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Fundamentals of Speaker Recognition Voice Recognition Machine Learning for Speaker Recognition Robust Speaker Recognition in Noisy Environments Information Security for Automatic Speaker Identification Automatic Speech Recognition Automatic Speech and Speaker Recognition Automatic Speech Recognition Forensic Speaker Identification Speech Processing in Mobile Environments Extraction of Prosody for Automatic Speaker, Language, Emotion and Speech Recognition Computer Speech Technology Visual Speech Recognition: Lip Segmentation and Mapping Statistical Methods for Speech Recognition Readings in Speech Recognition Robust Speaker Recognition in Noisy Environments Deep Learning for NLP and Speech Recognition Speaker Classification I The Application of Artificial Intelligence Privacy-Preserving Machine Learning for Speech Processing Robust Speech Recognition in Embedded Systems and PC Applications Speech Synthesis and Recognition Human and Automatic Speaker Recognition over Telecommunication Channels The Speaker

Identification Ability of Blind and Sighted Listeners The Art and Business of Speech Recognition Pathological Voice Analysis Forensic Speaker Recognition ICCCE 2019 Emotion Recognition using Speech Features Speech and Audio Signal Processing Comparison of Classic and Hybrid HMM Approaches to Speech Recognition Over Telephone Lines The Speaker Identification Ability of Blind and Sighted Listeners Automatic Speech Recognition on Mobile Devices and over Communication Networks Machine Learning for Speaker Recognition Encyclopedia of Biometrics Text, Speech and Dialogue Distant Speech Recognition Advances in Biometrics The Speech Chain Connectionist Speech Recognition

This book discusses large margin and kernel methods for speech and speaker recognition Speech and Speaker Recognition: Large Margin and Kernel Methods is a collation of research in the recent advances in large margin and kernel methods, as applied to the field of speech and speaker recognition. It presents theoretical and practical foundations of these methods, from support vector machines to large margin methods for structured learning. It also provides examples of large margin based acoustic modelling for continuous speech recognizers, where the grounds for practical large margin sequence learning are set. Large margin methods for discriminative language modelling and text independent speaker verification are also addressed in this book. Key Features: Provides an up-to-date snapshot of the current state of research in this field Covers important aspects of extending the binary support vector machine to speech and speaker recognition applications Discusses large margin and kernel method algorithms for sequence prediction required for acoustic modeling Reviews past and present work on discriminative training of language models, and describes different large margin algorithms for the application of part-of-speech tagging Surveys recent work on the use of kernel approaches to text-independent speaker verification, and introduces the main concepts and algorithms Surveys recent work on kernel approaches to learning a similarity matrix from data This book will be of interest to researchers, practitioners, engineers, and scientists in speech processing and machine learning fields. With the growing impact of information technology on daily life, speech is becoming increasingly important for providing a natural means of communication between humans and machines. This extensively reworked and updated new edition of Speech Synthesis and Recognition is an easy-to-read introduction to current speech technology. Aimed at advanced undergraduates and graduates in electronic engineering, computer science and information technology, the book is also relevant to professional engineers who need to understand enough about speech technology to be able to apply it successfully and to work effectively with speech experts. No advanced mathematical ability is required and no specialist prior knowledge of phonetics or of the properties of speech signals is assumed. Most people have experienced an automated speech-recognition system when calling a company. Instead of prompting callers to choose an option by entering numbers, the system asks questions and understands spoken responses. With a more advanced application, callers may feel as if they're having a conversation with another person. Not only will the system respond intelligently, its voice even has personality. The Art and Business of Speech Recognition examines both the rapid emergence and broad potential of speech-recognition applications. By explaining the nature, design, development, and use of such applications, this book addresses two particular needs: Business managers must understand the competitive advantage that speech-recognition applications provide: a more effective way to engage, serve, and retain customers over the phone. Application designers must know how to meet their most critical business goal: a satisfying

customer experience. Author Blade Kotelly illuminates these needs from the perspective of an experienced, business-focused practitioner. Among the diverse applications he's worked on, perhaps his most influential design is the flight-information system developed for United Airlines, about which Julie Vallone wrote in Investor's Business Daily "By the end of the conversation, you might want to take the voice to dinner." If dinner is the analogy, this concise book is an ideal first course. Managers will learn the potential of speech-recognition applications to reduce costs, increase customer satisfaction, enhance the company brand, and even grow revenues. Designers, especially those just beginning to work in the voice domain, will learn user-interface design principles and techniques needed to develop and deploy successful applications. The examples in the book are real, the writing is accessible and lucid, and the solutions presented are attainable today.
