A Segment-Based Speaker Verification System Using *SUMMIT*

by

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B.S., Cornell University, 1994

Submitted to the Department of Electrical Engineering
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Abstract

This thesis describes the development of a segment-based speaker verification system. Our investigation is motivated by past observations that speaker-specific cues may manifest themselves differently depending on the manner of articulation of the phonemes. By treating the speech signal as a concatenation of phone-sized units, one may be able to capitalize on measurements for such units more readily. A potential side benefit of such an approach is that one may be able to achieve good performance with unit (i.e., phonetic inventory) and feature sizes that are smaller than what would normally be required for a frame-based system, thus deriving the benefit of reduced computation.

To carry out our investigation, we started with the segment-based speech recognition system developed in our group called SUMMIT [43], and modified it to suit our needs. The speech signal was first transformed into a hierarchical segment network using frame-based measurements. Next, acoustic models for each speaker were developed for a small set of six phoneme broad classes. The models represented feature statistics with diagonal Gaussians, which characterized the principle components of the feature set. The feature vector included averages of MFCCs, plus three prosodic measurements: energy, fundamental frequency (F0), and duration. The size and content of the feature vector were determined through a greedy algorithm optimized on overall speaker verification performance.

To facilitate a comparison with previously reported work [19, 2], our speaker verification experiments were carried out using 168 speakers from the TIMIT corpus. Each speaker-specific model was developed from the eight SI and SX sentences. Verification was performed using the two SA sentences common to all speakers. To classify a speaker, a Viterbi forced alignment was determined for each test utterance, and the forced alignment score of the purported speaker was compared with those obtained with the models of the speaker's competitors. Ideally, the purported speaker's score should be compared to scores of every other system user. To reduce the computation, we adopted a procedure in which the score for the purported speaker is compared only to scores of a cohort set consisting of a small set of acoustically similar speakers. These scores were then rank ordered and the user was accepted if his/her model's score was within the top N scores, where N is a parameter we varied in our experiments. To test for false acceptance, we used only the members of a speaker's cohort set as impostors. We have found this method to significantly reduce computation while minimally affecting overall performance.

We were able to achieve a performance of 0% false rejection of true users and 4.85% false acceptance of impostors, with a simple system design. We reduced computation significantly through the use of a small number of features representing broad-classes, diagonal Gaussian speaker models, and using only cohort sets during testing.

Thesis Supervisor: Victor W. Zue

Title: Senior Research Scientist

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Chapter 1

Speaker Verification

1.1 Introduction

Speaker verification involves the task of automatically verifying a person's identity by his/her speech through the use of a computer. The outcome of speaker verification is a binary decision as to whether or not the incoming voice belongs to the purported speaker. Speaker verification has been pursued actively by researchers, because it is presently a palpable task with many uses that involve security access authorizations. In the past, applications for speaker verification systems mainly involved physical access control, automatic telephone transaction control (*e.g.*, bank-by-phone), and computer data access control. However, due to the revolution in telecommunications, uses for speaker verification systems also include Internet access control, and cellular telephone authorizations.

Figure 1-1 illustrates the basic components of a speaker verification system. The feature extraction component attempts to capture acoustic measurements from the user's speech signal that are relevant to inter-speaker differences. During training, the acoustic features are used to build speaker-specific models. During testing, measurements extracted from the test data are scored against the stored speaker models to see how well the test data match the reference models. The speaker is accepted or rejected based on this score. Of course, many details are left out of the block diagram, such as the *type of text* the system prompts, the *features* the system extracts, and

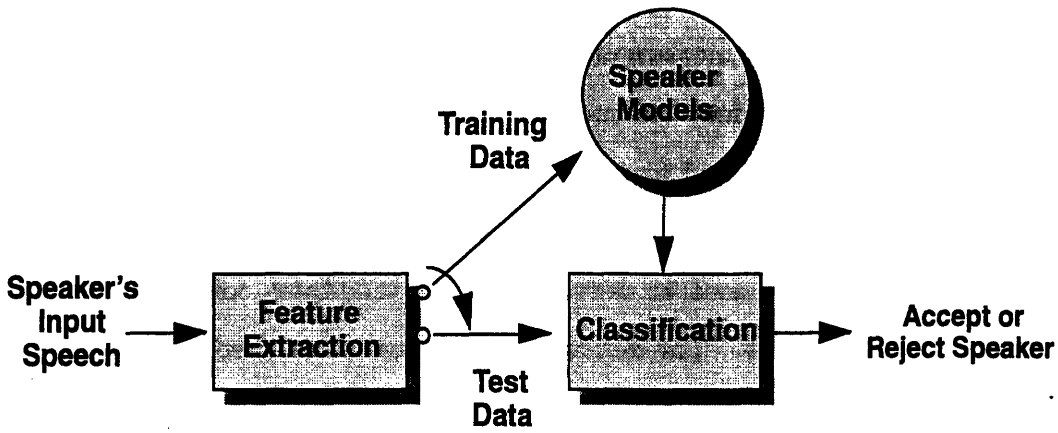

Figure 1-1: General Speaker Verification System

the *speaker models* and *classifiers* the system implements. For detailed tutorials on speaker verification, refer to [27, 6].

1.2 Previous Research

Research in speaker verification has been active for many years. In this section, we describe general approaches to speaker verification research in the last 3 decades, and illustrate these methods with a few specific examples.

During the late 1960's and 1970's, researchers mainly used knowledge-based approaches to speaker verification research. Since many of the researchers are speech scientists knowledgeable of the acoustic-phonetic encoding of speech, they focused their attention on the discovery of features, typically measured across speech segments. Speech segments, or phone units, were believed to be the appropriate choice of units, because speaker-specific cues may manifest themselves differently depending on the manner of articulation of phones. While these features may be sound on theoretical grounds, algorithms for automatically computing these features were inadequate. Consequently, investigators resorted to manually segmenting speech data and estimating features to conduct their studies, which constrained the amount of data observed, and the statistical validity of their results.

One example of research done in this era is the doctoral thesis of Wolf [40]. Wolf found specific segmental measurements that discriminated well among speakers. He investigated 17 different features such as, fundamental frequency (F0), glottal source spectral slopes, duration, and features characterizing vowel and nasal spectra. During training, 21 male speakers repeated 6 short sentences 10 times. Nine of the repetitions of each utterance were used to develop speaker templates consisting of means and variances of the features. The remaining sentences were used to test the speakers. During testing, Euclidean distances between test data and speaker templates were used to classify speakers. Wolf used the F-ratio analysis of variance to evaluate the speaker-discriminating abilities of the measurements. The F-ratio is a weighted ratio of the variance of speaker means to the average of speaker variances. Wolf found that features with high F-ratios resulted in 100% speaker classification accuracy.

Wolf's study showed that segment-based features discriminate well among speakers. Using phonetic units is also advantageous, because the verification can be independent of the particular words the users says. However, Wolf extracted the features from manually segmented speech data. Consequently, he could not build an automated speaker verification system that derived the benefits of his knowledge-based approach. Other studies that also used knowledge-based approaches to speaker verification are described in [37, 14].

In the 1980s, researchers abandoned the notion of using segment-based measurements for speaker verification, because algorithms to automatically segment speech remained inadequate. Instead, investigators began using measurements that are easily computed automatically, such as features extracted from speech frames. Frame-based features may not necessarily distinguish speakers well. However, these measurements allowed researchers to build automated systems. These systems typically modeled speakers with word templates. The templates represented speech frames of words with feature centroids. Just as before, speakers were classified with distances computed between test feature vectors and centroids.

One of the earliest automated speaker verification systems was implemented in the early 1980's at Texas Instruments (TI) corporate headquarters in Dallas, Texas [6].

The system automatically computed features from 6 frames for each word, regardless of the word's duration. Specifically, each frame used the output of a 14 channel filter bank, uniformly spaced between 300 and 3000Hz, as a 14x1 spectral amplitude feature vector. During training, templates for 16 words were constructed for each speaker. During testing, the system prompted 4-word utterances constructed randomly from the 16 word bank. A Euclidean distance between measurements of test frames and reference frames was then computed, and used to make a verification decision. At the time, the system achieved 99.1% acceptance rate of valid users, and 0.7% acceptance rate of impostors. Similar speaker verification systems that use template matching classification techniques are described in [15, 9].

As mentioned above, these pioneering systems typically modeled words with templates for each speaker. Templates do not capture variations in the acoustic feature space, because each frame is represented by a fixed acoustic centroid. Consequently, the templates are not robust models of speech. In addition, the system is dependent on the words the users says during verification.

In the early 1990s, statistical models of speech became popular for speech recognition, because the models represent the acoustic feature space with a distribution, rather than a fixed centroid. As a result, researchers began applying the technology to speaker verification. Specifically, speaker verification research focused on investigating hidden Markov models (HMMs), because HMMs were becoming very successful in speech recognition [32]. Many investigators simply modified existing speech recognition systems for speaker verification, in hopes of achieving high performance. HMMs are developed from frame-based features; therefore, investigators neglected to further explore segment-based features. In fact, most of the studies use frame-based cepstral measurements, and compare different HMM speaker models to each other.

An HMM models speech production as a process that is only capable of being in a finite number of different states, and each state generates either a finite number of outputs or a continuum of outputs. The system transitions from one state to another at discrete intervals of time, and each state produces a probabilistic output [27]. In a speaker verification system, each speaker is typically represented by an HMM, which may capture statistics of any component of speech such as a sub-phone, phone, sub-word, word etc. To verify the speaker, the test sentence is scored by the HMM. The score represents the probability of an observation sequence, given a test sequence and a speaker HMM.

Furui and Matsui investigated various HMM systems for speaker verification. In one study [25], they built a word-independent speaker verification system and compared discrete HMM to continuous HMM speaker models. The speaker verification system computed frame-based cepstral features, and the corpus consisted of 23 male and 13 female speakers, recorded during three sessions over a period of 6 months. Ten sentences were used to train both continuous and discrete HMMs for each speaker, and 5 sentences were used to test the speakers. During testing, the purported speaker's cumulative likelihood score was used to make a verification decision. Furui and Matsui reached a performance of 98.1% speaker verification rate, using continuous HMMs. Other studies that are based on HMMs include [24, 35, 34].

Recently, investigators have applied other statistical methods, such as neural networks, to speaker verification. Neural networks have also been successful in other tasks, such as speech and handwriting recognition. They are statistical pattern classifiers that utilize a dense interconnection of simple computational elements, or nodes [20]. The layers of nodes operate in parallel, with the set of node outputs in a given layer providing the inputs to each of the nodes in a subsequent layer. In a speaker verification system, each speaker is typically represented by a unique neural network. When a test utterance is applied, a verification decision is based on the score for the speaker's models. Some examples of systems that use neural networks to represent and classify speakers are [41, 3, 18, 28, 36].

1.3 Discussion

Thirty years ago, researchers manually computed segment-based acoustic features, and modeled the speech signal with templates consisting of acoustic centroids. Presently, systems automatically compute frame-based acoustic features, and use statistical models to represent the speech signals, such as HMMs and neural networks. As Matsui and Furui showed in one of their studies [25], most statistical methods give improved performance over template methods. In addition, frame-based measurements are easy to compute and are successful in speaker verification. However, segment-based features have been proven to carry speaker-specific cues, and may result in equivalent performance with less dimensionality.

1.4 Thesis Objective and Outline

The ultimate goal of speaker verification research is to develop user-friendly, high performance systems, that are computationally efficient and robust in all environments. In this study, we strive to develop a competitive segment-based speaker verification system that requires minimal computation. We automatically compute segment-based measurements, and use statistical models of speech to represent speakers. Essentially, we combine two successful approaches to speaker verification, knowledge-based and statistical. As a result, we hope to achieve competitive speaker verification performance with minimal computation. We do not investigate robustness issues specifically. However, we explore acoustic features that have been proven to be robust in the past, such as fundamental frequency and energy [41, 17].

To achieve our goal, we modified SUMMIT, a state-of-the-art speech recognition system developed at MIT [43], for speaker verification. We chose SUMMIT for the following reasons. First, SUMMIT treats the speech signal as a concatenation of segments, which allows us to capitalize on the speaker-discriminating abilities of such units. Second, SUMMIT allows us to model the features statistically; therefore we can also capture feature-varying attributes in the speech signal. Finally, SUMMIT employs search algorithms, which allows us to modify the algorithms to conduct a search for an optimal feature set. We search for an optimal feature set from an initial pool of measurements, which include cepstral and prosodic measurements.

Details of our speaker verification system and its design are given in chapter 2. The system description is followed by a presentation of our experimental results and analysis in chapter 3. Finally, chapter 4 summarizes conclusions of our system results, and proposes future work in speaker verification.

Chapter 2

System Description

2.1 Introduction

In this chapter, we describe the components of our speaker verification system. Figure 2-1 summarizes our system with a block diagram, whose building blocks are components of SUMMIT, modified to suit our needs. Initially, signal processing transforms the speech samples to frame-based acoustic features. These features are then used to propose a segmentation network for the utterance. Next, the acoustic measurements are averaged across segments, and rotated into a space that de-correlates them, via principal components analysis (PCA) (section 2.6). During training, diagonal Gaussian speaker models are developed. During testing, the speaker models are used to compute forced alignment scores (section 2.7.3) for test utterances. Finally, the scores (section 2.7.3) are used to classify speakers, and make a verification decision.

This chapter begins with a description of the corpus used to train and evaluate our system. Next, the acoustic features selected to represent the speech signal are discussed. Thereafter, the algorithm used to create a segmentation network from the frame-based features is described, and followed by a discussion of a search for an optimal set of segment-based measurements. Finally, details are given on how speaker models were developed, and how speakers were classified.

