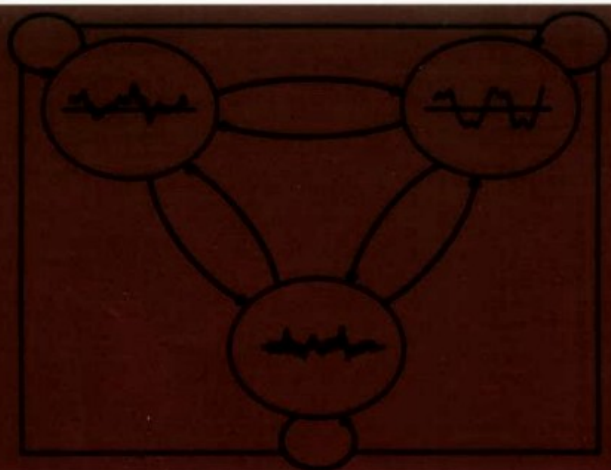


大学计算机教育丛书（影印版）

FUNDAMENTALS OF SPEECH RECOGNITION



语音识别 基本原理

*LAWRENCE RABINER
BIING-HWANG JUANG*

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Lawrence Rabiner
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PREFACE

This book is an outgrowth of an association between the authors which started over 10 years ago when one of us (BHJ) was a graduate student at the University of California at Santa Barbara and the other (LRR) was a supervisor at AT&T Bell Laboratories. We began our relationship with a mutual interest in the problem of designing and implementing vector quantization for speech processing. This association turned into a full technical partnership and strong friendship when Fred Juang joined Bell Laboratories, initially in the development area and subsequently in research. The spark that ignited formal work on this book was a series of short courses taught by one of us (LRR) on speech recognition. After several iterations of teaching, it became clear that the area of speech recognition, although still changing and growing, had matured to the point where a book that covered its theoretical underpinnings was warranted.

Once we had decided to write this book, there were several key issues that had to be resolved, including how deep to go into areas like linguistics, natural language processing, and the practical side of the problem; whether to discuss individual systems proposed by various research labs around the world; and how extensively to cover applications. Although there were no simple answers to these questions, it rapidly became obvious to us that the fundamental goal of the book would be to provide a theoretically sound, technically accurate, and reasonably complete description of the basic knowledge and ideas that constitute a modern system for speech recognition by machine. With these basic guiding principles in mind, we were able to decide consistently (and hopefully reasonably) what material had to be included, and what material would be presented in only a cursory

manner. We leave it up to you, the reader, to decide if our choices have been wise ones.

The formal organization of the book is as follows. Chapter 1, called “Fundamentals of Speech Recognition,” provides an overview of the entire field with a discussion of the breadth and depth of the various disciplines that are required for a deep understanding of all aspects of speech recognition. The concept of a task-oriented, speech-recognition system is introduced and it is shown that “base level” speech or sound recognition is only one step in a much larger process where higher-level task information, in the form of syntax, semantics, and pragmatics, can often play a major role. After a formal description of the material to be covered in each of the chapters, we give a brief history of speech recognition research in order to put the material presented in this book in its proper perspective.

Chapter 2, entitled the “The Speech Signal: Production, Perception, and Acoustic-Phonetic Characterization,” provides a review of the theory of acoustic-phonetics in which we try to characterize basic speech sounds according to both their linguistic properties and the associated acoustic measurements. We show that although there is a solid basis for the linguistic description of sounds and a good understanding of the associated acoustics of sound production, there is, at best, a tenuous relationship between a given linguistic sound and a repeatable, reliable, measureable set of acoustic parameters. As such a wide variety of approaches to speech recognition have been proposed, including those based on the ideas of acoustic-phonetics, statistical pattern-recognition methods, and artificial intelligence (so-called expert system) ideas. We discuss the relative advantages and disadvantages of each of these approaches and show why, on balance, the pattern-recognition approach has become the method of choice for most modern systems.

In Chapter 3, entitled “Signal Processing and Analysis Methods for Speech Recognition,” we discuss the fundamental techniques used to provide the speech features used in all recognition systems. In particular we discuss two well-known and widely used methods of spectrum analysis, namely the filter bank approach and the linear prediction method. We also show how the method of vector quantization can be used to code a spectral vector into one of a fixed number of discrete symbols in order to reduce the computation required in a practical system. Finally we discuss an advanced spectral analysis method that is based on processing within the human auditory system—an ear model. The ultimate goal of such a system is to increase the robustness of the signal representation and make the system relatively insensitive to noise and reverberation, in much the same way as the human ear.