0321154924B12242002 When Speech and Audio Signal Processing published in 1999, it stood out from its competition in its breadth of coverage and its accessible, intutiont-based style. This book was aimed at individual students and engineers excited about the broad span of audio processing and curious to understand the available techniques. Since then, with the advent of the iPod in 2001, the field of digital

audio and music has exploded, leading to a much greater interest in the technical aspects of audio processing. This Second Edition will update and revise the original book to augment it with new material describing both the enabling technologies of digital music distribution (most significantly the MP3) and a range of exciting new research areas in automatic music content processing (such as automatic transcription, music similarity, etc.) that have emerged in the past five years, driven by the digital music revolution. New chapter topics include: Psychoacoustic Audio Coding, describing MP3 and related audio coding schemes based on psychoacoustic masking of quantization noise Music Transcription, including automatically deriving notes, beats, and chords from music signals. Music Information Retrieval, primarily focusing on audio-based genre classification, artist/style identification, and similarity estimation. Audio Source Separation, including multi-microphone beamforming, blind source separation, and the perception-inspired techniques usually referred to as Computational Auditory Scene Analysis (CASA). Forensic Speaker Recognition: Law Enforcement and Counter-Terrorism is an anthology of the research findings of 35 speaker recognition experts from around the world. The volume provides a multidimensional view of the complex science involved in determining whether a suspect’s voice truly matches forensic speech samples, collected by law enforcement and counter-terrorism agencies, that are associated with the commission of a terrorist act or other crimes. While addressing such topics as the challenges of forensic case work, handling speech signal degradation, analyzing features of speaker recognition to optimize voice verification system performance, and designing voice applications that meet the practical needs of law enforcement and counter-terrorism agencies, this material all sounds a common theme: how the rigors of forensic utility are demanding new levels of excellence in all aspects of speaker recognition. The contributors are among the most eminent scientists in speech engineering and signal processing; and their work represents such diverse countries as Switzerland, Sweden, Italy, France, Japan, India and the United States. Forensic Speaker Recognition is a useful book for forensic speech scientists, speech signal processing experts, speech system developers, criminal prosecutors and counter-terrorism intelligence officers and agents. Originally published in 1963, The Speech Chain has been regarded as the classic, easy-to-read introduction to the fundamentals and complexities of speech communication. It provides a foundation for understanding the essential aspects of linguistics, acoustics and anatomy, and explores research and development into digital processing of speech and the use of computers for the generation of artificial speech and speech recognition. This interdisciplinary account will prove invaluable to students with little or no previous exposure to the study of language. Here's a scientific look at computer-generated speech verification and identification -- its underlying technology, practical applications, and future direction. You get a solid background in voice recognition technology to help you make informed decisions on which voice recognition-based software to use in your company or organization. It is unique in its clear explanations of mathematical concepts, as well as its full-chapter presentation of the successful new Multi-Granular Segregating System for accurate, context-free speech identification. This book discusses speaker recognition methods to deal with realistic variable noisy environments. The text covers authentication systems for; robust noisy background environments, functions in real time and incorporated in mobile devices. The book focuses on different approaches to enhance the accuracy of speaker recognition in presence of varying background environments. The authors examine: (a) Feature compensation using multiple background models, (b) Feature mapping using data-driven stochastic models,

