|C) - l(O|G)$$

(2)

where $l(O|C)$ is a frame-averaged log-likelihood value with claimed speaker’s SP-HMM $C$ and $l(O|G)$ is a frame-averaged log-likelihood value with general speaker’s SP-HMM $G$. The likelihood values are calculated under the assumption that each speaker can utter arbitrary four connected digits. This method can be extended to a text-prompted speaker verification system, in which a sequence of four digits to be spoken is prompted by the system and verification decision is made by combining a recognized
word sequence and voice similarity to the claimed speaker.

The log-likelihood values for a segmental-prosodic feature vector $O_{sp}$ are defined using Eq. (1) as follows:

\[ l(O_{sp}|C) = \lambda_s l(O_s|C) + \lambda_p l(O_p|C), \]
\[ l(O_{sp}|G) = \lambda_s l(O_s|G) + \lambda_p l(O_p|G). \]  

Then, the verification score $q(O_{sp})$ is calculated as

\[ q(O_{sp}) = \lambda_s q(O_s) + \lambda_p q(O_p). \]  

(5)

If the score is larger than a threshold value $\theta$, the speaker is accepted as the claimed speaker. Therefore, the discriminant function is $z = q(O_{sp}) - \theta$. If $z$ is positive, the speaker is accepted, and if it is less than or equal to 0, the speaker is rejected as being an imposter. In the experiments in this paper, the stream-weights are constrained by

\[ \lambda_s + \lambda_p = 1. \]  

(6)

3. Automatic Stream-Weight and Threshold Optimization Methods

In real-world applications, the stream-weights $\lambda_s$, $\lambda_p$ and the decision threshold $\theta$ parameters need to be estimated before verification\[15,16\]. In this section, our proposed method for automatically optimizing these parameters using the LDA and the Adaboost techniques\[17\] is explained.

3.1 Estimation by the LDA

As described in the previous section, speaker verification by SP-HMM uses the following discriminant function $z$:

\[ z = q(O_{sp}) - \theta = \lambda_s q(O_s) + \lambda_p q(O_p) - \theta. \]  

(7)

(8)

Since $z$ is a linear function, the stream-weights and the threshold can be estimated as coefficients of a linear function obtained by the LDA. The estimation process is as follows. First, segmental and prosodic scores, $q(O_s)$ and $q(O_p)$, calculated from both claimed speaker's and imposter's data included in the development set are plotted in a two-dimensional space composed of $q(O_s)$ and $q(O_p)$. Then, the LDA is applied to the space so as to obtain the discriminant function $z$ which distinguishes score distribution of claimed speakers from that of imposters.

Since the obtained function $z = a_s q(O_s) + a_p q(O_p) - b$ does not satisfy $a_s + a_p = 1$, it is transformed so that the sum of the coefficients becomes 1 according to the constraint Eq. (6). The estimated values of the stream-weights and the threshold are

\[ \tilde{\lambda}_s = \frac{a_s}{a_s + a_p}, \quad \tilde{\lambda}_p = \frac{a_p}{a_s + a_p}, \quad \tilde{\theta} = \frac{b}{a_s + a_p}. \]  

(9)

Thus, all the parameters are estimated according to the LDA criterion which maximizes discrimination performance between claimed speakers and imposters.

In the boosting process described in the next subsection, multiple discriminant functions obtained from boosting iterations are integrated. The normalization by Eq. (9) is necessary for correctly weighting these discriminant functions in the integration process.

There is no guarantee that the denominator $a_s + a_p$ does not become 0, and if it becomes 0, some special processing such as flooring needs to be applied. However, almost all the actual $a_s$ and $a_p$ values estimated in Sect. 4.5 and 4.6 were positive, and only a few negative values were observed in several low SNR conditions. Hence, practically $a_s + a_p$ never became 0.

3.2 Optimization by the Adaboost

The Adaboost, a class of boosting algorithms, constructs a high performance classifier by sequentially combining trained simple classifiers\[19\]. In our optimization method, the linear discriminant functions obtained by the LDA are used as simple classifiers for the Adaboost. By doing so, we can estimate more accurate weights and thresholds than those obtained by only using the LDA. In this paper, the stream-weights and the threshold are optimized to achieve equal error rates (EERs) at which FAR is equal to FRR.