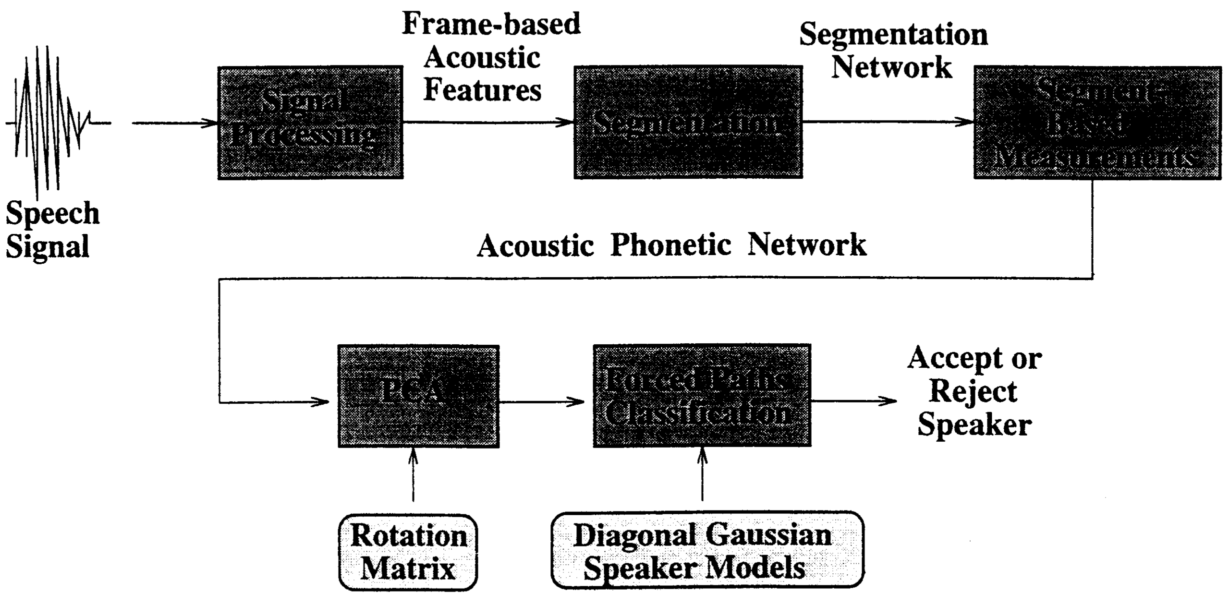

Figure 2-1: Speaker Verification System

2.2 Corpus

Many researchers in speaker verification use a variety of existing corpora, while others collect their own data. We chose to use the TIMIT corpus for a variety of reasons [10]. First, TIMIT is publicly available and widely used. Therefore, it facilitates a direct comparison of our work with that of others. Second, TIMIT contains data for many speakers, and provides time-aligned phonetic transcriptions. Thus, TIMIT allows us to easily develop phonetic models for each speaker. In addition, TIMIT consists of sentences, which create a more natural environment for users than, for example, passwords or digit combinations. YOHO, a corpus specifically designed for speaker verification, contains large amounts of data per speaker and a large number of speakers. However, the corpus consists solely of digits [4]. Finally, NTIMIT, a corpus obtained by transmitting TIMIT over a telephone network, is also publicly available [16]. Since our future work may include investigating speaker verification performance in noisy environments, such as the telephone domain, the availability of NTIMIT will allow us to replicate experiments under noisy conditions, and to make meaningful comparisons to clean speech (TIMIT) results.

2.2.1 TIMIT

TIMIT consists of 630 speakers, 70% male and 30% female, who represent 8 major dialect regions of the United States. We selected a subset of 168 speakers (TIMIT's standard NIST-test and NIST-dev sets) for evaluation. Each speaker read a total of 10 sentences, 2 dialect (SA), 5 phonemically rich (SX), and 3 other (SI) sentences. The 2 SA utterances are the same across all speakers, while the 3 SI sentences are unique to each speaker. A collection of 450 SX sentences in TIMIT are each read by 7 speakers, whereas 1890 sentences from the Brown corpus were each read by one speaker. We used 8 sentences (SX,SI) to develop each speaker model, and the remaining 2 SA sentences to test each speaker. Since 8 utterances may not adequately model a speaker's sound patterns, it is necessary to compensate for the lack of training data. In this study, the complexity of the speaker models is reduced by forming broad phonetic classes.

2.2.2 Broad Classes

As mentioned above, 8 utterances do not contain enough tokens to adequately model all phones separately. Therefore, we increased the number of tokens per model by collapsing phones into broad classes. For the speaker verification task, the broad classes should capture speaker-specific cues. Since past observations have shown that speaker trends are easily captured in the broad manner classes [30, 40], we chose to collapse the 61 TIMIT-labeled phones into 6 broad manner classes. As a result, each speaker is represented by 6 broad class distributions, as opposed to 61 phone distributions, and the average number of tokens per model increases by a factor of 10.[1]

The manner classes are obtained based on our knowledge about acoustic phonetics, and consist of vowels, nasals, weak fricatives, strong fricatives, stops, and silence. The exact content of each manner class is shown in Table 2-1.

[1] The average number of tokens per phone is 5, whereas the average number of tokens per broad class is 50.

CLASS	PHONES
Vowels	iy,ih,eh,aa,ay,ix,ey,oy,aw,w,r,l,el,er,ah,ax,ao,ow,uh,axr,ax-h,ux,ae
Stops	b,d,g,p,t,k
Nasals	m,em,n,en,nx,ng,eng,dx,q
Strong Frics	s,sh,z,zh,ch,jh
Weak Frics	f,th,dh,v,hh,hv
Silence	pcl,tcl,kcl,bcl,dcl,gcl,pau,epi,h#

Table 2-1: Phone Distributions of Broad Manner Classes

The selection of the classes affects the performance of each feature set. For example, voiced and unvoiced stops are clustered together into one stop class. Voiced and unvoiced stops differ significantly in duration, because voiceless stops have added aspiration. Thus, speaker distributions for the stop class, using duration as a feature, will have large variances. These large variances make it difficult to distinguish among the users, and may result in poor speaker verification performance.

2.3 Signal Representations

After choosing a corpus, we collected 17 features to represent the speech signal. The features include measurements that are commonly used in speaker verification systems, such as MFCCs, in addition to three prosodic measurements: fundamental frequency, energy and duration. Below, we describe why the above features were selected for the speaker verification task, and how we computed them.

2.3.1 MFCCs

Mel-frequency-based cepstral coefficients (MFCCs) are perhaps the most widely used features in speaker verification. MFCCs are cepstral features obtained from a system that approximates the frequency response of the human ear. Presumably, MFCCs have been successful in speaker verification because they capture inter-speaker differences. It can be shown via cepstral analysis of speech [29] that MFCCs carry vocal tract information (i.e., formant frequency locations), as well as fundamental frequency information. The vocal tract system function is dependent on the shape and size of the vocal tract, which is unique to a speaker and the sound that is being produced. Fundamental frequency (F0) also carries speaker-specific information, because F0 is dependent on accents, different phonological forms, behavior and other individualistic factors [41, 1].

To compute MFCCs, the speech signal was processed through a number of steps. First, the digitized utterances were initially passed through a pre-emphasis filter, which enhances higher frequency components of the speech samples, and attenuates lower frequency components. Next, a short time Fourier transform (STFT) of the samples was computed at an analysis rate of 200 Hz, using a 20.5 ms Hamming window. The STFT thus produced one frame of spectral coefficients every 5 seconds. Then, each of the coefficients was squared component-wise to produce the power spectral density (PSD) for each frame. Thereafter, the logarithm of the PSD was computed and the resulting coefficients were processed by an auditory filter bank, which produced mel-frequency spectral coefficients (MFSCs). Finally, the MFSCs were rotated by the discrete cosine transform (DCT) matrix. The matrix transformed the mel-frequency spectral coefficients (MFSCs) to 14 less correlated MFCCs. More details are given in Appendix A.

2.3.2 Prosodic Features

In addition to MFCCs, we decided to explore three prosodic features: fundamental frequency (F0), energy and duration. These features attempt to measure psychophysical perceptions of intonation, stress, and rhythm, which are presumably characteristics humans use to differentiate between speakers [6]. Prosodic features have also proven to be robust in noisy environments [41, 17, 1]. Therefore, these features show great potential for the speaker verification task.

To estimate F0, we used the ESPS tracker, in particular the FORMANT function [7]. For each frame of sampled data, FORMANT estimates speech formant trajectories, fundamental frequency, and other related information. The ESPS formant tracker implements the linear prediction analysis method, described in Appendix B, to estimate F0. FORMANT also uses dynamic programming and continuity constraints to optimize the estimates of F0 over frames. Although the tracker also estimates probabilities of voicing for each frame, we retained F0 information for every frame, regardless of whether the underlying sounds were voiced or unvoiced.

To compute energy, the power spectral density coefficients for each frame, obtained in the same manner as described in section 2.3.1, were summed. We computed the logarithm of this sum to convert energy to the decibel (dB) scale. The logarithm of duration was also computed in our experiments.

2.4 Segmentation

Once frame-based acoustic features are computed, the system proposes possible segmentations for the utterance. The goal of the segmenter is to prune the segment search space using inexpensive methods, without deleting valid segments. During segmentation, frame-based MFCCs are used to first establish acoustic landmarks in the utterance. Then, a network of possible acoustic-phonetic segments are created from the landmarks.

Acoustic landmarks are established in two steps. First, the algorithm identifies regions of abrupt spectral changes, and places primary landmarks at these locations. Next, secondary landmarks are added to ensure that a specified number of boundaries are marked within a given duration. To create the network of possible acoustic-phonetic segments, the procedure then fully connects all possible primary landmarks for every deleted secondary landmark.

An analysis of the networks proposed using this algorithm shows that on a development set, there are an average of 2.4 landmarks proposed for every transcription landmark, and 7 segments hypothesized for every transcription segment [12]. The multi-level description of the segmentation is illustrated in Figure 2-2 for the utterance "Delta three fifteen". The segmentation algorithm is described in more detail in [11].

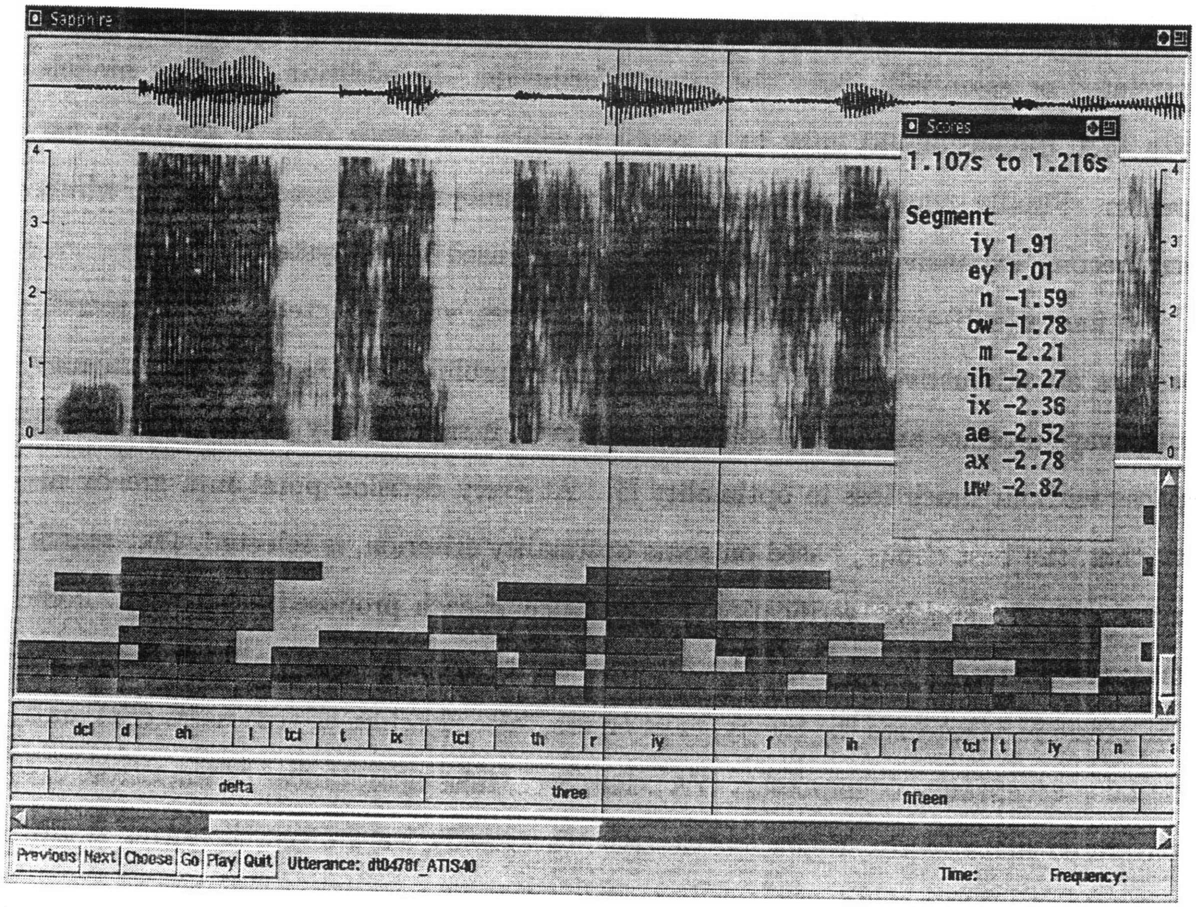

Figure 2-2: Segmentation Network Proposed by SUMMIT: The waveform of the utterance is displayed at the top of the figure. Below the speech waveform is a spectrogram, and the segmentation network proposed is illustrated below the spectrogram. Finally, the phonetic labels of the utterance are given underneath the segmentation network.

2.5 Measurement Search

Each of the segments proposed by the segmentation algorithm is described by a set of acoustic features. The set of 17 measurements discussed above represents a pool of possible features to characterize segments. We did not use all 17 measurements in the system for the following reasons. First, some features may be useful in discriminating speakers well, while others may not. Second, some of the measurements may be correlated or essentially carry the same information. In addition, training models with high dimensionality may be a problem since not much data is available per speaker. Finally, computation increases as the number of features increases, which may become expensive if all 17 measurements are used in the system.

To find a (sub)-optimal subset of the 17 features, we conducted a greedy search, because an exhaustive search is computationally prohibitive. A greedy search may not always produce an optimal solution. However, it significantly prunes large search spaces without much loss in optimality [5]. At every decision point in a greedy algorithm, the best choice, based on some optimality criterion, is selected. Our search criterion is the speaker verification performance of each proposed feature set. Performance is measured in terms of a distance metric describe in detail in section 3.2. The measure minimizes the two types of errors, false rejection of true users (FR) and the false acceptance of impostors (FA). However, false acceptances of impostors are considered more costly. Below, we describe the greedy feature search, which is also illustrated in Figure 2-3 for an initial pool of 5 features.