(c) Design of super vector- based GMM-SVM framework for robust speaker recognition, (d) Total variability modeling (i-vectors) in a discriminative framework and (e) Boosting method to fuse evidences from multiple SVM models. This work addresses the evaluation of the human and the automatic speaker recognition performances under different channel distortions caused by bandwidth limitation, codecs, and electro-acoustic user interfaces, among other impairments. Its main contribution is the demonstration of the benefits of communication channels of extended bandwidth, together with an insight into how speaker-specific characteristics of speech are preserved through different transmissions. It provides sufficient motivation for considering speaker recognition as a criterion for the migration from narrowband to enhanced bandwidths, such as wideband and super-wideband. The advances in computing and networking have sparked an enormous interest in deploying automatic speech recognition on mobile devices and over communication networks. This book brings together academic researchers and industrial practitioners to address the issues in this emerging realm and presents the reader with a comprehensive introduction to the subject of speech recognition in devices and networks. It covers network, distributed and embedded speech recognition systems. Learn fundamental and advanced machine learning techniques for robust speaker recognition and domain adaptation with this useful toolkit. "Robust Speech Recognition in Embedded Systems and PC Applications reviews the problems of robust speech recognition, summarizes the current state of the art of robust speech recognition, provides perspective on the state of the art, and reviews the complementary technologies (i.e., dialog and user interface) necessary to build an application.". Speech Recognition has a long history of being one of the difficult problems in Artificial Intelligence and Computer Science. As one goes from problem solving tasks such as puzzles and chess to perceptual tasks such as speech and vision, the problem characteristics change dramatically: knowledge poor to knowledge rich; low data rates to high data rates; slow response time (minutes to hours) to instantaneous response time. These characteristics taken together increase the computational complexity of the problem by several orders of magnitude. Further, speech provides a challenging task domain which embodies many of the requirements of intelligent behavior: operate in real time; exploit vast amounts of knowledge, tolerate errorful, unexpected unknown input; use symbols and abstractions; communicate in natural language and learn from the environment. Voice input to computers offers a number of advantages. It provides a natural, fast, hands free, eyes free, location free input medium. However, there are many as yet unsolved problems that prevent routine use of speech as an input device by non-experts. These include cost, real time response, speaker independence, robustness to variations such as noise, microphone, speech rate and loudness, and the ability to handle non-grammatical speech. Satisfactory solutions to each of these problems can be expected within the next decade. Recognition of unrestricted spontaneous continuous speech appears unsolvable at present. However, by the addition of simple constraints, such as clarification dialog to resolve ambiguity, we believe it will be possible to develop systems capable of accepting very large vocabulary continuous speechdictation. This book is a collection research papers and articles from the 2nd International Conference on Communications and Cyber-Physical Engineering (ICCCE – 2019), held in Pune, India in Feb 2019. Discussing the latest developments in voice and data communication engineering, cyber-physical systems, network science, communication software, image- and multimedia processing research and applications, as well as communication technologies and other related technologies, it includes contributions from both academia and industry. This book presents a unique, understandable view of machine learning using many practical examples and access to free professional software and open source code. The user-friendly software can immediately be used to apply everything you learn in the book without the need for programming. After an introduction to machine learning and artificial intelligence, the chapters in Part II present deeper explanations of machine learning algorithms, performance evaluation of machine learning models, and how to consider data in machine learning environments. In Part III the author explains automatic speech recognition, and in Part IV biometrics recognition, face- and speaker-recognition. By Part V the author can then explain machine learning by example, he offers cases from real-world applications, problems, and techniques, such as anomaly detection and root cause analyses, business process improvement, detecting and predicting diseases, recommendation AI, several engineering applications, predictive maintenance, automatically classifying datasets, dimensionality reduction, and image recognition. Finally, in Part VI he offers a detailed explanation of the AI-TOOLKIT, software he developed that allows the reader to test and study the examples in the book and the application of machine learning in professional environments. The author introduces core machine learning concepts and supports these with practical examples of their use, so professionals will appreciate his approach and use the book for self-study. It will also be useful as a supplementary resource for advanced undergraduate and graduate courses on machine learning and artificial intelligence. This book discusses speaker recognition methods to deal with realistic variable noisy environments. The text covers authentication systems for; robust noisy background environments, functions in real time and incorporated in mobile devices. The book focuses on different approaches to enhance the accuracy of speaker recognition in presence of varying background environments. The authors examine: (a) Feature compensation using multiple background models, (b) Feature mapping using data-driven stochastic models, (c) Design of super vector- based GMM-SVM framework for robust speaker recognition, (d) Total variability modeling (i-vectors) in a discriminative framework and (e) Boosting method to fuse evidences from multiple SVM models. An emerging technology, Speaker Recognition is becoming well-known for providing voice authentication over the telephone for helpdesks, call centres and other enterprise businesses for business process automation. "Fundamentals of Speaker Recognition" introduces Speaker Identification, Speaker Verification, Speaker (Audio Event)