Details of the optimization algorithm are as follows, where $n$ represents the number of data in the development set and $K$ represents the number of iterations. Let $(x_i) (i = 1, \cdots, n)$ be the development data plotted into the two-dimensional space composed of $q(O_s)$ and $q(O_p)$, and $(w_i^{(k)}) (i = 1, \cdots, n, \text{and } k = 1, \cdots, K)$ be the weights of each data in $k$-th iteration.

i. Initialize the weights of data $w_1^{(1)} = 1/n$.

ii. Iterate the following processes for $k = 1, \cdots, K$.

a. Choose $n$ samples from $(x_i)$ allowing duplications, using $(w_i^{(k)})$ as a probability distribution to make the re-sampled data $(x_{i}^{(k)})$.

b. Obtain a linear discriminant function

\[ z_k = \lambda_s^{(k)} q(O_s) + \lambda_p^{(k)} q(O_p) - \theta^{(k)} + \delta_{k-1} \]  

(10)

by applying the LDA to the re-sampled data $(x_{i}^{(k)})$, where $\delta_{k-1}$ is an offset value for adjusting FAR and FRR to EER. The offset value is obtained from the $(k - 1)$-th boosting iteration, where the initial value $\delta_0$ is set at 0.

c. Classify all the data in the development set $(x_{i})$ using $z_k$, and calculate FAR, FRR, and the weighted discriminant errors $e_s$, $e_p$, and $e_{FR}$:

\[ \text{FAR}_k = \frac{\sum_{i:x_{i}s > \theta^{(k)}} 1}{\sum_{i} 1}, \] 
\[ \text{FRR}_k = \frac{\sum_{i:x_{i}p > \theta^{(k)}} 1}{\sum_{i} 1}, \] 
\[ e_k = \sum_{i} w_i^{(k)}, \]

(11) (12) (13)


where \( \sum_{i_{FA}} \) and \( \sum_{i_{FR}} \) are the summations for all \( i \) with the results of FA and FR, respectively, and \( \sum_{i_{err}} \) is the summation for all \( i \) with the classification error, FA or FR, in the classification of \( \{x_i\} \). \( \sum_{i_{err}} \) and \( \sum_{i_{FA}} \) are the summations for all \( i \) belonging to imposters and claimed speakers, respectively. The offset value \( \delta_k \) is determined by the following equation:

\[
\delta_k = \alpha \cdot \frac{1 - \text{FAR}_k/\text{FRR}_k}{1 + \text{FAR}_k/\text{FRR}_k},
\]

where \( \alpha \) is a scaling factor of the offset value.

d. Calculate \( u_k \) as the weight of \( z_k \) by the following equations:

\[
u_k = c(e_k) \cdot c(d_k),
\]

\[
d_k = \frac{|\text{FAR}_k - \text{FRR}_k|}{2},
\]

where the cost function \( c(x) \) is defined by the following equation:

\[
c(x) = \frac{1}{2} \log \frac{1 - x}{x}.
\]

e. Update \( \{w_i^{(k)}\} \) by the following formula:

\[
w_i^{(k+1)} = \begin{cases} w_i^{(k)} e^{-\alpha \text{err}_k} & (\text{classify } x_i \text{ accurately}) \\ w_i^{(k)} e^{\alpha \text{err}_k} & (\text{misclassify } x_i) \end{cases}
\]

where \( \text{err}_k \) is a weighted error rate in \( k \)-th iteration calculated as follows:

\[
\text{err}_k = \frac{\text{FAR}_k + \text{FRR}_k}{2}.
\]

f. Normalize \( \{w_i^{(k+1)}\} \) to meet \( \sum_{i=1}^{n} w_i^{(k+1)} = 1 \).

iii. Let the conclusive classifier \( z \) be the weighted majority vote of \( z_k \):

\[
z = \sum_{k=1}^{K} (u_k z_k).
\]

iv. Normalize the coefficients of \( z \) so that the sum of them becomes 1.

v. Set the normalized coefficients as the estimated streamweights and the threshold.