Chapter 4, entitled “Pattern-Comparison Techniques,” deals with the fundamental problems of defining speech feature vector patterns (from spoken input), and comparing pairs of feature vector patterns both locally (i.e., at some point in time), and globally (i.e., over the entire pattern) so as to derive a measure of similarity between patterns. To solve this pattern-comparison problem requires three types of algorithms, namely a speech-detection method (which essentially separates the speech signal from the background), a spectral vector comparison method (which compares two individual spectral vectors), and a global pattern comparison method which aligns the two patterns locally in time and compares the aligned patterns over the entire duration of the patterns. It is shown that a key issue is the way in which time alignment between patterns is achieved.

Chapter 5, entitled “Speech Recognition System Design and Implementation Issues,” discusses the key issues of training a speech recognizer and adapting the recognizer pa-

rameters to different speakers, transmission conditions, and speaking environments. A key concept in most modern systems is that of learning, namely improving recognizer performance over time based on additional training provided by the user of the system. Adaptation methods provide a formalism for such learning.

In Chapter 6, “Theory and Implementation of Hidden Markov Models,” we discuss a basic set of statistical modeling techniques for characterizing speech. The collection of methods, popularly called Hidden Markov Models, is a powerful set of tools for providing a statistical model of both the static properties of sounds and the dynamical changes that occur across sounds. Methods for time aligning patterns with models are discussed along with different ways of building the statistical models based on the type of representation, the sound being modeled, the class of talkers, and so forth.

Chapters 7 and 8, entitled “Speech Recognition Based on Connected Word Models” and “Large Vocabulary Continuous Speech Recognition,” extend the speech-recognition problem from single word sequences to fluent speech. Modeling techniques based on whole word models are discussed in Chapter 7 where we assume that we are interested in recognizing sequences of digits, alphanumerics, and so forth. For this type of system whole-word models are most reasonable since the vocabulary is typically small and highly constrained. Hence the statistical properties of the word models, in all word contexts, can be learned from a reasonably sized training set. Modeling techniques based on subword units are discussed in Chapter 8 where we assume unlimited size vocabulary. Hence a key issue is what units are used, how context dependent the units should be, how unit models are trained reliably (and robustly to different vocabularies and tasks), and how large vocabulary recognition systems based on such units are efficiently implemented.

Finally, in Chapter 9, entitled “Task-Oriented Applications of Automatic Speech Recognition,” we come full circle and return to the concept of a task-oriented system. We discuss the basic principles that make some tasks successful while others fail. By way of example we discuss, in fairly general terms, a couple of task-oriented recognizers and show how they perform in practice.

The material in this book is primarily intended for the practicing engineer, scientist, linguist, programmer, and so forth, who wants to learn more about this fascinating field. We assume a basic knowledge of signal processing and linear system theory as provided in an entry level course in digital signal processing. Although not intended as a formal university course, the material in this book is indeed suitable for a one-semester course at the graduate or high undergraduate level. Within almost every chapter we have provided “exercises” for the student to assess how well he or she understands the material. Solutions to the exercises are provided immediately following the exercise. Hence, for maximum effectiveness, each student must exercise self-discipline to work through an answer before comparing it with the published solution.

In order to truly understand the fundamentals of speech recognition, a person needs hands-on experience with the software, hardware, and platforms. Hence we strongly encourage all serious readers of this book to program the algorithms, implement the systems, and literally build applications. Without such practical experience the words in this book will not come alive for most people.

ACKNOWLEDGMENTS

Although the authors take full responsibility for the material presented in this book, we owe a great debt to our colleagues, both with AT&T Bell Laboratories and outside, for their many technical contributions which underly the material presented. In particular the authors owe a debt of gratitude to Dr. James L. Flanagan (currently director of the CAIP Institute at Rutgers University) for his roles in guiding and shaping both our careers and the field of speech processing. Without Jim's understanding and inspiration, this book would never have existed.

The number of people who have made substantial contributions to speech recognition are too numerous to mention. However there are three individuals who have had a profound influence on the field and they deserve special mention. The first is Professor Raj Reddy of Carnegie-Mellon University who was essentially the first person to realize the vast potential of speech recognition and has devoted over 25 years as a leader, innovator, and educator in this field. The second individual of note is Dr. Jack Ferguson (retired from Institute for Defense Analyses in Princeton) who is the person most responsible for development of the theory of the Hidden Markov Model as applied to speech recognition. Dr. Ferguson, as editor of the key textbook in this area and lecturer, par excellence, has spread the word on Hidden Markov Models so that this technology has rapidly risen from technical obscurity to become the preeminent method of speech recognition today. Finally, the third individual of note is Dr. Fred Jelinek of IBM, who has led the world's largest speech-recognition research group for almost two decades and has been responsible for a large number of major innovations in large vocabulary speech recognition. These three individuals have played major roles in nurturing the technology and enabling it to reach the state of maturity it has achieved today.