Classification, Speaker Detection, Speaker Tracking and more. The technical problems are rigorously defined, and a complete picture is made of the relevance of the discussed algorithms and their usage in building a comprehensive Speaker Recognition System. Designed as a textbook with examples and exercises at the end of each chapter, "Fundamentals of Speaker Recognition" is suitable for advanced-level students in computer science and engineering, concentrating on biometrics, speech recognition, pattern recognition, signal processing and, specifically, speaker recognition. It is also a valuable reference for developers of commercial technology and for speech scientists. Please click on the link under "Additional Information" to view supplemental information including the Table of Contents and Index. This book provides a comprehensive overview of the recent advancement in the field of automatic speech recognition with a focus on deep learning models including deep neural networks and many of their variants. This is the first automatic speech recognition book dedicated to the deep learning approach. In addition to the rigorous mathematical treatment of the subject, the book also presents insights and theoretical foundation of a series of highly successful deep learning models. Almut Braun carried out forensic phonetic speaker identification experiments (voice lineups) with 306 lay listeners. Blind listeners significantly outperformed sighted listeners when the speech recordings were presented in studio quality. For recordings in mobile phone quality or of whispering voices, blind and sighted listeners achieved similar results. The data can be used as reference material for real cases with blind earwitnesses. Furthermore, it is discussed whether blind individuals are particularly suitable to work as forensic audio analysts for law enforcement agencies. While voice is widely used in speech recognition and speaker identification, its application in biomedical fields is much less common. This book systematically introduces the authors' research on voice analysis for biomedical applications, particularly pathological voice analysis. Firstly, it reviews the field to highlight the biomedical value of voice. It then offers a comprehensive overview of the workflow and aspects of pathological voice analysis, including voice acquisition systems, voice pitch estimation methods, glottal closure instant detection, feature extraction and learning, and the multi-audio fusion approaches. Lastly, it discusses the experimental results that have shown the superiority of these techniques. This book is useful to researchers, professionals and postgraduate students working in fields such as speech signal processing, pattern recognition, and biomedical engineering. It is also a valuable resource for those involved in interdisciplinary research. “Emotion Recognition Using Speech Features” provides coverage of emotion-specific features present in speech. The author also discusses suitable models for capturing emotion-specific information for distinguishing different emotions. The content of this book is important for designing and developing natural and sophisticated speech systems. In this Brief, Drs. Rao and Koolagudi lead a discussion of how emotion-specific information is embedded in speech and how to acquire emotion-specific knowledge using appropriate statistical models. Additionally, the authors provide information about exploiting multiple evidences derived from various features and models. The acquired emotion-specific knowledge is useful for synthesizing emotions. Features includes discussion of: • Global and local prosodic features at syllable, word and phrase levels, helpful for capturing emotion-discriminative information; • Exploiting complementary evidences obtained from excitation sources, vocal tract systems and prosodic features in order to enhance the emotion recognition performance; • Proposed multi-stage and hybrid models for improving the emotion recognition performance. This brief is for researchers working in areas related to speech-based products such as mobile phone manufacturing companies, automobile companies, and entertainment products as well as researchers involved in basic and applied speech processing research. This textbook explains Deep Learning Architecture, with applications to various NLP Tasks, including Document Classification, Machine Translation, Language Modeling, and Speech Recognition. With the widespread