The search algorithm begins by obtaining FR rates and FA rates for the 168 test speakers, using each of the 17 features. Thus we obtain 17 performance results corresponding to each measurement. The feature that results in the smallest distance measure (best performance) is chosen as the best 1-dimensional measurement. Next, the best 1-dimensional feature is combined with each of the remaining measurements. Two-dimensional feature sets are grouped in this fashion, and are each used to test the 168 speakers. The best 2-dimensional feature vector, in terms of speaker verification performance, is then used for the next stage of the search. The search continues to accumulate dimensions in the feature set until there is no longer significant improvement in speaker verification performance, or if performance actually degrades as more features are added.

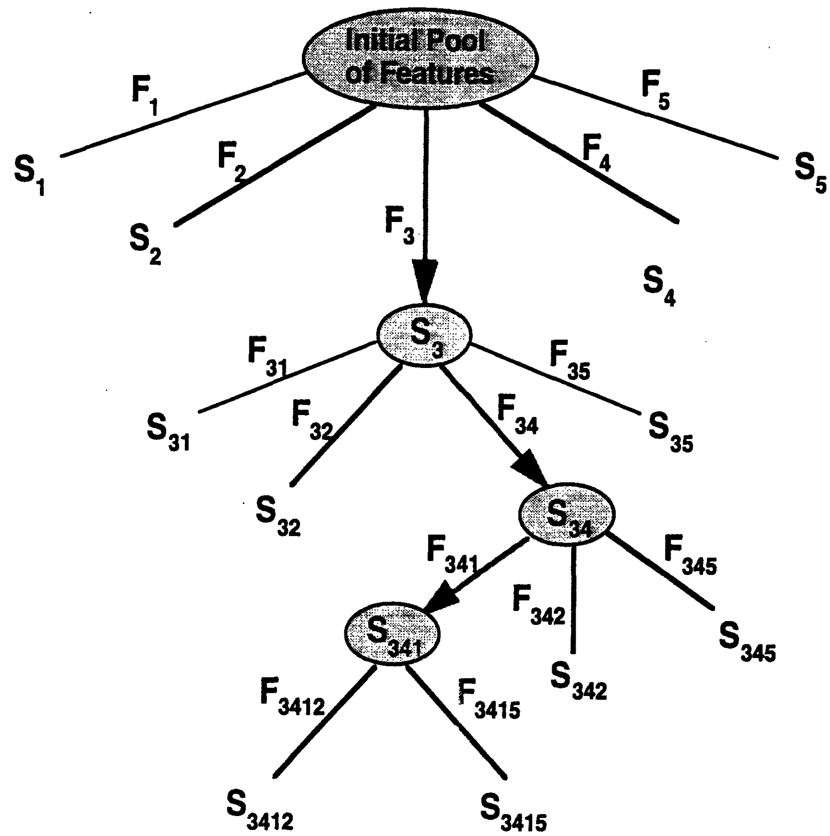

Figure 2-3: Illustrative Example of our Greedy Feature Search: F_{ijk} is the set of features i, j, and k. S_{ijk} is the corresponding verification score in terms of a distance measure. First, each feature is individually tested, and feature #3 results in the best speaker verification performance. Next, feature #3 is combined with each of the 4 remaining features to form 2-dimensional sets. Features #3,4 then result in the best performance (which is significantly better than the 1-dimensional set). This 2-dimensional set is then combined with the 3 remaining measurements to form 3-dimensional sets. Finally, features #3,4,1 is the optimal set, because performances of the two 4-dimensional sets fail to significantly improve over the 3-dimensional set.

2.6 Speaker Models

During training, statistical models of segment-based acoustic features are developed for each speaker. Specifically, the speaker models consist of diagonal Gaussian probability density functions (pdfs). We chose to represent the acoustic space with Gaussian distributions because features of speech data, such as cepstral coefficients, fit these bell-shaped curves well [38]. Diagonal distributions were implemented because they have few parameters to train (diagonal covariance matrices), and thus do not require much training data to accurately estimate the parameters. However, features that are correlated are not modeled well with diagonal covariance matrices.

To ensure that the features fit the diagonal models better, principal components analysis (PCA) was performed on the acoustic features before developing the models. PCA rotates a d-dimensional space to a set of orthogonal dimensions (less than or equal to the dimension d). As a result, the full covariance matrix of the original space is transformed to a diagonal matrix in the new space. In principle, PCA also allows us to reduce the dimensionality of the feature vectors. However, in our experiments, we did not reduce dimensionality with PCA since the feature search already prunes the number of features used in the system.

The Gaussian distributions that model the acoustic features for each speaker are developed using the maximum likelihood (ML) estimation procedure. The mathematical expressions for the ML estimates for the means, variances and the *a priori* class probability estimates for a particular speaker model are shown below. An example of a speaker model developed using the ML procedure is shown in Figure 2-4. Figure 2-4 illustrates a histogram of a speaker's training data and the corresponding model developed. It is apparent that a single diagonal Gaussian cannot completely model the data for each class. Mixtures of diagonal Gaussians may fit the data better. However, there are more parameters to train mixtures of Gaussians, which require more data than are available.

$$j = \text{the } jth \text{ broad class}$$

$$n_j = \text{the number of tokens for class } j$$

$n =$ the total number of tokens for all classes

$x_{j,k} =$ the kth data token for class j

$\overline{\mu}_j =$ the ML estimate of the mean for class j

$\overline{\sigma}_j^2 =$ the ML estimate of the variance for class j

$P(j) =$ *a priori* probability for class j

$$\overline{\mu}_j = \frac{1}{n_j} \sum_{k=1}^{n_j} x_{j,k}$$

$$\overline{\sigma}_j^2 = \frac{1}{n_j} \sum_{k=1}^{n_j} (x_{j,k} - \overline{\mu}_j)^2$$

$$P(j) = \frac{n_j}{n}$$

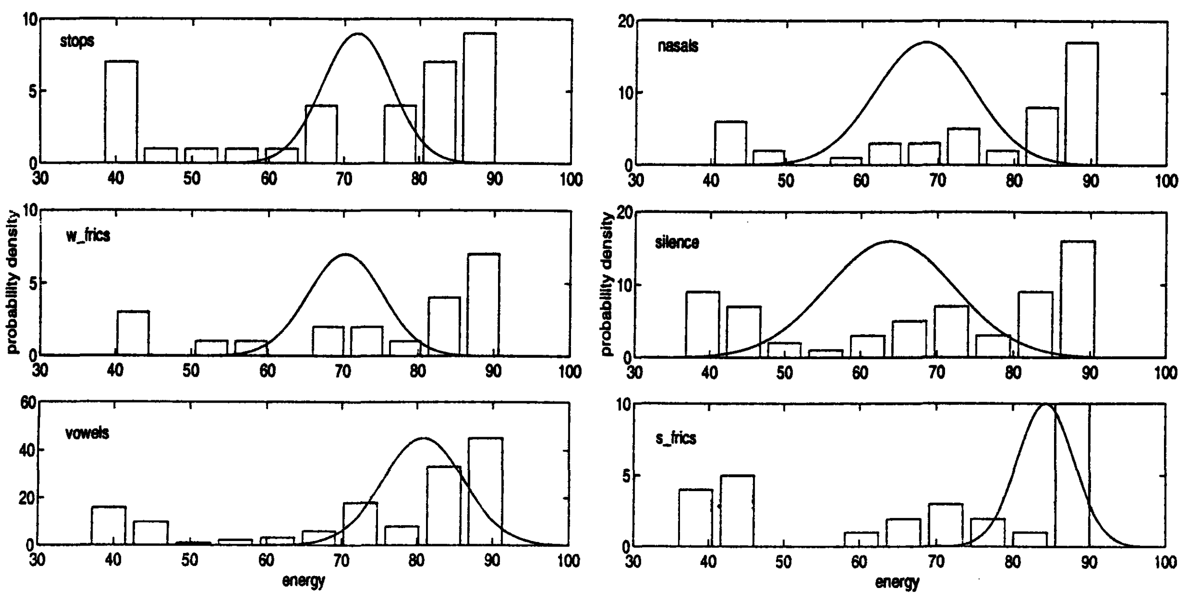

Figure 2-4: Histogram of Data and Corresponding ML Model of a Speaker

2.7 Speaker Classification

Once speaker models are developed, test utterances are scored against these models to classify speakers and make verification decisions. Below we describe our testing conditions, which use the concept of cohort normalization. Next, we describe the verification process and conclude with a description of how scores are computed.

2.7.1 Cohort Sets

During testing, it is ideal to compare the utterances to all speaker models in the system, and accept the purported speaker if his/her model scores best against the test data. However, computation becomes more expensive as speakers are added to the system. Since speaker verification is simply a binary decision of accepting or rejecting a purported speaker, the task should be independent of the user population size.

To keep our system independent of the number of users and computationally efficient, we implemented a technique called cohort normalization. For each speaker, we pre-detected a small set of speakers, called a cohort set, who are acoustically similar to the purported speaker.[2] During testing, we only test the speakers in the cohort set for the purported speaker. Speakers outside the cohort set are considered outliers that have low probabilities of scoring well against the purported speaker's test data. Therefore, results using just cohorts during testing may minimally affect speaker verification performance, and can be normalized to emulate results using all speakers during testing. Detailed results of the normalization are given in section 3.4.

For each feature set, we found S nearest neighbors (cohorts) for each speaker using the Mahalanobis distance metric [39]. Specifically, μ_1 and μ_2, σ_1^2 and σ_2^2, are d-dimensional mean vectors and dxd-dimensional covariance matrices for two speaker models, respectively. The Mahalanobis distance squared, D^2, between the two speakers is then

$$D^2(1,2) = \sum_{i=1}^{d} \frac{(\mu_{1i} - \mu_{2i})^2}{\sigma_i^2}$$

where

$$\sigma_i^2 = \frac{n_1 \sigma_{1i}^2}{n_1 + n_2} + \frac{n_2 \sigma_{2i}^2}{n_1 + n_2}$$

[2]The size of each cohort set is a parameter we varied in our experiments.

and n_1 and n_2 are the number of data vectors for speaker one and speaker two, respectively.

Once the speaker models were developed, this metric was applied to every possible pair of speakers. The distances were then sorted for each speaker, and the cohorts were chosen to be the S closest neighbors to each speaker.

An example of a female speaker and her cohorts' models for the 6 broad classes, using F0 as a feature, is shown in Figure 2-5. Distributions of 4 cohorts are plotted along with models of 2 outliers (1 female and 1 male) of the cohort set for that speaker. As expected, the cohort models are very similar to the speaker's model, while there is more disparity between the true speaker's models and the models of the outliers.

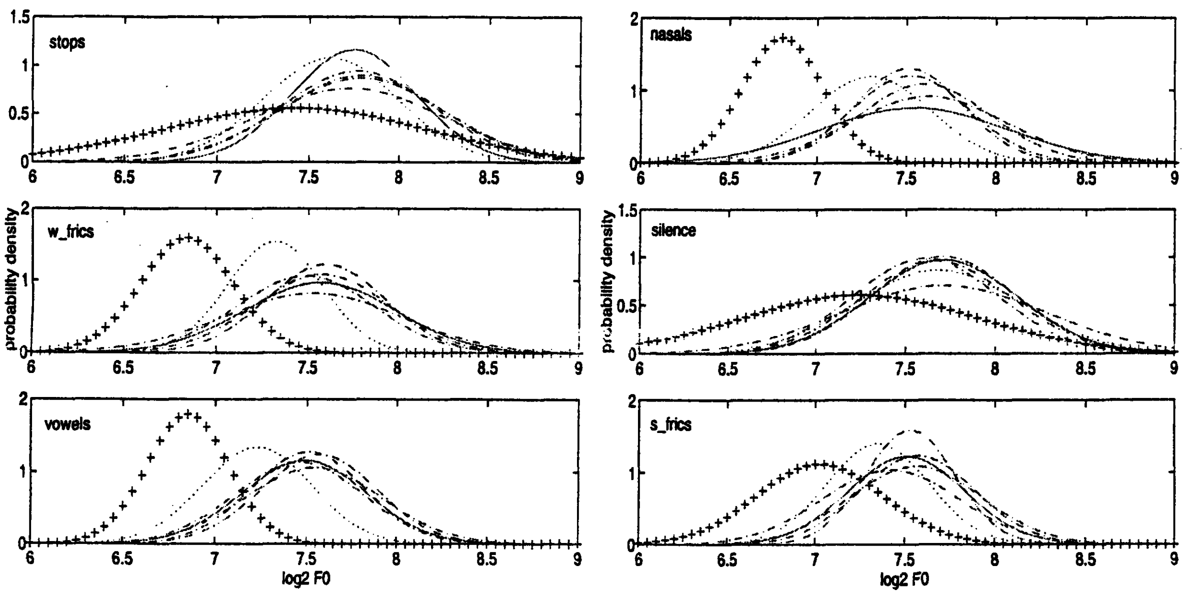

Figure 2-5: Speaker F0 Models and Cohorts: '-' models represent the true female speaker and '-.' models are 5 of her cohorts. The '.' models represent a female outlier and the '+' models represent a male outlier of the true speaker's cohort set.

2.7.2 Verification Process

To accept or reject a speaker, we compute forced alignment scores, described in 2.7.3, for the purported speaker's two test utterances. The scores are computed from $S+1$ models, the speaker's model and his/her S cohort models. These scores are then sorted, and the speaker is accepted if the score using his/her model is in the top N

scores of the $S+1$ results.[3] The verification procedure is illustrated in Figure 2-6.