Within Bell Laboratories the authors have drawn freely from the research of our former and current colleagues. We would like to acknowledge the direct support and contributions of the following individuals: Prof. Ronald Schafer (currently at Georgia Tech.), Dr. Steve Levinson, Dr. Bishnu Atal, Dr. Esther Levin, Dr. Tali Tishby (currently at the Hebrew University in Jerusalem), Dr. Oded Ghitza, Jay Wilpon, Dr. Frank Soong, Dr. Mohan Sondhi, Dr. Yariv Ephraim, Dr. Cory Myers (currently at Atlantic Aerospace), Dr. Aaron Rosenberg, Dr. Chin Lee and Dr. Roberto Pieraccini. We thank these colleagues for their research contributions and for their friendship and guidance over the years.

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The production of a book is an essentially infinite task which is seemingly an endless one. However all good things do come to an end and this one was no exception. The authors owe a great deal to Ms. Martina Sharp of the Bell Labs Word Processing group, who entered all the text material for the book using the L^AT_EX system. Tina worked on three

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Lawrence R. Rabiner
Biing-Hwang Juang

CONTENTS

LIST OF FIGURES	xiii
LIST OF TABLES	xxix
PREFACE	xxxi
1 FUNDAMENTALS OF SPEECH RECOGNITION	1
1.1 Introduction	1
1.2 The Paradigm for Speech Recognition	3
1.3 Outline	3
1.4 A Brief History of Speech-Recognition Research	6
2 THE SPEECH SIGNAL: PRODUCTION, PERCEPTION, AND ACOUSTIC-PHONETIC CHARACTERIZATION	11
2.1 Introduction	11
2.1.1 The Process of Speech Production and Perception in Human Beings	11
2.2 The Speech-Production Process	14
2.3 Representing Speech in the Time and Frequency Domains	17
2.4 Speech Sounds and Features	20

2.4.1	The Vowels	21
2.4.2	Diphthongs	28
2.4.3	Semivowels	29
2.4.4	Nasal Consonants	30
2.4.5	Unvoiced Fricatives	31
2.4.6	Voiced Fricatives	32
2.4.7	Voiced and Unvoiced Stops	33
2.4.8	Review Exercises	37
2.5	Approaches to Automatic Speech Recognition by Machine	42
2.5.1	Acoustic-Phonetic Approach to Speech Recognition	45
2.5.2	Statistical Pattern-Recognition Approach to Speech Recognition	51
2.5.3	Artificial Intelligence (AI) Approaches to Speech Recognition	52
2.5.4	Neural Networks and Their Application to Speech Recognition	54
2.6	Summary	65
3	SIGNAL PROCESSING AND ANALYSIS METHODS FOR SPEECH RECOGNITION	69
3.1	Introduction	69
3.1.1	Spectral Analysis Models	70
3.2	The Bank-of-Filters Front-End Processor	73
3.2.1	Types of Filter Bank Used for Speech Recognition	77
3.2.2	Implementations of Filter Banks	80
3.2.3	Summary of Considerations for Speech-Recognition Filter Banks	92
3.2.4	Practical Examples of Speech-Recognition Filter Banks	93
3.2.5	Generalizations of Filter-Bank Analyzer	95
3.3	Linear Predictive Coding Model for Speech Recognition	97
3.3.1	The LPC Model	100
3.3.2	LPC Analysis Equations	101
3.3.3	The Autocorrelation Method	103
3.3.4	The Covariance Method	106
3.3.5	Review Exercise	107
3.3.6	Examples of LPC Analysis	108
3.3.7	LPC Processor for Speech Recognition	112
3.3.8	Review Exercises	117
3.3.9	Typical LPC Analysis Parameters	121
3.4	Vector Quantization	122
3.4.1	Elements of a Vector Quantization Implementation	123
3.4.2	The VQ Training Set	124
3.4.3	The Similarity or Distance Measure	125
3.4.4	Clustering the Training Vectors	125
3.4.5	Vector Classification Procedure	128
3.4.6	Comparison of Vector and Scalar Quantizers	129