adoption of deep learning, natural language processing (NLP),and speech applications in many areas (including Finance, Healthcare, and Government) there is a growing need for one comprehensive resource that maps deep learning techniques to NLP and speech and provides insights into using the tools and libraries for real-world applications. Deep Learning for NLP and Speech Recognition explains recent deep learning methods applicable to NLP and speech, provides state-of-the-art approaches, and offers real-world case studies with code to provide hands-on experience. Many books focus on deep learning theory or deep learning for NLP-specific tasks while others are cookbooks for tools and libraries, but the constant flux of new algorithms, tools, frameworks, and libraries in a rapidly evolving landscape means that there are few available texts that offer the material in this book. The book is organized into three parts, aligning to different groups of readers and their expertise. The three parts are: Machine Learning, NLP, and Speech Introduction The first part has three chapters that introduce readers to the fields of NLP, speech recognition, deep learning and machine learning with basic theory and hands-on case studies using Python-based tools and libraries. Deep Learning Basics The five chapters in the second part introduce deep learning and various topics that are crucial for speech and text processing, including word embeddings, convolutional neural networks, recurrent neural networks and speech recognition basics. Theory, practical tips, state-of-the-art methods, experimentations and analysis in using the methods discussed in theory on real-world tasks. Advanced Deep Learning Techniques for Text and Speech The third part has five chapters that discuss the latest and cutting-edge research in the areas of deep learning that intersect with NLP and speech. Topics including attention mechanisms, memory augmented networks, transfer learning, multi-task learning, domain adaptation, reinforcement learning, and end-to-end deep learning for speech recognition are covered using case studies. This book constitutes the refereed proceedings of the 10th International Conference on Text, Speech and Dialogue, TSD 2007, held in Pilsen, Czech Republic, September 3-7, 2007. The 80 revised full papers presented together with 4 invited papers were carefully reviewed and selected from 198 submissions. The papers present a wealth of state-of-the-art research results in the field of natural language processing with an emphasis on text, speech, and spoken dialogue ranging from theoretical and methodological issues to applications in various fields and with special focus on corpora, texts and transcription, speech analysis, recognition and synthesis, as well as their intertwining within NL dialogue systems. This new book is the first to provide a truly understandable, non-technical overview of all the major areas in the computer processing of human speech -- speech recognition, speech synthesis, speaker recognition, language identification, lip synchronization, and co-channel separation. It takes a unique, nonmathematical approach in exploring the nature of human language and its impact on the science and methodologies of computer voice technology. In one, easy-to-read source, you gain a deeper understanding of the fundamentals, uses, and applications of the technology itself and of the strengths and weaknesses of various systems. A time-saving glossary of technical terms is included. This book focuses on speech processing in the presence of low-bit rate coding and varying background environments. The methods presented in the book exploit the speech events which are robust in noisy environments. Accurate estimation of these crucial events will be useful for carrying out various speech tasks such as speech recognition, speaker recognition and speech rate modification in mobile environments. The authors provide insights into designing and developing robust methods to process the speech in mobile environments. Covering temporal and spectral enhancement methods to minimize the effect of noise and examining methods and models on speech and speaker recognition applications in mobile environments. With an A–Z format, this encyclopedia provides easy access to relevant information on all aspects of biometrics. It features approximately 250 overview entries and 800 definitional entries. Each entry includes a definition, key words, list of synonyms, list of related entries, illustration(s), applications, and a bibliography. Most entries include useful literature references providing the reader with a portal to more detailed information. Speech recognition by machine : a review / D.R. Reddy -- The value of speech recognition systems / W.A. Lea -- Digital representations of speech signals / R.W. Schafer and L.R. Rabiner -- Comparison of parametric representations for monosyllabic word recognition in continuously spoken sentences / S.B. Davis and P. Mermelstein -- Vector quantization / R.M. Gray -- A joint synchrony-mean-rate model of auditory speech processing / S. Seneff -- Isolated and connected word recognition : theory and selected applications / L.R. Rabiner and S.E. Levinson -- Minimum prediction residual principle applied