In the original Adaboost algorithm, the conclusive classifier \( z \) is defined by \( z = \sum_{k=1}^{K} (u_k z_k) \). However, this cannot be directly used for streamweight estimation, since its form is not a linear discriminant function. Thus, we approximate \( z \) by \( z = \sum_{k=1}^{K} (u_k z_k) \) as shown in Eq. (22).

---

4. Experiments

4.1 Experimental Conditions

Speech data were recorded in 5 sessions separated by intervals of approximately one month. The data were collected from 36 male speakers and sampled at 16 kHz with a 16 bit resolution. Each speaker uttered 50 four-connected-digit strings in Japanese at each session. Each digit appeared the same number of times in this set of 50 randomly arranged four digit strings.

For each speaker, 150 strings recorded at sessions 1 ~ 3 were used for training and 100 strings recorded at sessions 4 and 5 were used for testing.

The database was separated into three groups in terms of speakers as shown in Fig. 3. The figure shows the case where speaker #01 was used as the claimed speaker. The general speaker’s model was trained using utterances by all the speakers in the speaker group 2, which did not include the claimed speaker, and the stream-weights and the threshold were optimized using the data of speaker group 3. Each speaker model of speaker group 3 was trained using utterances from sessions 1 ~ 3 for the parameter optimization. Utterances from sessions 4 and 5 of group 3 were used as estimation data for the parameter optimization. Another experiment was conducted in which the general speaker’s model was trained by the data of speaker group 3 and the weights and the threshold were optimized by the data of speaker group 2. Next, group 1 & 3 are used for training and development for testing group 2, and finally group 1 & 2 are used for training and development for testing group 3. There are six combinations of the training set, the development set and the testing set. The result averaged over the six experiments was used for evaluation.

White noise was added to the training set at a 30 dB SNR level to increase robustness against noisy speech, and the development and testing sets were contaminated with
white, in-car or elevator-hall noise at 5, 10, 15, 20 and 30 dB SNR conditions. The latter two noises are included in the noise database distributed by the Japan Electronic Industry Development Association (JEIDA) [23]. The development set contaminated with the same noise at the same SNR as the testing set was used for automatically optimizing the stream-weights and the decision threshold.

In our preliminary experiments using S-HMMs with a white noise SNR of 30 dB, the most accurate verification performance was achieved using S-HMMs with four mixtures. Thus four mixture S-HMMs were also used in the following experiments. Each mixture component for all the HMMs (S-HMMs, P-HMMs, and SP-HMMs) was modeled using a diagonal-covariance Gaussian distribution.

4.2 Effectiveness of Multi-Stream HMMs

We first investigated the effectiveness of using multi-stream HMMs to integrate segmental and prosodic information. A two-dimensional prosodic feature vector, consisting of $\log F_0$ and $\Delta \log F_0$ extracted by the Hough transform, was used for this experiment.

The EERs of the following two cases were compared:

(a) The case where the $q_{sp}(O_{sp})$ obtained from SP-HMMs is used as the verification score.

(b) The case where $\omega_s q_s(O_s) + \omega_p q_p(O_p)$ which is the weighted sum of $q_s(O_s)$ and $q_p(O_p)$ obtained separately from S-HMMs and P-HMMs is used as the verification score.

The subscript $m$ of $q_m$ represents the model from which the score is calculated, S-HMMs, P-HMMs or SP-HMMs. In this experiment, $m$ was either $s$, $p$ or $sp$. (a) and (b) in Fig. 4 show the flow charts of the speaker verification process in each case. The integration weights, $\omega_s$ and $\omega_p$, and the stream-weights, $\lambda_s$ and $\lambda_p$, were manually optimized in the 0 to 1 range using the testing set ex post facto for each noise condition. The number of mixtures in prosodic models was optimized for each case at the white noise 30 dB SNR condition; the best number of mixtures was four in both cases.