Contents	vii
3.4.7 Extensions of Vector Quantization	129
3.4.8 Summary of the VQ Method	131
3.5 Auditory-Based Spectral Analysis Models	132
3.5.1 The EIH Model	134
3.6 Summary	139
4 PATTERN-COMPARISON TECHNIQUES	141
4.1 Introduction	141
4.2 Speech (Endpoint) Detection	143
4.3 Distortion Measures—Mathematical Considerations	149
4.4 Distortion Measures—Perceptual Considerations	150
4.5 Spectral-Distortion Measures	154
4.5.1 Log Spectral Distance	158
4.5.2 Cepstral Distances	163
4.5.3 Weighted Cepstral Distances and Liftering	166
4.5.4 Likelihood Distortions	171
4.5.5 Variations of Likelihood Distortions	177
4.5.6 Spectral Distortion Using a Warped Frequency Scale	183
4.5.7 Alternative Spectral Representations and Distortion Measures	190
4.5.8 Summary of Distortion Measures—Computational Considerations	193
4.6 Incorporation of Spectral Dynamic Features into the Distortion Measure	194
4.7 Time Alignment and Normalization	200
4.7.1 Dynamic Programming—Basic Considerations	204
4.7.2 Time-Normalization Constraints	208
4.7.3 Dynamic Time-Warping Solution	221
4.7.4 Other Considerations in Dynamic Time Warping	229
4.7.5 Multiple Time-Alignment Paths	232
4.8 Summary	238
5 SPEECH RECOGNITION SYSTEM DESIGN AND IMPLEMENTATION ISSUES	242
5.1 Introduction	242
5.2 Application of Source-Coding Techniques to Recognition	244
5.2.1 Vector Quantization and Pattern Comparison Without Time Alignment	244
5.2.2 Centroid Computation for VQ Codebook Design	246
5.2.3 Vector Quantizers with Memory	254
5.2.4 Segmental Vector Quantization	256
5.2.5 Use of a Vector Quantizer as a Recognition Preprocessor	257
5.2.6 Vector Quantization for Efficient Pattern Matching	263
5.3 Template Training Methods	264
5.3.1 Casual Training	265

5.3.2	Robust Training	266
5.3.3	Clustering	267
5.4	Performance Analysis and Recognition Enhancements	274
5.4.1	Choice of Distortion Measures	274
5.4.2	Choice of Clustering Methods and kNN Decision Rule	277
5.4.3	Incorporation of Energy Information	280
5.4.4	Effects of Signal Analysis Parameters	282
5.4.5	Performance of Isolated Word-Recognition Systems	284
5.5	Template Adaptation to New Talkers	285
5.5.1	Spectral Transformation	286
5.5.2	Hierarchical Spectral Clustering	288
5.6	Discriminative Methods in Speech Recognition	291
5.6.1	Determination of Word Equivalence Classes	294
5.6.2	Discriminative Weighting Functions	297
5.6.3	Discriminative Training for Minimum Recognition Error	302
5.7	Speech Recognition in Adverse Environments	305
5.7.1	Adverse Conditions in Speech Recognition	306
5.7.2	Dealing with Adverse Conditions	309
5.8	Summary	317
6	THEORY AND IMPLEMENTATION OF HIDDEN MARKOV MODELS	321
6.1	Introduction	321
6.2	Discrete-Time Markov Processes	322
6.3	Extensions to Hidden Markov Models	325
6.3.1	Coin-Toss Models	326
6.3.2	The Urn-and-Ball Model	328
6.3.3	Elements of an HMM	329
6.3.4	HMM Generator of Observations	330
6.4	The Three Basic Problems for HMMs	333
6.4.1	Solution to Problem 1—Probability Evaluation	334
6.4.2	Solution to Problem 2—“Optimal” State Sequence	337
6.4.3	Solution to Problem 3—Parameter Estimation	342
6.4.4	Notes on the Reestimation Procedure	347
6.5	Types of HMMs	348
6.6	Continuous Observation Densities in HMMs	350
6.7	Autoregressive HMMs	352
6.8	Variants on HMM Structures—Null Transitions and Tied States	356
6.9	Inclusion of Explicit State Duration Density in HMMs	358
6.10	Optimization Criterion—ML, MMI, and MDI	362
6.11	Comparisons of HMMs	364
6.12	Implementation Issues for HMMs	365
6.12.1	Scaling	365
6.12.2	Multiple Observation Sequences	369
6.12.3	Initial Estimates of HMM Parameters	370