to speech recognition / F. Itakura -- Dynamic programming algorithm optimization for spoken word recognition / S. Hakoe and S. Chiba -- Speaker-independent recognition of isolated words using clustering techniques / L.R. Rabiner [and others]Two-level DP-matching : a dynamic programming-based pattern matching algorithm for connected word recognition / H. Sakoe -- The use of a one-stage dynamic pr ... "This book introduces the readers to the various aspects of visual speech recognitions, including lip segmentation from video sequence, lip feature extraction and modeling, feature fusion and classifier design for visual speech recognition and speaker verification" résumé de l'éditeur. This thesis discusses the privacy issues in speech-based applications such as biometric authentication, surveillance, and external speech processing services. Author Manas A. Pathak presents solutions for privacy-preserving speech processing applications such as speaker verification, speaker identification and speech recognition. The author also introduces some of the tools from cryptography and machine learning and current techniques for improving the efficiency and scalability of the presented solutions. Experiments with prototype implementations of the solutions for execution time and accuracy on standardized speech datasets are also included in the text. Using the framework proposed may now make it possible for a surveillance agency to listen for a known terrorist without being able to hear conversation from non-targeted, innocent civilians. Connectionist Speech Recognition: A Hybrid Approach describes the theory and implementation of a method to incorporate neural network approaches into state of the art continuous speech recognition systems based on hidden Markov models (HMMs) to improve their performance. In this framework, neural networks (and in particular, multilayer perceptrons or MLPs) have been restricted to well-defined subtasks of the whole system, i.e. HMM emission probability estimation and feature extraction. The book describes a successful five-year international collaboration between the authors. The lessons learned form a case study that demonstrates how hybrid systems can be developed to combine neural networks with more traditional statistical approaches. The book illustrates both the advantages and limitations of neural networks in the framework of a statistical systems. Using standard databases and comparison with some conventional approaches, it is shown that MLP probability estimation can improve recognition performance. Other approaches are discussed, though there is no such unequivocal experimental result for these methods. Connectionist Speech Recognition is of use to anyone intending to use neural networks for speech recognition or within the framework provided by an existing successful statistical approach. This includes research and development groups working in the field of speech recognition, both with standard and neural network approaches, as well as other pattern recognition and/or neural network researchers. The book is also suitable as a text for advanced courses on neural networks or speech processing. A complete overview of distant automatic speech recognition The performance of conventional Automatic Speech Recognition (ASR) systems degrades dramatically as soon as the microphone is moved away from the mouth of the speaker. This is due to a broad variety of effects such as background noise, overlapping speech from other speakers, and reverberation. While traditional ASR systems underperform for speech captured with far-field sensors, there are a number of novel techniques within the recognition system as well as techniques developed in other areas of signal processing that can mitigate the deleterious effects of noise and reverberation, as well as separating speech from overlapping speakers. Distant Speech Recognitionpresents a contemporary and comprehensive description of both theoretic abstraction and practical issues inherent in the distant ASR problem. Key Features: Covers the entire topic of distant ASR and offers practical solutions to overcome the problems related to it Provides documentation and sample scripts to enable readers to construct state-of-the-art distant speech recognition systems Gives relevant background information in acoustics and filter techniques, Explains the extraction and enhancement of classification relevant speech features Describes maximum likelihood as well as discriminative parameter estimation, and maximum likelihood normalization techniques Discusses the use of multi-microphone configurations for speaker tracking and channel combination Presents several applications of the methods and technologies described in this book Accompanying website with open source software and tools to construct state-of-the-art distant speech recognition systems This reference will be an invaluable resource for researchers, developers, engineers and other professionals, as well as advanced students in speech technology, signal processing, acoustics, statistics and artificial intelligence fields. A voice is much more than just a string of words. Voices, unlike fingerprints, are inherently complex. They signal a great deal of information in addition to