The results in the white noise, in-car noise and elevator-hall noise conditions are shown in Table 2. The symbol "**" indicates that the EER obtained by the feature level integration implemented by SP-HMMs is significantly better at 1% significance level than that obtained when the scores obtained separately from S-HMMs and P-HMMs are integrated afterward.

These results show that the speaker verification method using multi-stream HMMs for integrating segmental and prosodic information is effective in most of the noise conditions, except for a small number of conditions where a minor degradation is observed. Our previous research on speech recognition [12] showed that using multi-stream HMMs which integrated segmental and prosodic features yielded better time alignment of digits in noisy conditions. Probably this is also the reason why EERs are decreased by using SP-HMMs in speaker verification.

![Flow chart of speaker verification process](image)

**Table 2** Comparison of the EERs (%) of the case where the S-HMMs and P-HMMs are separately used with the case where the SP-HMMs are used.

<table>
<thead>
<tr>
<th>Noise</th>
<th>SNR</th>
<th>S-HMM &amp; P-HMM</th>
<th>SP-HMM</th>
</tr>
</thead>
<tbody>
<tr>
<td>white</td>
<td>30dB</td>
<td>0.66</td>
<td>0.63</td>
</tr>
<tr>
<td></td>
<td>20dB</td>
<td>3.42</td>
<td>3.28</td>
</tr>
<tr>
<td></td>
<td>15dB</td>
<td>9.40</td>
<td>8.74**</td>
</tr>
<tr>
<td></td>
<td>10dB</td>
<td>17.6</td>
<td>16.1**</td>
</tr>
<tr>
<td></td>
<td>5dB</td>
<td>23.8</td>
<td>23.5</td>
</tr>
<tr>
<td>in-car noise</td>
<td>30dB</td>
<td>11.3</td>
<td>10.5**</td>
</tr>
<tr>
<td></td>
<td>20dB</td>
<td>11.1</td>
<td>10.9</td>
</tr>
<tr>
<td></td>
<td>15dB</td>
<td>11.2</td>
<td>11.2</td>
</tr>
<tr>
<td></td>
<td>10dB</td>
<td>11.7</td>
<td>11.9</td>
</tr>
<tr>
<td></td>
<td>5dB</td>
<td>13.3</td>
<td>13.6</td>
</tr>
<tr>
<td>elevator-hall</td>
<td>30dB</td>
<td>3.71</td>
<td>3.61</td>
</tr>
<tr>
<td>noise</td>
<td>20dB</td>
<td>5.07</td>
<td>5.10</td>
</tr>
<tr>
<td></td>
<td>15dB</td>
<td>9.86</td>
<td>9.60</td>
</tr>
<tr>
<td></td>
<td>10dB</td>
<td>19.7</td>
<td>18.7**</td>
</tr>
<tr>
<td></td>
<td>5dB</td>
<td>28.9</td>
<td>28.5</td>
</tr>
</tbody>
</table>
Table 3  Comparison of the EERs when using the prosodic feature vector with/without the Hough transform.

<table>
<thead>
<tr>
<th>Noise</th>
<th>SNR</th>
<th>S-HMM</th>
<th>SP-HMM (NH-LD)</th>
<th>SP-HMM (H-LD)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>30dB</td>
<td>0.88</td>
<td>0.63</td>
<td>0.63</td>
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<tr>
<td>white</td>
<td>20dB</td>
<td>4.91</td>
<td>3.77</td>
<td>3.28**</td>
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<tr>
<td>noise</td>
<td>15dB</td>
<td>14.7</td>
<td>10.9</td>
<td>8.74**</td>
</tr>
<tr>
<td></td>
<td>10dB</td>
<td>37.1</td>
<td>20.9</td>
<td>16.1**</td>
</tr>
<tr>
<td></td>
<td>5dB</td>
<td>37.5</td>
<td>31.1</td>
<td>23.5**</td>
</tr>
<tr>
<td>in-car</td>
<td>30dB</td>
<td>15.9</td>
<td>12.4</td>
<td>10.5**</td>
</tr>
<tr>
<td>noise</td>
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<td>15.9</td>
<td>12.3</td>
<td>10.9**</td>
</tr>
<tr>
<td></td>
<td>15dB</td>
<td>15.7</td>
<td>12.3</td>
<td>11.2**</td>
</tr>
<tr>
<td></td>
<td>10dB</td>
<td>16.0</td>
<td>13.0</td>
<td>11.9**</td>
</tr>
<tr>
<td></td>
<td>5dB</td>
<td>18.4</td>
<td>15.1</td>
<td>13.6**</td>
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<tr>
<td>elevator-hall noise</td>
<td>30dB</td>
<td>4.48</td>
<td>3.69</td>
<td>3.61</td>
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<td></td>
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<td>5.79</td>
<td>5.02</td>
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<td></td>
<td>15dB</td>
<td>11.7</td>
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</tr>
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<td></td>
<td>10dB</td>
<td>24.8</td>
<td>19.9</td>
<td>18.7**</td>
</tr>
<tr>
<td></td>
<td>5dB</td>
<td>40.2</td>
<td>32.3</td>
<td>28.5**</td>
</tr>
</tbody>
</table>