6.12.4	Effects of Insufficient Training Data	370
6.12.5	Choice of Model	371
6.13	Improving the Effectiveness of Model Estimates	372
6.13.1	Deleted Interpolation	372
6.13.2	Bayesian Adaptation	373
6.13.3	Corrective Training	376
6.14	Model Clustering and Splitting	377
6.15	HMM System for Isolated Word Recognition	378
6.15.1	Choice of Model Parameters	379
6.15.2	Segmental K-Means Segmentation into States	382
6.15.3	Incorporation of State Duration into the HMM	384
6.15.4	HMM Isolated-Digit Performance	385
6.16	Summary	386
7	SPEECH RECOGNITION BASED ON CONNECTED WORD MODELS	390
7.1	Introduction	390
7.2	General Notation for the Connected Word-Recognition Problem	393
7.3	The Two-Level Dynamic Programming (Two-Level DP) Algorithm	395
7.3.1	Computation of the Two-Level DP Algorithm	399
7.4	The Level Building (LB) Algorithm	400
7.4.1	Mathematics of the Level Building Algorithm	401
7.4.2	Multiple Level Considerations	405
7.4.3	Computation of the Level Building Algorithm	407
7.4.4	Implementation Aspects of Level Building	410
7.4.5	Integration of a Grammar Network	414
7.4.6	Examples of LB Computation of Digit Strings	416
7.5	The One-Pass (One-State) Algorithm	416
7.6	Multiple Candidate Strings	420
7.7	Summary of Connected Word Recognition Algorithms	423
7.8	Grammar Networks for Connected Digit Recognition	425
7.9	Segmental K-Means Training Procedure	427
7.10	Connected Digit Recognition Implementation	428
7.10.1	HMM-Based System for Connected Digit Recognition	429
7.10.2	Performance Evaluation on Connected Digit Strings	430
7.11	Summary	432
8	LARGE VOCABULARY CONTINUOUS SPEECH RECOGNITION	434
8.1	Introduction	434
8.2	Subword Speech Units	435
8.3	Subword Unit Models Based on HMMs	439
8.4	Training of Subword Units	441

8.5	Language Models for Large Vocabulary Speech Recognition	447
8.6	Statistical Language Modeling	448
8.7	Perplexity of the Language Model	449
8.8	Overall Recognition System Based on Subword Units	450
8.8.1	Control of Word Insertion/Word Deletion Rate	454
8.8.2	Task Semantics	454
8.8.3	System Performance on the Resource Management Task	454
8.9	Context-Dependent Subword Units	458
8.9.1	Creation of Context-Dependent Diphones and Triphones	460
8.9.2	Using Interword Training to Create CD Units	461
8.9.3	Smoothing and Interpolation of CD PLU Models	462
8.9.4	Smoothing and Interpolation of Continuous Densities	464
8.9.5	Implementation Issues Using CD Units	464
8.9.6	Recognition Results Using CD Units	467
8.9.7	Position Dependent Units	469
8.9.8	Unit Splitting and Clustering	470
8.9.9	Other Factors for Creating Additional Subword Units	475
8.9.10	Acoustic Segment Units	476
8.10	Creation of Vocabulary-Independent Units	477
8.11	Semantic Postprocessor for Recognition	478
8.12	Summary	478
9	TASK ORIENTED APPLICATIONS OF AUTOMATIC SPEECH RECOGNITION	482
9.1	Introduction	482
9.2	Speech-Recognizer Performance Scores	484
9.3	Characteristics of Speech-Recognition Applications	485
9.3.1	Methods of Handling Recognition Errors	486
9.4	Broad Classes of Speech-Recognition Applications	487
9.5	Command-and-Control Applications	488
9.5.1	Voice Repertory Dialer	489
9.5.2	Automated Call-Type Recognition	490
9.5.3	Call Distribution by Voice Commands	491
9.5.4	Directory Listing Retrieval	491
9.5.5	Credit Card Sales Validation	492
9.6	Projections for Speech Recognition	493
	INDEX	497

LIST OF FIGURES

1.1	General block diagram of a task-oriented speech-recognition system.	3
2.1	Schematic diagram of speech-production/speech-perception process (after Flanagan [unpublished]).	12
2.2	Alternative view of speech-production/speech-perception process (after Rabiner and Levinson [1]).	13
2.3	Mid-sagittal plane X-ray of the human vocal apparatus (after Flanagan et al. [2]).	15
2.4	Schematic view of the human vocal mechanism (after Flanagan [3]).	16
2.5	Glottal volume velocity and resulting sound pressure at the start of a voiced sound (after Ishizaka and Flanagan [4]).	16
2.6	Schematic representation of the complete physiological mechanism of speech production (after Flanagan [3]).	17
2.7	Waveform plot of the beginning of the utterance "It's time."	18
2.8	Wideband and narrowband spectrograms and speech amplitude for the utterance "Every salt breeze comes from the sea."	19