the intended message: the speakers' sex, for example, or their emotional state, or age. Although evidence from DNA analysis grabs the headlines, DNA can't talk. It can't be recorded planning, Almut Braun carried out forensic phonetic speaker identification experiments (voice lineups) with 306 lay listeners. Blind listeners significantly outperformed sighted listeners when the speech recordings were presented in studio quality. For recordings in mobile phone quality or of whispering voices, blind and sighted listeners achieved similar results. The data can be used as reference material for real cases with blind earwitnesses. Furthermore, it is discussed whether blind individuals are particularly suitable to work as forensic audio analysts for law enforcement agencies. This volume and its companion volume LNAI 4441 constitute a state-of-the-art survey in the field of speaker classification. Together they address such intriguing issues as how speaker characteristics are manifested in voice and speaking behavior. The nineteen contributions in this volume are organized into topical sections covering fundamentals, characteristics, applications, methods, and evaluation. rd It is a pleasure and an honour both to organize ICB 2009, the 3 IAPR/IEEE Inter- tional Conference on Biometrics. This will be held 2–5 June in Alghero, Italy, hosted by the Computer Vision Laboratory, University of Sassari. The conference series is the premier forum for presenting research in biometrics and its allied technologies: the generation of new ideas, new approaches, new techniques and new evaluations. The ICB series originated in 2006 from joining two highly reputed conferences: Audio and Video Based Personal Authentication (AVBPA) and the International Conference on Biometric Authentication (ICBA). Previous conferences were held in Hong Kong and in Korea. This is the first time the ICB conference has been held in Europe, and by Programme Committee, arrangements and by the quality of the papers, ICB 2009 will continue to maintain the high standards set by its predecessors. In total we received around 250 papers for review. Of these, 36 were selected for oral presentation and 93 for poster presentation. These papers are accompanied by the invited speakers: Heinrich H. Bülthoff (Max Planck Institute for Biological Cybernetics, Tüb- gen, Germany) on “What Can Machine Vision Learn from Human Perception?”, -daoki Furui (Department of Computer Science, Tokyo Institute of Technology) on “40 Years of Progress in Automatic Speaker Recognition Technology” and Jean-Christophe Fondeur (SAGEM Security and Morpho, USA) on “Large Scale Deployment of Biom- rics and Border Control”. This book reflects decades of important research on the mathematical foundations of speech recognition. It focuses on underlying statistical techniques such as hidden Markov models, decision trees, the expectation-maximization algorithm, information theoretic goodness criteria, maximum entropy probability estimation, parameter and data clustering, and smoothing of probability distributions. The author's goal is to present these principles clearly in the simplest setting, to show the advantages of self-organization from real data, and to enable the reader to apply the techniques. This book will help readers understand fundamental and advanced statistical models and deep learning models for robust speaker recognition and domain adaptation. This useful toolkit enables readers to apply machine learning techniques to address practical issues, such as robustness under adverse acoustic environments and domain mismatch, when deploying speaker recognition systems. Presenting state-of-the-art machine learning techniques for speaker recognition and featuring a range of probabilistic models, learning algorithms, case studies, and new trends and directions for speaker recognition based on modern machine learning and deep learning, this is the perfect resource for graduates, researchers, practitioners and engineers in electrical engineering, computer science and applied mathematics. The author covers the fundamentals of both information and communication security including current developments in some of the most critical areas of automatic speech recognition. Included are topics on speech watermarking, speech encryption, steganography, multilevel security systems comprising speaker identification, real transmission of watermarked or encrypted speech signals, and more. The book is especially useful for information security specialist, government security analysts, speech development professionals, and for individuals involved in the study and research of speech recognition at advanced levels. This updated book expands upon prosody for recognition applications of speech processing. It includes importance of prosody for speech processing applications; builds on why prosody needs to be incorporated in speech processing applications; and presents methods for extraction and representation of prosody for applications such as speaker recognition, language recognition and speech recognition. The updated book also includes information on the significance of prosody for emotion recognition and various prosody-based approaches for automatic emotion recognition from speech.