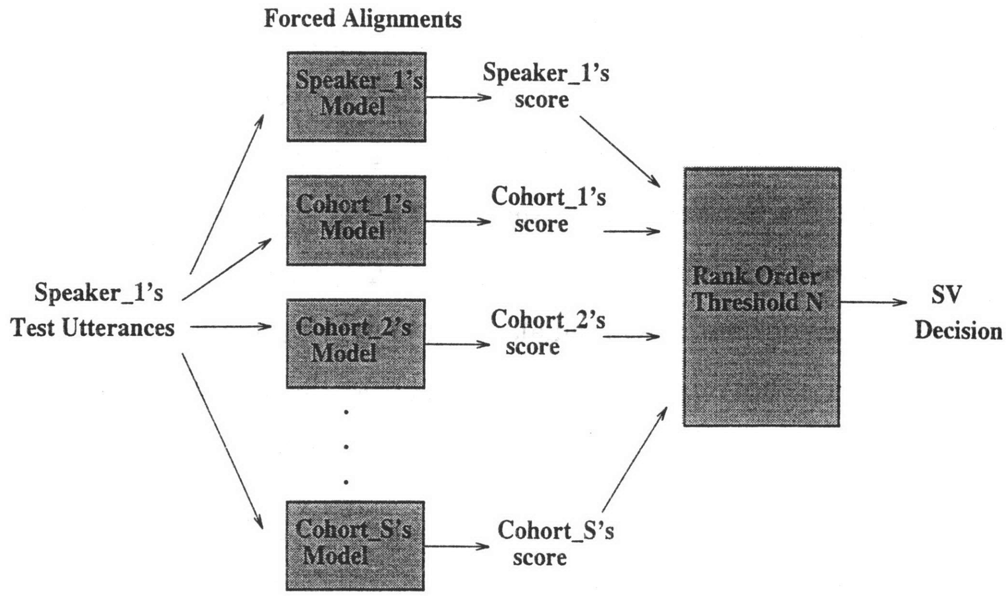

Figure 2-6: Speaker Verification Testing Procedure

False acceptance rates are obtained as they would be in a real impostor situation. If $speaker_a$ poses as $speaker_b$, $speaker_a$'s test utterances are scored by $speaker_b$'s model and $speaker_b$'s cohort models. These scores are then sorted and rank ordered. If the score using $speaker_b$'s model is in the top N scores, he/she is falsely accepted.

2.7.3 Scoring

The scores used to classify speakers correspond to likelihood probabilities accumulated from paths of speech segments. Specifically, a score reflects the probability of observing feature vectors across the path of segments in a forced alignment, for given broad class labels.

A forced alignment is the result of a constrained search, that assigns an utterance's broad class labels to a path of segments. The search is constrained because the broad class labels are known *a priori*; thus it is not necessary to consider all possible classes for each proposed segment. During the search, each possible alignment for an

[3]The rank threshold, N, is a parameter that we varied for each feature set.

utterance accumulates likelihood scores. These likelihood scores reflect the probabilities of observing feature vectors across the segments in the alignment, for the given labels. The path of segments that corresponds to the highest likelihood score, which is used to classify speakers, is chosen as the forced alignment for the test utterance.

Normally, likelihood scores are accumulated along all possible paths. However, the system implements the Viterbi algorithm to find the forced alignment without scoring all possible segmentation paths. The Viterbi algorithm is based on dynamic programming methods, and prunes the search without any loss in optimality. Details of the Viterbi algorithm can be found in [33, 5].

Chapter 3 presents the detailed results of our feature search, followed by analysis. The performance effects of using only cohorts during testing is then illustrated, and the overall system performance is compared to that of two similar speaker verification systems.

Chapter 3

Experimental Results & Analysis

3.1 Overview

To evaluate a speaker verification system, it is necessary to observe how performance is affected as components and test conditions alter. For example, acoustic features, models and classifiers are system components that affect performance. In our case, we have limited the scope of our investigation to examining the sensitivity of performance to acoustic features. We then varied test conditions by observing performances of features using a different set of speakers.

This chapter first describes performance measures used for speaker verification. Then, the results of the greedy feature search conducted on the set of 168 test speakers is presented. In addition, we conducted the same experiment using a set of 80 speakers who are not part of the test set to ensure that the feature selection process is independent of the speakers used. Thereafter, system performance using cohort sets as well as the performance using the entire test set are reported. Finally, our results are compared to those of two other systems that are also evaluated on the TIMIT corpus.

3.2 Performance Measures

The performance of a speaker verification system is typically measured in terms of two types of errors: false rejections of true users (FR) and false acceptances of impostors (FA). Performance is often illustrated with conventional receiver operating characteristic (ROC) curves, which plot the rates of FR versus the rates of FA for some varying parameter.

A popular single number measure of performance is the equal error rate (EER), which is the rate at which the two errors (FR and FA) are equal. EER is thus the intersection between the ROC curve and the line FR=FA. Many researchers design speaker verification systems to minimize the EER. However, minimizing this measure does not allow for different costs to be associated with FA and FR. For high security applications such as bank-by-phone authorizations, minimizing false acceptances of impostors is the first priority. Rejecting a true user may annoy the user. However, accepting an impostor may be costly to the customer.

While our goal is to minimize both types of errors, we have chosen to weigh the cost of false acceptances of impostors more than the cost of false rejections of true users. Specifically, we first obtain the ROC for each feature set by varying the rank threshold, N (3, 6, 9, 12). The system's performance is then measured in terms of a distance between the point on the feature's ROC curve that corresponds to the rank threshold $N = 3$, to the origin, which corresponds to the ideal performance of 0% error. Figure 3-1 uses the ROC curve for energy as an example to illustrate how we computed this distance. The smaller the distance, the more robust the system is to false acceptances of impostors.

3.3 Feature Search

In this section, we present the results of the greedy feature search using 168 test speakers, which is followed by a discussion of the second partial search conducted on a different set of 80 speakers. Both searches were conducted using a cohort set size

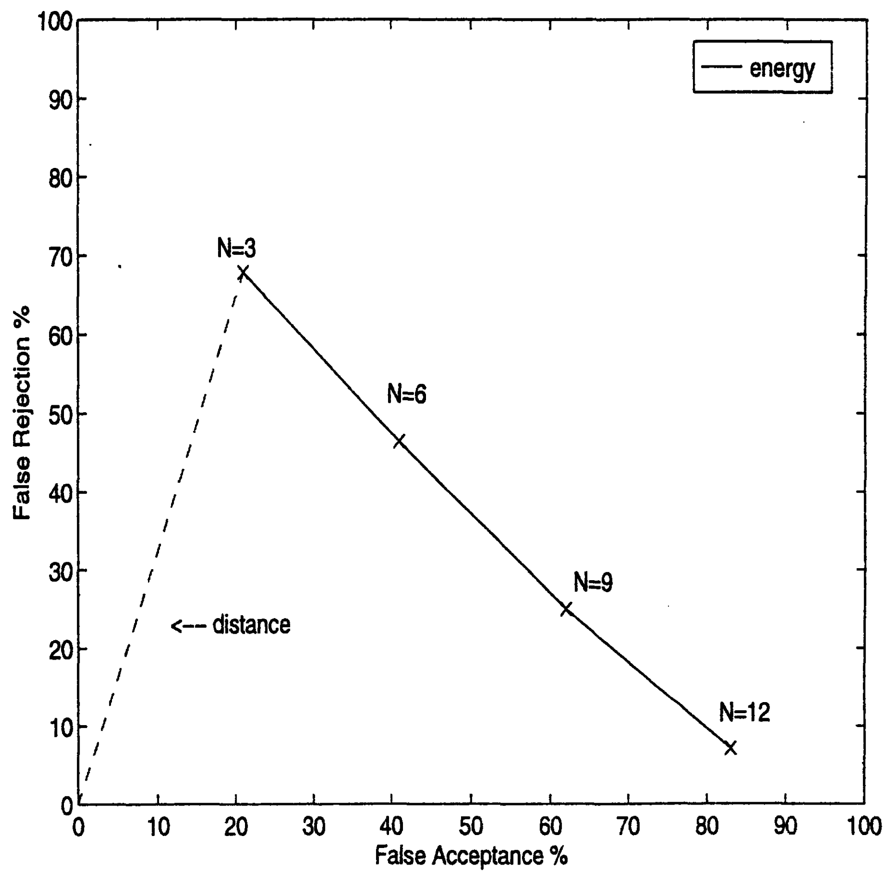

Figure 3-1: ROC Curve for Energy and Corresponding Performance Distance

of 14, and the results reported are not normalized.[1]

3.3.1 Results Using 168 Speakers

The first stage of the search evaluates the speaker verification performance of individual features. Performances of the one-dimensional measurements are given in Table 3-1, and are illustrated in Figure 3-2. Figure 3-2 shows that the top 10 features, in particular the higher order MFCCs, F0, and energy, result in similar performances. We disregarded all other features for subsequent search stages, because they resulted in significantly worse performance than the top 10 features. We realized that such pruning will result in a search that is not greedy in the strictest sense of the word.

Past observations have shown that MFCCs and prosodic features are useful for speaker verification [19, 41, 17]. In our search, we found that two of the three prosodic measurements investigated performed well. Specifically, energy ranked first in the set of 17 features, and F0 ranked fourth. However, duration ranked last in the first stage of the feature search. Perhaps duration performed poorly because of the way the broad classes were formed. Many of the 6 manner classes selected, such as stops and weak fricatives, consist of both voiced and unvoiced phones, which are mainly distinguished by duration. Consequently, the variances of duration for these classes are large for all distributions, and the speaker models are often indistinguishable if the means do not differ by much. As a result, speaker verification performance is poor.

Figure 3-3 illustrates these large variances (on the order of 10^5) of 4 speakers' duration models of stop consonants and weak fricatives. The 4 speakers are within a cohort set. Thus, during testing, these speakers are compared to each other and the remaining members of the cohort set. As shown in Figure 3-3, it is difficult to reliably distinguish among the 4 distributions. In fact, we computed the average Mahalanobis distance between the 4 cohort models for the best and worst features, energy and duration, respectively. These distances are shown in Table 3-2, which

[1] After the search, the normalization is applied to the optimal feature set to obtain the estimated system performance.

FEATURE SET	DISTANCE
F0	72.36
Energy	71.01
Duration	81.51
MFCC1	76.40
MFCC2	74.64
MFCC3	74.35
MFCC4	76.37
MFCC5	72.93
MFCC6	71.24
MFCC7	78.22
MFCC8	72.62
MFCC9	74.50
MFCC10	72.16
MFCC11	78.96
MFCC12	79.25
MFCC13	77.39
MFCC14	73.63

Table 3-1: One Dimensional Feature Set Results

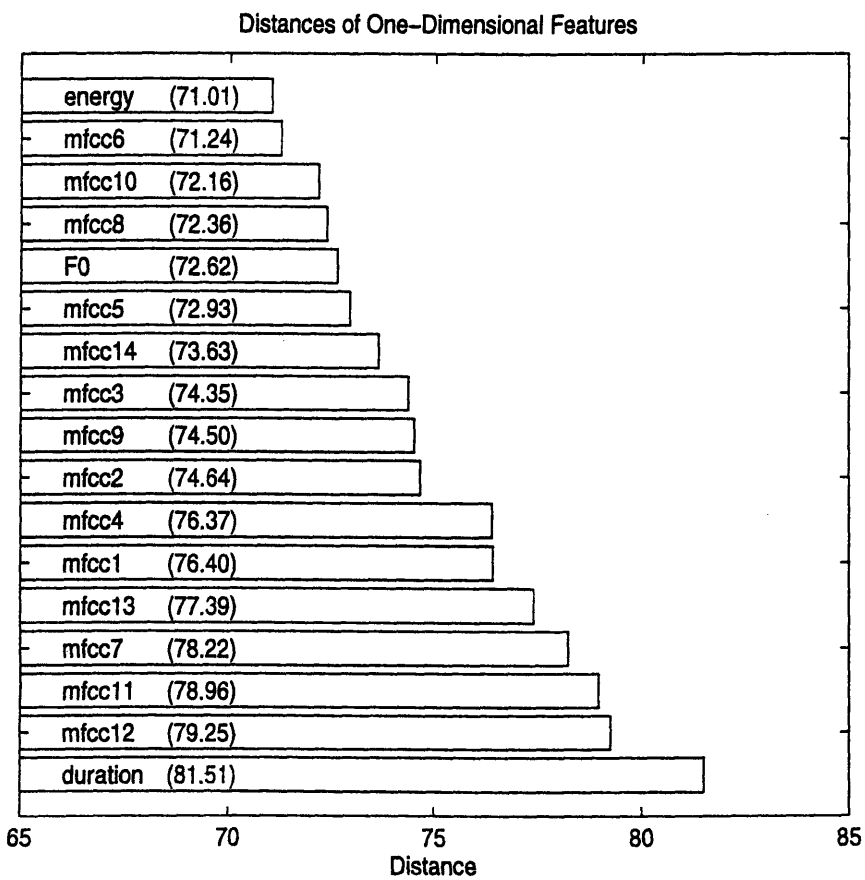

Figure 3-2: Distances for 1-Dimensional Feature Sets

illustrates that the duration models are more similar (smaller distance) to each other than the energy models, discussed below.

Perhaps duration as a measurement could have performed better if our broad classes were selected knowing *a priori* that duration was to be the measured feature. An appropriate selection of broad classes would then be voiced stops, unvoiced stops, voiced fricatives, unvoiced fricatives, long vowels, short vowels etc. Essentially, the classes would have similar duration characteristics.