Fig. 5  The EERs as a function of the prosodic stream weight ($\lambda_p$) at each SNR in the white noise environment.

determine the offset value $\delta_k$ were experimentally set to 200 and 0.005, respectively.

Table 4 shows decision costs $C = 0.5 \cdot FAR + 0.5 \cdot FRR$ obtained by using the proposed optimization method in each noise condition. Since the target of the FAR and the FRR is an EER in this experiment, the weights of both the FAR and the FRR are 0.5. The results in “LDA only” indicate that using the stream-weights and the threshold estimated by only the LDA; and, the results in “AdaBoost” were obtained by the proposed optimization method combining the boosting technique. The bottom line shows the target decision costs which are obtained when the stream-weights and the threshold were manually optimized for the testing set ex post facto so that the EERs were minimized.

The results show that the AdaBoost-based optimization method outperforms the method using only the LDA in most of the SNR conditions with white and elevator-hall noise. In an in-car noise environment, since the method using only the LDA is effective enough in adjusting the FARs and FRRs to the target EERs, the AdaBoost-based method does not further improve them.

4.6  Effects of the Number of the AdaBoost Iterations

Figures 6 (a), (b), and (c) show FARs and FRRs for the development set as a function of the number of AdaBoost iterations\(^1\) at 15 dB SNR condition in white, in-car, and elevator-hall noise environment, respectively.

FARs and FRRs smoothly converges to the EERs. These results indicate that the proposed method is stable over a variable number of AdaBoost iterations.

5. Conclusions

This paper has evaluated a speaker verification method using multi-stream HMMs which combine segmental and prosodic features in various noise conditions. This method uses an automatic stream-weight and threshold optimization method based on the LDA and the AdaBoost approaches.

\(^1\)They have been denoted by “FAR\(_q\)” and “FRR\(_q\)” in Eqs. (11) and (12).
The prosodic features are extracted by an $F_0$ feature extraction technique based on the Hough transform. Experimental results using Japanese connected digit utterances show that: 1) the Hough transform is effective for increasing robustness in extracting the $F_0$ features at various noise conditions; 2) the multi-stream verification method is robust against the change of the stream weight; 3) the Adaboost-based parameter optimization method is effective in white and elevator-hall noise conditions; and 4) the optimization method is stable over the variation of the number of Adaboost iterations.

Our future work include: 1) investigating prosodic features other than $F_0$-based features, such as durations; 2) improving the SP-HMM topology; 3) using voiced/unvoiced information; 4) improving the stream-weight and threshold optimization algorithm so that FAR-FRR ratio can be freely adjusted; 5) evaluating performance when applying the proposed optimization method to multi-stream speaker verification systems using a larger number of streams; 6) investigating a parameter optimization method using a testing set without having labelled speaker IDs, instead of using a development set; 7) generalizing the proposed method so that it can be applied to speaker verification tasks using arbitrary words or sentences; and 8) evaluating performance of the proposed method on real-world data recorded under actual noisy environment.

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