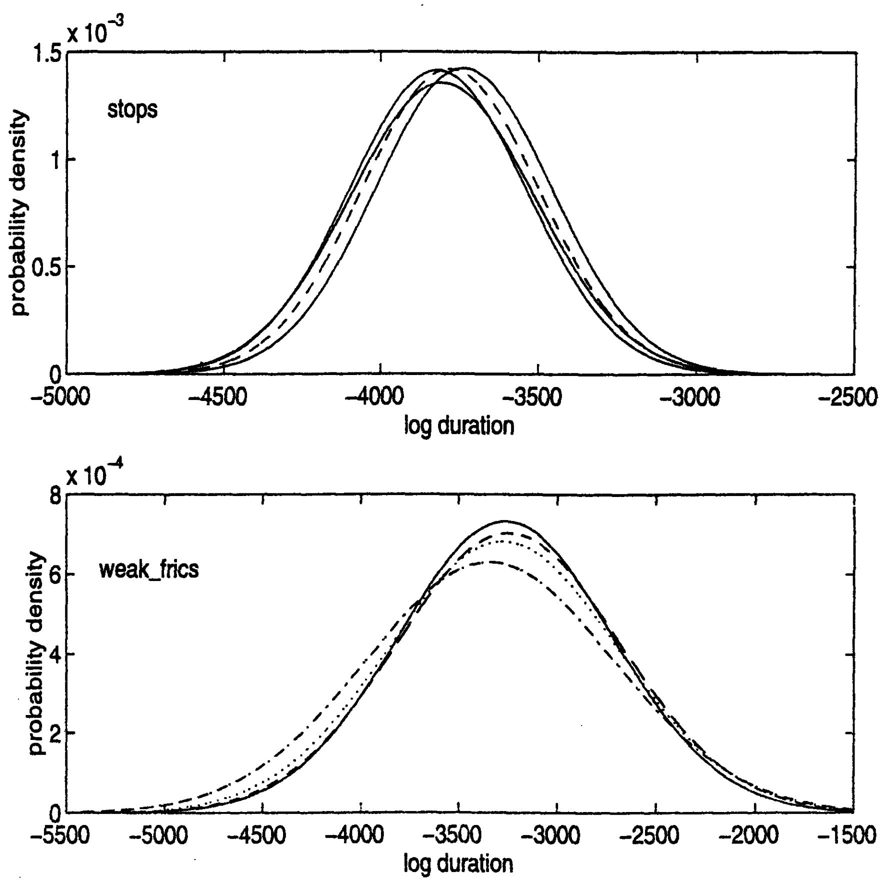

Figure 3-3: Duration models of 4 Speakers

Energy, on the other hand, performed the best in the first stage of the search, suggesting that the energy characteristics within classes are similar. Thus, we expect the opposite trends in the statistics of energy. For example, the energy of strong fricatives is much larger than the energy of weak fricatives. Thus, the strong fricatives' and weak fricatives' models for energy have smaller variances than the duration models for these same two classes. When the variances are small for every speaker model, it is possible to distinguish between speakers with different means.

Figure 3-4, is a similar plot of 4 speakers' (within a cohort set) energy models for weak fricatives and stop consonants. Figure 3-4 illustrates the larger differences between the 4 speaker models of energy than the models of duration, suggesting that distinguishing between speakers is easier using energy as a feature. In fact, as shown in Table 3-2, the average Mahalanobis distance for the energy models is approximately twice that of the duration models, implying that the energy speaker models are much more different than the duration speaker models within cohort sets.

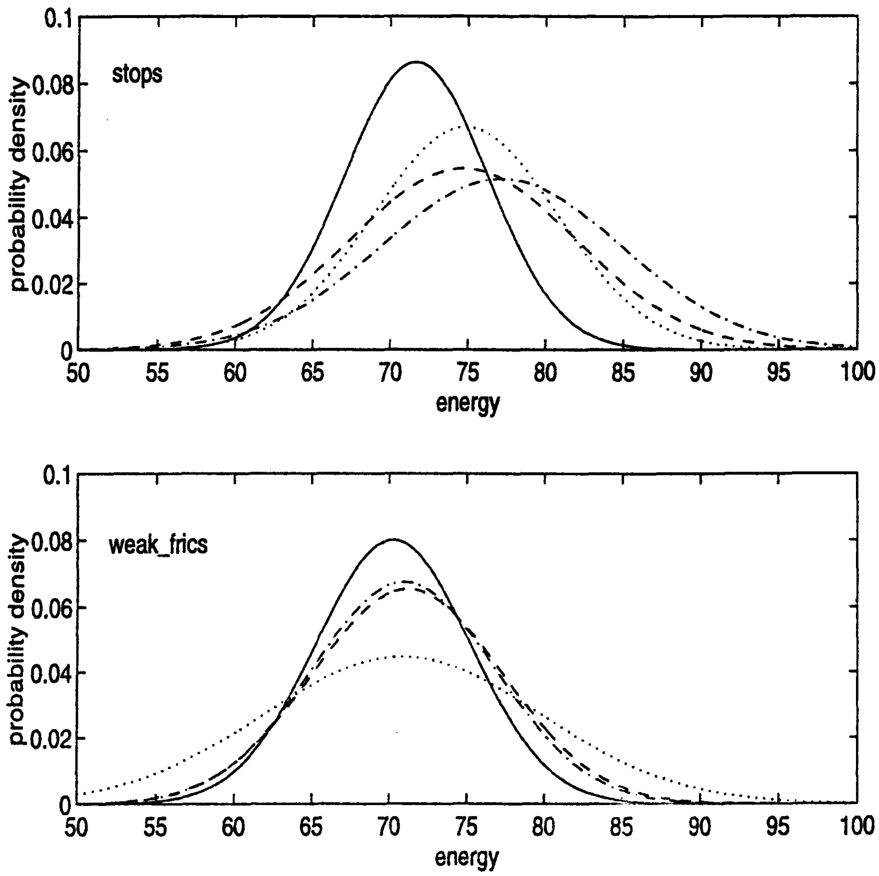

Figure 3-4: Energy models of 4 Speakers

During the second stage of the search, we explored pairs of features, combining energy with each of the remaining 9 features. The 2-dimensional results are shown in Figure 3-5, which illustrates that energy combined with MFCC10 results in the best

Feature	Mahalanobis Distance
Duration	0.479
Energy	0.926

Table 3-2: Average Mahalanobis Distances for 4 Speakers' Duration and Energy Models

speaker verification performance, according to our distance measure. Furthermore, there is a noticeable improvement in performance by the addition of another measurement to the feature set, since the distances are smaller for the 2-dimensional sets than for individual measurements. The performance improvement suggests that the additional features carry further speaker-specific information. Also, the dimension of the feature set is small enough that model parameters can be sufficiently estimated from the 8 training utterances available per speaker.

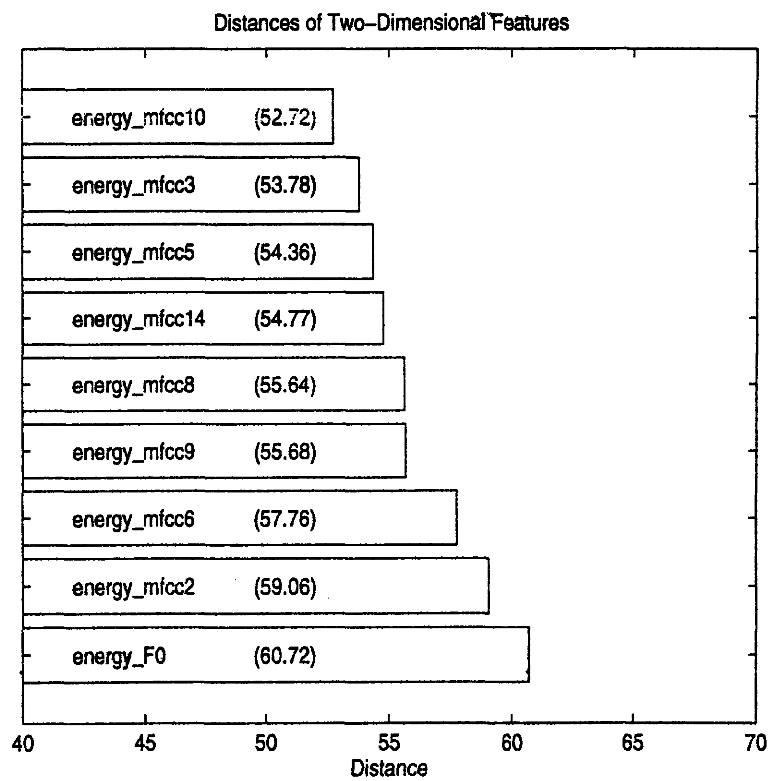

Figure 3-5: Distances for 2-Dimensional Feature Sets

During the third stage of the search, energy and MFCC10 were combined with the 8 remaining measurements. The 3-dimensional results are shown in Figure 3-6, which illustrates that the best 3-dimensional feature set is energy combined with MFCC10 and MFCC5. We continued our feature search, since performance continued to significantly improve, and accumulated dimensions to the feature vector in the manner illustrated above. The results for 4 and 5-dimensional feature sets, along with the numerical results of stages 2-7, are given in appendix C.

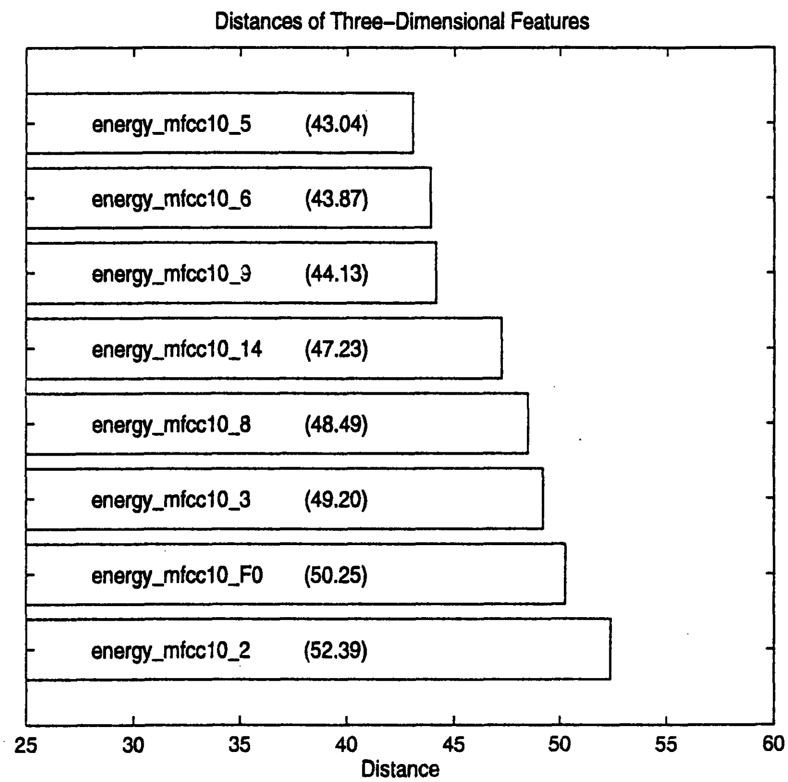

Figure 3-6: Distances for 3-Dimensional Feature Sets

The 5-dimensional feature set that resulted in the best speaker verification performance included energy, MFCC10, MFCC5, MFCC8, and MFCC6. These measurements were then combined with each of the 5 remaining features. The six-dimensional results are shown in Figure 3-7, which illustrates that the best 6-dimensional feature set consists of energy, MFCC10, MFCC5, MFCC8, MFCC6, and MFCC14.

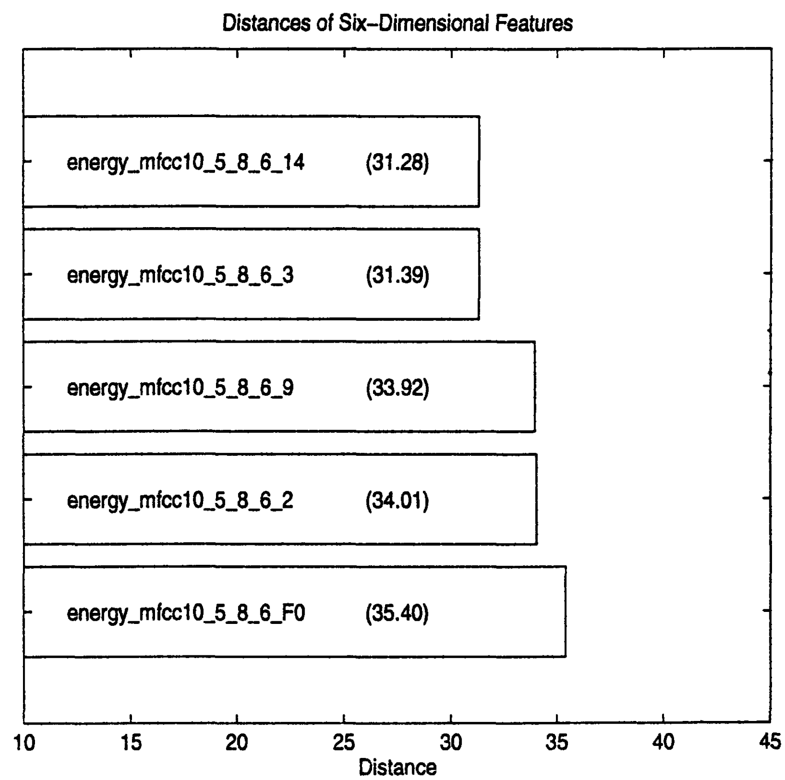

Figure 3-7: Distances for 6-Dimensional Feature Sets

Next, we combined the 7-dimensional features sets to see if additional features improved performance over the best set of 6 features found above. The results for the 7-dimensional sets are illustrated in Figure 3-8, which shows that all 7-dimensional feature sets perform worse than the best 6-dimensional feature set. To observe whether performance continued to degrade as features accumulated, we formed 8 and 9-dimensional feature sets. These sets were formed by adding F0 and MFCC9 to the best 7-dimensional feature set.

Figure 3-9 shows the results for the 8 and 9-dimensional feature sets, along with the best results of each stage of the search. As illustrated in Figure 3-9, speaker verification performance initially improves as more measurements are added to the feature set, because the additional features contribute further speaker-specific information. Also, there are sufficient amounts of training data to accurately estimate the model parameters. However, adding features eventually degrades performance, presumably because not enough training data is available to accurately estimate the model parameters.

The search terminated as a result of the performance degradation, and the best 6-dimensional set, listed below, was considered the (sub-)optimal subset of the 17 collected features.

1. Energy
2. MFCC10
3. MFCC5
4. MFCC8
5. MFCC6
6. MFCC14

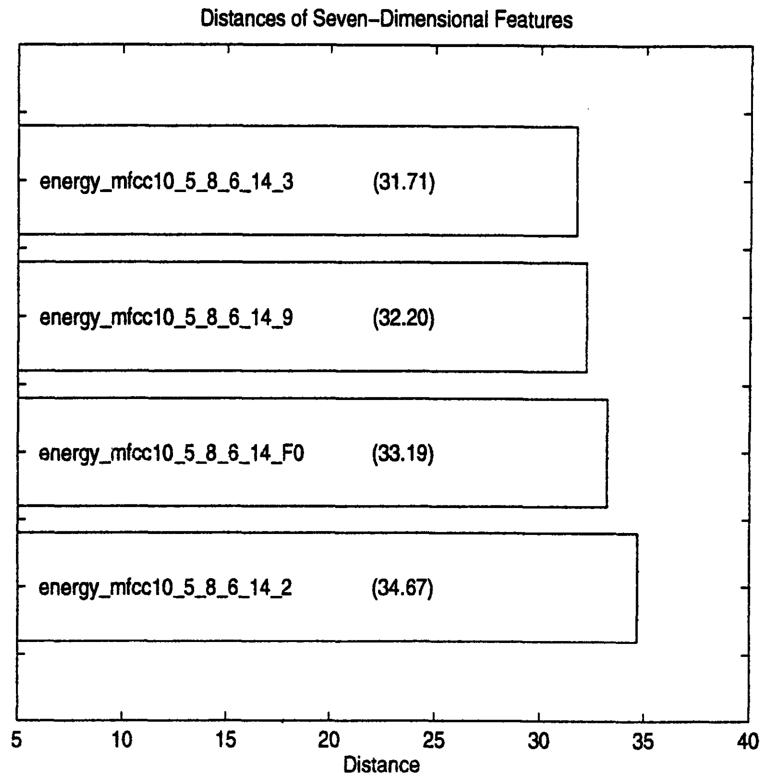

Figure 3-8: Distances for 7-Dimensional Feature Sets

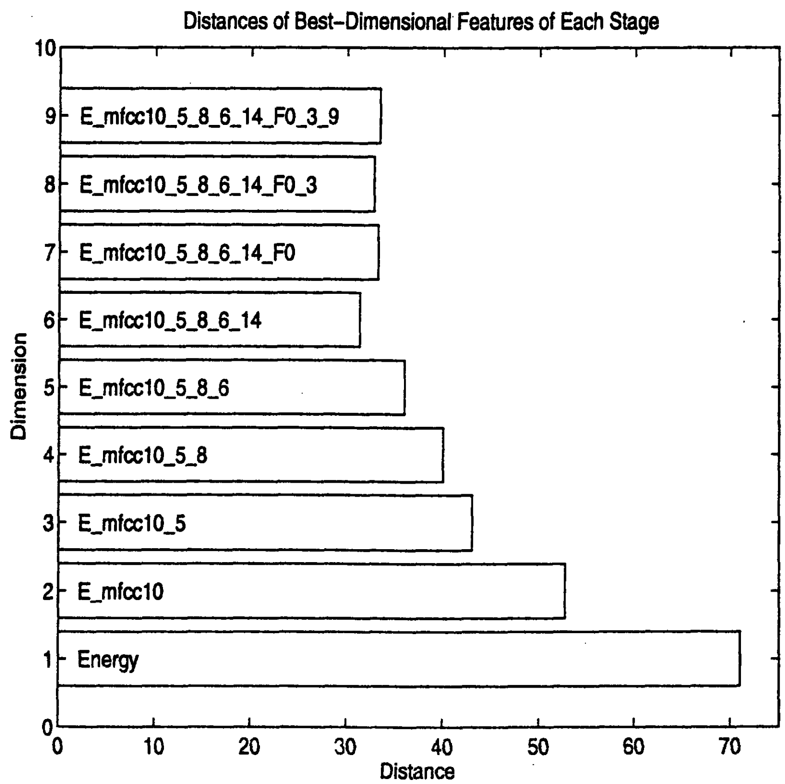

Figure 3-9: Distances for Best Feature Sets of Each Search Stage

3.3.2 Results Using 80 Speakers

The (sub-)optimal 6-dimensional feature set found using 168 test speakers should be independent of the user population. To ensure that the feature search does not produce significantly different results using another set of speakers, we conducted an identical search using a set of 80 different users. The 80 speakers are a subset of TIMIT's NIST-train set. This speaker set does not contain any speakers in the original test set of 168 speakers. However, the 80 speakers have the same ratio of males to females (2 to 1) as the first test set.

As before, we began the search by testing individual features from the initial pool of 17 measurements. Table 3-3 and Figure 3-10 illustrate the results of the first stage of the search. The results are similar to those obtained previously in the first stage. Specifically, 8 out of the top 10 features from this search are included in the top 10 features from the first search. However, the rankings of most features changed. Unlike the first search, most of the 17 features in this search result in similar performances and show potential to be useful features for speaker verification. However, to replicate our search experiments, we still eliminated 7 features and kept the top 10 measurements for the remaining stages of the search. The top ranking features still consisted of energy, F0, and the higher order MFCCs, with the exception of the highly ranked MFCC2. As before, energy performed the best, and duration performed the worst.

Since energy is still the best 1-dimensional feature, it was combined with the remaining 9 of the top 10 features. These 2-dimensional sets were evaluated, and performance results are given in Table 3-4. Figure 3-11 shows that the best pair of features is energy and MFCC10, which is the same top performing 2-dimensional set of the first search.

Due to time constraints, we terminated the search at this stage. However, we believe that the performances of features have not significantly changed by testing on a new set of speakers. The differences in the performances of some features are presumably due to the fact that the new set is half the size of the former test set of

FEATURE SET	DISTANCE (168 SPEAKERS)	DISTANCE (80 SPEAKERS)	DIFFERENCE (MAGNITUDE)
F0	72.36	74.85	2.49
Energy	71.01	68.86	2.15
Duration	81.51	82.64	1.13
MFCC1	76.40	79.42	3.02
MFCC2	74.64	70.71	3.93
MFCC3	74.35	73.31	1.04
MFCC4	76.37	78.08	1.71
MFCC5	72.93	79.50	6.57
MFCC6	71.24	73.29	2.05
MFCC7	78.22	72.04	6.18
MFCC8	72.62	68.87	3.75
MFCC9	74.50	72.39	2.11
MFCC10	72.16	72.33	0.17
MFCC11	78.96	75.33	3.63
MFCC12	79.25	73.64	5.61
MFCC13	77.39	75.47	1.92
MFCC14	73.63	74.92	1.29

Table 3-3: One Dimensional Feature Set Results

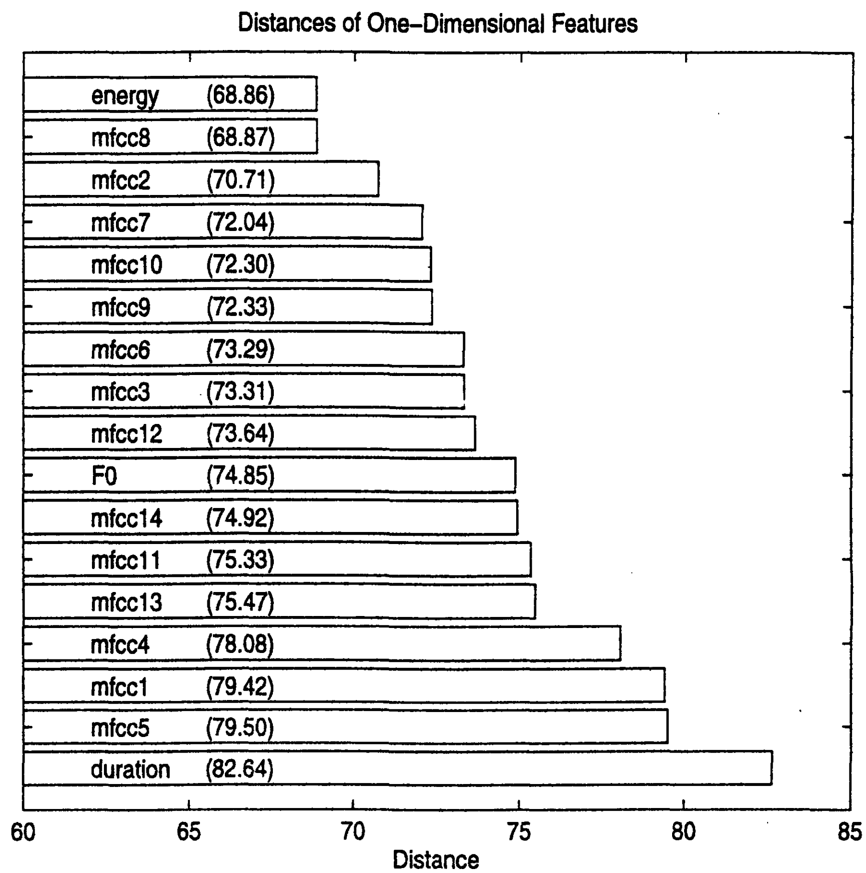

Figure 3-10: Distances for the First Stage of Search using New Data

168 speakers.

FEATURE SET	DISTANCE (168 SPEAKERS)	DISTANCE (80 SPEAKERS)	DIFFERENCE (MAGNITUDE)
Energy_MFCC8	55.64	58.71	3.07
Energy_MFCC2	59.06	57.05	2.01
Energy_MFCC7	–	58.05	–
Energy_MFCC10	52.72	49.50	3.22
Energy_MFCC9	55.68	59.67	3.99
Energy_MFCC6	57.76	60.02	2.26
Energy_MFCC3	53.78	65.96	12.18
Energy_MFCC12	–	57.80	–
Energy_F0	60.72	53.95	6.77

Table 3-4: Two Dimensional Feature Set Results

3.4 System Performance

As mentioned in section 2.7.1, computation during testing is reduced by only scoring test data against the purported speaker's model and models of the purported speaker's cohort set, as opposed to all speaker models in the system. This technique is based on the assumption that speaker models outside of the cohort set will not adversely affect speaker verification performance. Since these outliers are considered too different from the purported speaker, their models are expected to match the test data poorly compared to the speaker models within the cohort set. If this is the case, the ROC curves corresponding to performance using all speakers during testing can be obtained from the ROC curves using only cohort sets during testing, via normalization. The normalization divides the number of false acceptances obtained for a feature set, using for each speaker only the S speakers in his/her cohort set as impostors, by the number of possible false acceptances when all the remaining 167 speakers pose as impostors for each speaker (168 speakers x 167 impostors).

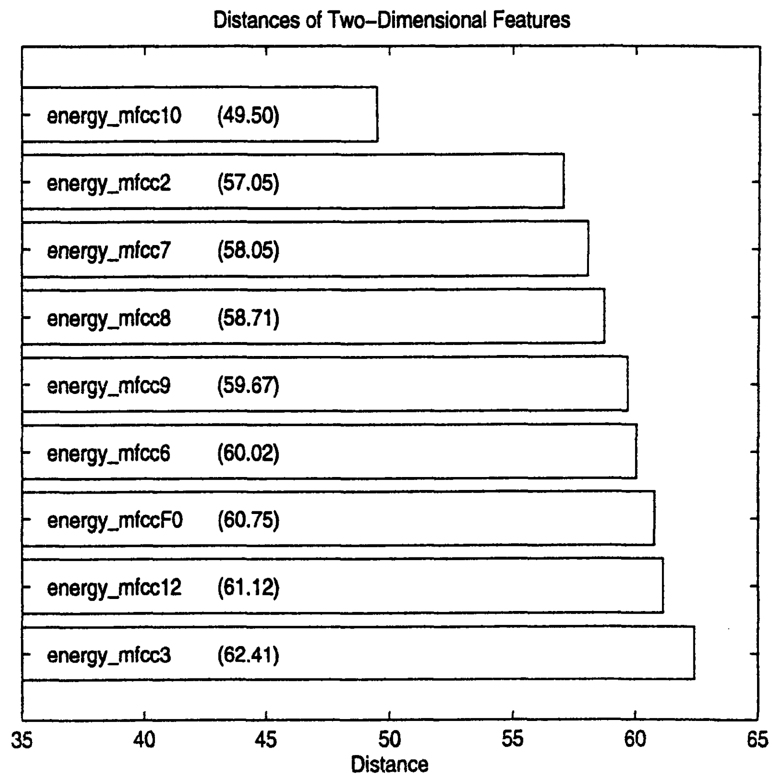

Figure 3-11: Distances for the Second Stage of Search using New Data

In both searches described above, we used a cohort set size of 14. We normalized the results for the optimal 6-dimensional feature set found using 168 test speakers, and obtained a performance of 0% false rejection of true users and 6.54% false acceptance of impostors. To verify whether these normalized approximations are reasonably close to performance using all speakers during testing, we repeated the experiment on the optimal feature set using all 168 speakers during testing. Figure 3-12 plots the ROC curve for the optimal feature set obtained using all speakers as impostors, and the normalized ROC curve obtained using 14 impostors per speaker. As Figure 3-12 illustrates, the normalized results are very similar to the results obtained using all speakers during testing (0% false rejection of true users and 4.85% false acceptance of impostors).

The curves do not match exactly, suggesting that the ranks of the speaker model's score and cohort speaker models' scores were not always the top 15 of 168 scores. In fact, the rank of the true speaker's model within the 15 cohort scores is always greater than or equal to the rank of the same model's score within 168 scores. This results in smaller rates of FR and FA for each rank threshold for the normalized cohort results (better performance). However, the curves are similar enough that using cohort sets to reduce computation during testing appears to have no significant adverse affects on speaker verification performance.

3.5 Performance Comparison

In order to evaluate the advantages and disadvantages of our approach to the speaker verification task, it is necessary to compare our system's performance and design to those of other systems. Often, it is difficult to compare systems unequivocally because the data used to evaluate the systems and the evaluation methods may differ. In order to make somewhat meaningful comparisons, we compare our system with two other systems, described below, that also use the TIMIT corpus.

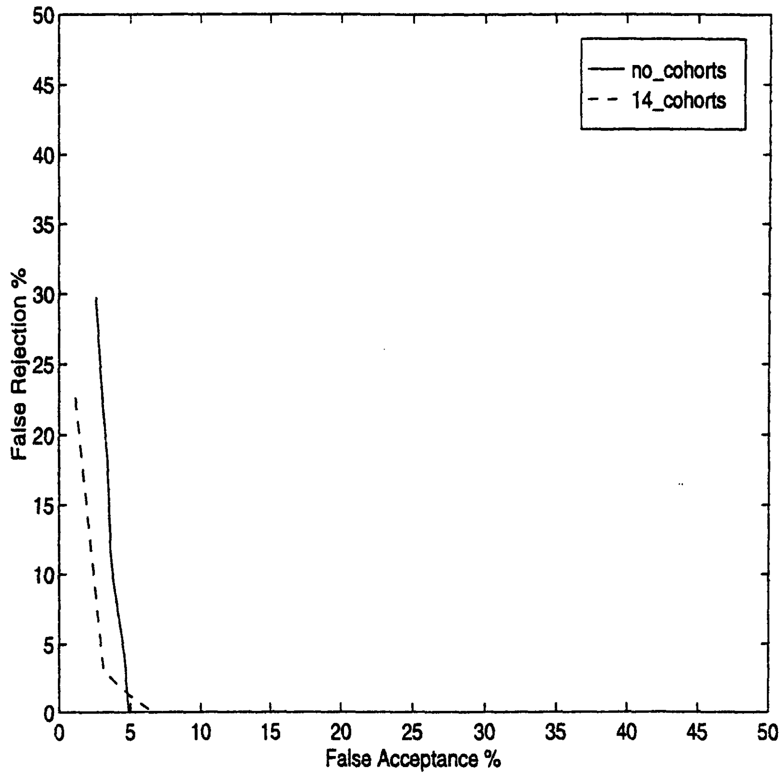

Figure 3-12: ROC Curves Using 168 Speakers and Normalizing Results Using 14 Cohorts

3.5.1 HMM Approach

A state-of-the-art HMM speech recognition system, built by Lamel and Gauvain [19], was recently modified for speaker recognition. The system extracts frame-based acoustic features, which include 15 MFCCs, first derivatives of the MFCCs, energy, and the first and second derivative of energy. During training, 8 utterances (2 SA, 3 SX and 3 SI) were used to build speaker models. To develop the speaker models, a general speaker-independent model of 40 phonetic classes was trained on the 462 speakers in the TIMIT NIST-train set. This model then served as a seed model to be used to adapt, via the maximum a posteriori procedure (MAP discussed in section 4.2.5), each speaker-specific model. During adaptation, the speaker models were modified to represent 31 broad phonetic classes, rather than 40. During testing, 168 speakers from TIMIT's NIST-test and NIST-dev sets were evaluated. The 168 speaker models were combined in parallel into one large HMM, which was used to recognize the test speech of the remaining 2 SX sentences of each user. To classify speakers, the system used the phone-based acoustic likelihoods produced by the HMM on the set of 31 broad class models. The speaker model with the highest acoustic likelihood was identified as the speaker.

Lamel and Gauvain reported 98.8% speaker identification accuracy using 1 test utterance and 100% accuracy using 2 test utterances. Since we perform mini-speaker identification tests in our system, these HMM results can be compared to our results when we use all speakers during testing. Essentially, if we were to convert the HMM speaker identification system above into a speaker verification system that implements our decision algorithm, the system achieves 0% false rejection of true users with 0% false acceptance of impostors.

Lamel's system is evaluated on the same set of 168 test speakers as our test set. However, the sentences used during testing are two SX, whereas we test each speaker using the 2 SA utterances. Unlike the SA sentences, the SX sentences are each repeated 7 times by 7 different speakers. Thus, a test sentence may be included in the training set, suggesting that the system may have seen the same sequence of phones (spoken by different speakers) in both testing and training stages. As a result, better performance may result over a system which tests completely different orthography than the training data.

3.5.2 Neural Network Approach

Another competitive system that uses the TIMIT corpus is a neural network-based speaker identification system built by Younes Bennani [2]. The system computes 16 frame-based cepstral coefficients derived from linear prediction coefficients (LPCs). Before training, acoustic vectors computed from the 5 SX utterances for 102 speakers were grouped together into homogeneous classes, via a non-supervised k-means algorithm. [2] Each of the 102 test speakers was then assigned to the class to which the majority of the speaker's acoustic vectors belonged. During training, a typology detector and a set of expert modules (neural networks), which discriminate between speakers of the same typology, were developed. During testing, 102 speakers were evaluated using 3 SI sentences. To classify speakers, a score computed from a weighting of scores of the typology detection module with those of the expert modules is used.

Bennani's neural network system achieved a performance of 100% identification accuracy. Again, if the system implements the speaker verification decision algorithm we use, it would result in 0% false rejection of true users with 0% false acceptance of impostors. Table 3-5 summarizes the design and performance of the two systems discussed above and of our system.

3.5.3 Performance versus Computational Efficiency

As illustrated above, there are high performance speaker verification systems that are evaluated on the TIMIT corpus. Although both of the systems described above use TIMIT, they evaluate the systems on either a different set of sentences, or a different

[2]Exact speaker set is not reported in reference paper, but the set is not the same as our test set of 168 speakers.

Parameter	HMM	Neural Network	SUMMIT
Speaker Models	Mixtures of Gaussians	Neural Network	Diagonal Gaussians
Classifier	HMM scores	Neural Network	forced-alignment scores
# of System Users	168	102	168
Type of Measurements	Frame-based	Frame-based	Segment-based
Model Modifications	MAP adaptation	-	-
# of Broad Classes	31	-	6
Feature Vector Size	32	16	6
Selected Features	MFCCs, delta MFCCs, Energy, and delta Energy	Cepstral Coefficients	Energy and MFCCs
Performance	0% FR, 0% FA	0% FR, 0% FA	0% FR, 4.85% FA

Table 3-5: Comparing SUMMIT to an HMM and Neural Network System

HMM	SUMMIT
$\leq$4312685	13140

Table 3-6: Number of Training Parameters

set of speakers than our sets. The different sets makes direct comparisons between our system and the two systems described above difficult. However, we may still make some meaningful comparisons concerning system design and computation.

Unlike the HMM and neural network system discussed above, our system does not achieve perfect performance (0% false rejection of true users with 4.85% false acceptance of impostors). However, performance degradation is somewhat compensated by computational efficiency. We designed a simple system and reduced computation in a variety of ways. First, we used only 6 acoustic features, as opposed to 16 or 32, to represent the speech signal. Second, we developed speaker models of 6 broad phonetic classes, as opposed to 31 for the HMM system. Third, each of the 6 broad classes is represented by a single diagonal Gaussian distribution, as opposed to mixtures of Gaussians or the nonlinear distributions that neural networks typically produce. The two latter models have more parameters to estimate, and hence require more computation during training. Finally, we reduce computation during testing by using only a set of speaker models similar to the purported speaker's model, as opposed to using all the speaker models in the system. Computation in terms of the number of training parameters during training are approximated in Table 3-6. Not enough information is given for the neural network system to approximate the number of training parameters reliably. As illustrated in Table 3-6, the HMM system estimates (with the same amount of training data) on the order of 10^6 parameters, while we estimate on the order of 10^4 parameters.

If the speaker verification application does not have serious consequences when 5% of the impostors are accepted, our method may be preferred since it is computationally efficient. However, in high security applications, a more robust system against impostors may be desired, regardless of computational expenses. Our system may improve in performance and robustness by better modeling of features, and/or better selections of broad classes, for example. Such future work is discussed in the final chapter.

Chapter 4

Conclusions & Future Work

4.1 Summary

This thesis attempted to achieve two goals. The first was to build a competitive segment-based speaker verification system, and the second goal was to build a computationally efficient system. Often, these goals cannot be achieved simultaneously. Systems that achieve 0% error may not be computationally efficient. Below, we briefly discuss how we significantly reduced computation while maintaining good speaker verification performance.

As described in section 3.2, our system achieves a performance of 0% false rejection of true users and 4.85% false acceptance of impostors. Although the false acceptance rate is not 0%, we significantly reduced computation in many ways. As previously mentioned, the system uses a small number of features, a small number of phonetic models per speaker, few model parameters, and few competing speakers during testing. We believe that the system is able to achieve good performance with a simple design because we treated speech as a concatenation of segments, rather than frames. Past observations show that speech segments carry speaker-specific information. Therefore, by considering the speech signal as a concatenation of phone-size units, we capitalized on measurements for such units more readily.

4.2 Future Work

In this section, we discuss possible future work in connection with our research. This work includes exploring robustness issues, conducting an exhaustive search for optimal acoustic features, selecting broad classes based on acoustic criteria, representing features with more complex distributions, and adapting speaker models. Finally, we plan to incorporate our speaker verification system into a web-based information access system called GALAXY.

4.2.1 Robustness Issues

An important future topic to investigate is the robustness of the system in various acoustic environments. Although we achieved good speaker verification performance on the TIMIT corpus, the training environment matched the testing environment. In reality, these two environments usually differ. For example, training data may be collected in a quiet environment over a microphone, while test data are transmitted through a noisier environment over a telephone. The noisy environment and limited bandwidth cause feature statistics to change; thus test data are mis-matched to trained models.

To appreciate the magnitude of the degradation in performance due to mismatched environments, we evaluated our system using speaker models trained on TIMIT and tested on NTIMIT. As mentioned in section 2.2, NTIMIT is TIMIT transmitted over a telephone network. Figure 4-1 gives an indication of how SV performance degrades when testing on mis-matched data. To achieve 0% false rejection of true users, the system falsely accepts 48.27% of impostors on NTIMIT test data, suggesting the necessity to design a robust system for noisy environments. Perhaps an algorithm could be adopted to re-estimate the speaker model parameters trained on clean speech to better fit noisy test data. Alternatively, it may be necessary to search for better acoustic features for the noisy environment.

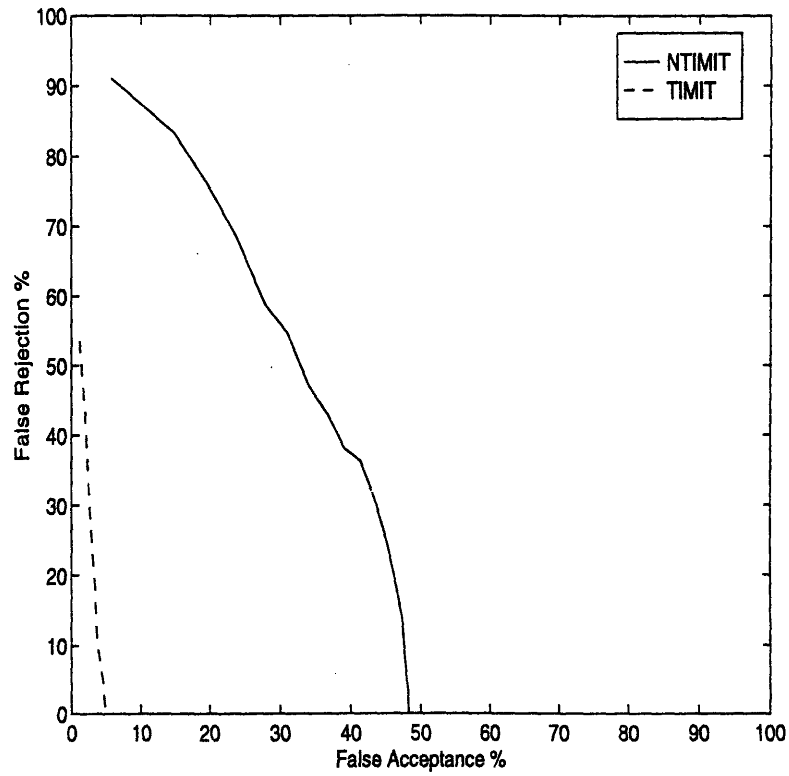

Figure 4-1: ROC curves for Optimal Feature Set: Models are trained on TIMIT data and tested on either TIMIT or NTIMIT.

4.2.2 Exhaustive Search for Robust Features

In this thesis, we conducted a greedy search with pruning for a (sub)-optimal set of acoustic features. Since we did not explore all possible feature sets formed from the 17 selected measurements, we do not know whether or not the best feature set found from the greedy search is optimal. To ensure that a feature set formed from a pool of measurements is optimal, an exhaustive search without pruning should be conducted. Optimality may be more important in domains where performance degrades significantly with different feature sets, as in noisy environments. As illustrated above, our (sub)-optimal feature set results in good performance when train and test environments are clean. However, performance degrades significantly with the same feature set, when testing in a noisy domain.

In the future, we plan to use a program called SAILS to help us extract optimal and robust features. SAILS [31] was originally used to extract optimal acoustic attributes that signify phonetic contrasts for speech recognition. It allows the user to vary parameters such as frequency range and time interval for measuring any set of features for selected speakers' phonemes, and their left and right phonetic contexts. For example, if the algorithm explores MFCCs, SAILS finds optimal places to start and end measuring the coefficients (SAILS specifies a range in the segment to compute over, such as 30%-70% of the segment), as well as which coefficients best discriminate between speakers' phonemes.

4.2.3 Feature-Motivated Broad Class Selections

As observed in this thesis, our selection of the broad manner classes affected the performance of various features, especially duration. In the past, duration has been proven to be robust and speaker-specific [41]. However, the classes we selected did not reflect different duration characteristics. As a result, the variances of duration were large for all speakers models, and the performance scores using duration ranked last in the scores for the 1-dimensional stage for both searches conducted. Thus, duration was eliminated in the search for optimal features.

In order to prevent disregarding potentially useful features for speaker verification, and to ensure that each broad class has small variances, we plan to select broad classes by using an unsupervised clustering algorithm. Unsupervised clustering algorithms, such as the K-means algorithm, group phones into classes based on acoustic characteristics. Thus, unlike the manner classes, each broad class should have similar acoustic statistics. As a result, the speaker models will have small variances for all features, which makes distinguishing between speakers easier than if the models have large variances. In turn, we hope to improve speaker verification performance.

4.2.4 Representing Features With More Complex Distributions

Future work also includes exploring more complex feature distributions than diagonal Gaussians. We chose to represent the broad class acoustic statistics with diagonal Gaussians, which have few parameters to train, to reduce computation. As a result we traded model accuracy for computation. Essentially, we forced the acoustic features for each class to be represented by a mean vector and a diagonal covariance matrix, which assumes that the features are uncorrelated random variables. Features may be more accurately modeled with mixtures of diagonal Gaussians, or full covariance Gaussians. Given enough data, more complex models may improve speaker verification performance. However, computation increases, since complex models have many parameters to estimate during training.

4.2.5 Adaptation of Models

Often, little training data are available per speaker. As a result the speaker models estimated from the data are not reliable. Ideally, one would like to obtain accurate models from little training data so that users will not be required to speak many utterances before being able to use the system. To reliably represent speakers with little training data, many investigators apply adaptation techniques to the speaker models. Specifically, the means, variances, and *a priori* broad class probabilities are typically adapted from the statistics of a well-trained speaker-independent model.

As a first attempt to observe performance effects due to adaptation of speaker models, we modified the *a priori* class probabilities of each speaker model. Specifically, we first trained a speaker-independent (SI) model using data from the 462 speakers' data from the NIST-train set of TIMIT. These estimates were then adapted to each speaker model. This simple technique forced the *a priori* estimates to be accurate and consistent across all speakers. The *a priori* class probabilities should be independent of speakers since the probability of observing a particular broad class in a segment is dependent only on the lexicon in the corpus.

Figure 4-2 illustrates the performance, before and after applying our adaptation method, evaluated on the original 168 test speakers using the optimal 6-dimensional feature set. As shown in the figure, there is no significant improvement in performance when we only adapt the *a priori* class probabilities. Perhaps the *a priori* estimates did not differ significantly from speaker to speaker before adaptation, resulting in little performance differences. The insignificant improvement after adaptation of the *a prioris* suggests that more complex adaptation techniques that modify means and variances are required to improve the speaker models, and in turn improve speaker verification performance.

In the future, we plan to implement the maximum *a-posteriori* probability (MAP) adaptation procedure, a common method for adapting all the statistics of models. MAP provides a way to incorporate prior information into the estimation process, by assuming an *a-priori* distribution of the parameters that are being estimated. Details on the MAP technique can be found in [39, 23].

4.2.6 Incorporating into GALAXY

Finally, we plan to incorporate our speaker verification system into the GALAXY conversational system [13]. GALAXY is a system currently under development in our group that enables information access using spoken dialogue. Presently, GALAXY can access the information sources on the Internet via speech for four applications: weather reports, airline travel information, automobile sales information, and the

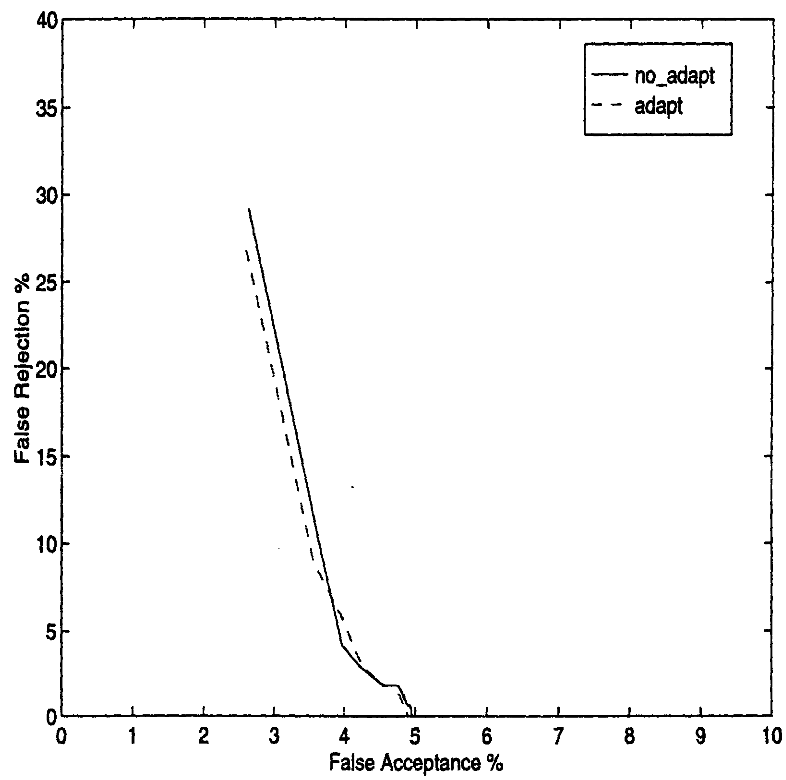

Figure 4-2: ROC curves for the Optimal Feature Set Before and After Adaptation

Boston city guide.

Appendix A

Mel-frequency Cepstral Coefficients

To extract MFCCs from speech, speech samples are initially modulated by a Hamming window of approximately 25 msec in duration. The discrete Fourier transform (DFT) of the modulated interval of speech is then computed and squared component-wise to obtain the power spectral density (PSD or energy) of the speech interval. The samples are then transformed logarithmically and filtered by the mel-frequency-based banks. These auditory triangular filter banks consist of 40 constant-area filters designed to approximate the frequency response of the human ear. The filters are on a mel-frequency scale, which is linear up to 1000 Hz and logarithmic thereafter. These filters are shown for a particular range of frequencies in Figure A-1 below.

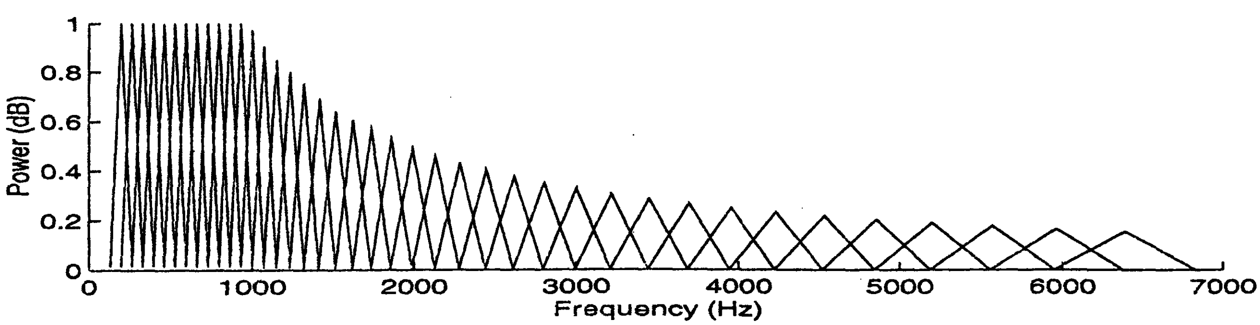

Figure A-1: MFSC Filter Banks

Collectively these coefficients form the N-dimensional mel-frequency-based spectral coefficient (MFSC) vector for the windowed speech. Finally, M (a number not necessarily equal to N) MFCCs are calculated from these spectral coefficients via the following discrete cosine transform (DCT),

$$Y_i = \sum_{k=1}^{N} X_k cos[(k - \frac{1}{2})\frac{\pi}{N}]$$

where X_k for k = 1,2,..N are the mel-frequency spectral coefficients (MFSCs), and Y_i for i = 1,2,...M are the mel-frequency cepstral coefficients (MFCCs). The details of the signal processing described above is summarized in the block diagram below. More details on computing MFCCs can be found in [26]. Some SV systems that compute MFCCs are [30, 21, 8].

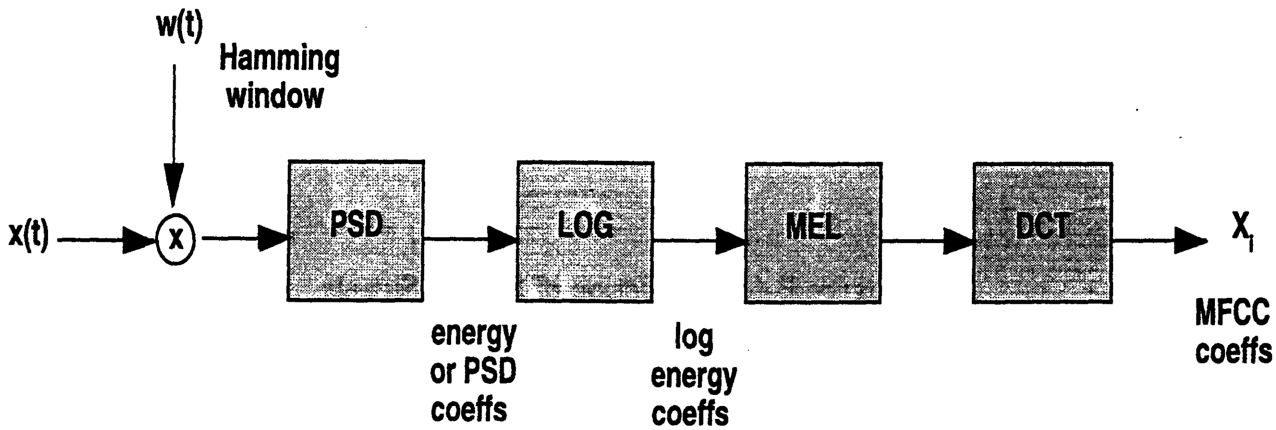

Figure A-2: Block Diagram for Computing MFCCs

Appendix B

Linear Prediction Analysis

The principles of linear prediction involve modeling the vocal tract system with an all-pole system function. The processing of a speech signal is shown in Figure B-1.

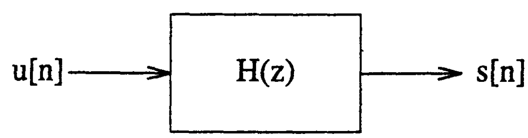

Figure B-1: Production of Speech Signals

The speech signal, shown as the output of the discrete-time system in Figure B-1, is produced by exciting the vocal tract system with a wide-band excitation $u[n]$. The vocal tract, $H(z)$, changes slowly with time, hence for short time intervals, the vocal tract can be modeled as a fixed pth-order all-pole system. Specifically,

$$H(z) = \frac{S(z)}{U(z)} = \frac{G}{1 - \sum_{k=1}^{p} a_k z^{-k}}$$

Thus, by cross multiplying and taking the inverse Bilateral z-transform of both sides, we obtain:

$$s[n] = \sum_{k=1}^{p} a_k s[n-k] + Gu[n]$$

The goal of linear prediction analysis is to estimate the a_k's and G from $s[n]$. The predicted signal is defined as:

$$\tilde{s}[n] = \sum_{k=1}^{p} \alpha_k s[n-k]$$

which leaves a residual error $e[n] = s[n] - \tilde{s}[n] \simeq Gu[n]$ (this is approximately true when the estimates α_k's are very good). The α_k's are chosen to minimize the residual error. SV systems that use LPCs for acoustic features are [18, 42, 3]. For a tutorial on LPC analysis, refer to [22].

B.1 Estimation of Fundamental Frequency

There are many methods to approximate F0, such as cepstral analysis and LPC analysis. We describe the approximation of F0 using linear prediction analysis below. Refer to [29] for the method of approximating F0 from cepstral coefficients. In order to estimate the fundamental frequency using LPC analysis, the autocorrelation of the error function is computed. During a fixed time interval of the speech signal, $s[n]$ can be assumed to be N points in length, which makes the autocorrelation function of the error, $R_e[k]$, a finite sum for each n. Specifically,

$$R_e[k] = \sum_{n=1}^{N-1-k} e[n]e[n+k]$$

When the speech signal is voiced, $u[n]$ is assumed a train of narrow glottal pulses. The signal is then windowed over an interval, and a few of the pulses remain in the interval if the window is larger than a few pulse periods. Note that the fundamental frequency is simply the reciprocal of the fundamental period of the pulses. The autocorrelation function of the residual error exhibits local maxima where the pulses occur. An example of the error autocorrelation function for a voiced time interval is illustrated in Figure B-2. These functions are plotted for every frame, and the distance between the first two peaks in $R_e[k]$ are estimates of the fundamental period (1/F0).

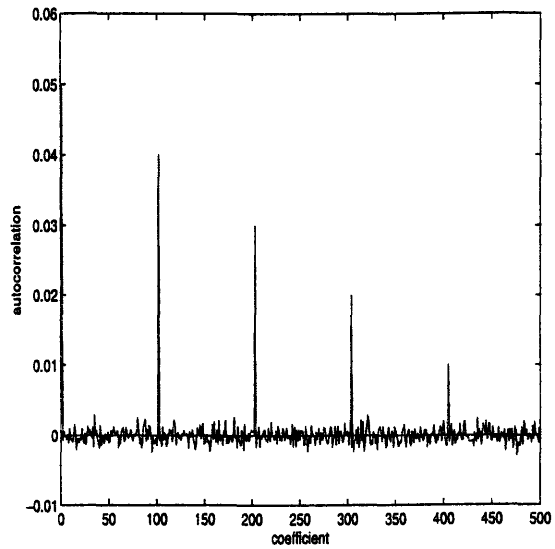

Figure B-2: The Autocorrelation Function of the Error Residual for a Short Time Interval

Appendix C

Feature Search Results

Below, the results for the 2nd-7th stages of our feature search evaluated on 168 speakers.

FEATURE SET	DISTANCE
Energy_MFCC6	57.76
Energy_MFCC10	52.72
Energy_F0	60.72
Energy_MFCC8	55.64
Energy_MFCC5	54.36
Energy_MFCC14	54.77
Energy_MFCC3	53.78
Energy_MFCC9	55.68
Energy_MFCC2	59.06

Table C-1: Two Dimensional Feature Set Results

FEATURE SET	DISTANCE
Energy_MFCC10_6	43.87
Energy_MFCC10_F0	50.25
Energy_MFCC10_8	48.49
Energy_MFCC10_5	43.04
Energy_MFCC10_14	47.23
Energy_MFCC10_3	49.20
Energy_MFCC10_9	44.13
Energy_MFCC10_2	52.39

Table C-2: Three Dimensional Feature Set Results

FEATURE SET	DISTANCE
Energy_MFCC10_5_6	42.63
Energy_MFCC10_5_F0	44.49
Energy_MFCC10_5_8	39.99
Energy_MFCC10_5_14	41.27
Energy_MFCC10_5_3	40.62
Energy_MFCC10_5_9	40.11
Energy_MFCC10_5_2	46.15

Table C-3: Four-Dimensional Feature Set Results

FEATURE SET	DISTANCE
Energy_MFCC10_5_8_6	35.97
Energy_MFCC10_5_8_F0	38.78
Energy_MFCC10_5_8_14	36.71
Energy_MFCC10_5_8_3	37.28
Energy_MFCC10_5_8_9	36.89
Energy_MFCC10_5_8_2	41.86

Table C-4: Five-Dimensional Feature Set Results

FEATURE SET	DISTANCE
Energy_MFCC10_5_8_6_F0	35.40
Energy_MFCC10_5_8_6_14	31.28
Energy_MFCC10_5_8_6_3	31.39
Energy_MFCC10_5_8_6_9	33.92
Energy_MFCC10_5_8_6_2	34.01

Table C-5: Six Dimensional Feature Set Results

FEATURE SET	DISTANCE
Energy_MFCC10_5_8_6_14_F0	33.19
Energy_MFCC10_5_8_6_14_3	31.71
Energy_MFCC10_5_8_6_14_9	32.20
Energy_MFCC10_5_8_6_14_2	34.67

Table C-6: Seven Dimensional Feature Set Results

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