Automatic Speech Recognition On Mobile Devices And Over Communication Networks Advances In Computer Vision And Pattern Recognition

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Robust Speech Recognition of Uncertain or Missing Data Springer
Robust Automatic Speech Recognition: A Bridge to Practical Applications establishes a solid foundation for automatic speech recognition that is robust against acoustic environmental distortion. It provides a thorough overview of classical and modern noise-and reverberation robust techniques that have been developed over the past thirty years, with an emphasis on practical methods that have been proven to be successful and which are likely to be further developed for future applications. The strengths and weaknesses of robustness-enhancing speech recognition techniques are carefully analyzed. The book covers noise-robust techniques designed for acoustic models which are based on both Gaussian mixture models and deep neural networks. In addition, a guide to selecting the best methods for practical applications is provided. The reader will: Gain a unified, deep and systematic understanding of the state-of-the-art technologies for robust speech recognition Learn the links and relationship between alternative technologies for robust speech recognition Be able to use the technology analysis and categorization detailed in the book to guide future technology development Be able to develop new noise-robust methods in the current era of deep learning for acoustic modeling in speech recognition The first book that provides a comprehensive review on noise and reverberation robust speech recognition methods in the era of deep neural networks Connects robust speech recognition techniques to machine learning paradigms with rigorous mathematical treatment Provides elegant and structural ways to categorize and analyze noise-robust speech recognition techniques Written by leading researchers who have been actively working on the subject matter in both industrial and academic organizations for many years

Spoken Language Understanding John Wiley & Sons
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.

Speech Technology at Work John Wiley & Sons
This book provides a cross-disciplinary reference to speech in mobile and pervasive environments Speech in Mobile and Pervasive Environments addresses the issues related to speech processing on resource-constrained mobile devices. These include speech recognition in noisy environments, specialised hardware for speech recognition and synthesis, the use of context to enhance recognition and user experience, and the emerging software standards required for interoperability. This book takes a multi-disciplinary look at these matters, while offering an insight into the opportunities and challenges of speech processing in mobile environs. In developing regions, speech-on-mobile is set to play a momentous role, socially and economically; the authors discuss how voice-based solutions and applications offer a compelling and natural solution in this setting. Key Features Provides a holistic overview of all speech technology related topics in the context of mobility Brings together the latest research in a logically connected way in a single volume Covers hardware, embedded recognition and synthesis, distributed speech recognition, software technologies, contextual interfaces Discusses multimodal dialogue systems and their evaluation Introduces speech in mobile and pervasive environments for developing regions This book provides a comprehensive overview for beginners and experts alike. It can be used as a textbook for advanced undergraduate and postgraduate students in electrical engineering and computer science. Students, practitioners or researchers in the areas of mobile computing, speech processing, voice applications, human-computer interfaces, and information and communication technologies will also find this reference insightful. For experts in the above domains, this book complements their strengths. In addition, the book will serve as a guide to practitioners working in telecom-related industries.

Robust Automatic Speech Recognition Addison-Wesley Professional
This book constitutes of the major results of the EU COST (European Cooperation in the field of Scientific and Technical Research) Action 277: NSP, Nonlinear Speech Processing, running from April 2001 to June 2005. Coverage includes such areas as speech analysis for speech synthesis, speech recognition, speech-non speech discrimination and voice quality assessment, speech enhancement, and emotional state detection.

Discriminative Learning for Speech Recognition John Wiley & Sons
Voice user interfaces (VUIs) are becoming all the rage today. But how do you build one that people can actually converse with? Whether you're designing a mobile app, a toy, or a device such as a home assistant, this practical book guides you through basic VUI design principles, helps you choose the right speech recognition engine, and shows you how to measure your VUI's performance and improve upon it. Author Cathy Pearl also takes product managers, UX designers, and VUI designers into advanced design topics that will help make your VUI not just functional, but great.Understand key VUI design concepts, including command-and-control and conversational systemsDecide if you should use an avatar or other visual representation with your VUIExplore speech recognition technology and its impact on your designTake your VUI above and beyond the basic exchange of informationLearn practical ways to test your VUI application with usersMonitor your app and learn how to quickly improve performanceGet real-world examples of VUIs for home assistants, smartwatches, and car systems

Automatic Speech and Speaker Recognition Morgan & Claypool Publishers
Human Factors and Voice Interactive Systems highlights the importance of human factors in speech technologies and presents and demonstrates the use of human factors, principles, methods, techniques, and tools in the design of speech-enabled applications. Included is coverage of automatic speech recognition, synthetic speech, and interactive voice response systems. Some chapters are devoted to specific applications of speech technology, and other chapters are either issue-oriented or provide a comprehensive view of human factors knowledge and `lessons learned' in a specific applications area. This book places special emphasis on interactive voice response (IVR), devoting seven of its fourteen chapters to both speech-enabled and `traditional' touch-tone-based IVR applications. Other chapters emphasize speech recognition application development, natural language processing, synthetic speech, and the use of speech technology in assistive devices for people with disabilities to further the goal of universal access to information technology for all.

New Systems and Architectures for Automatic Speech Recognition and Synthesis Springer Science & Business Media
This book on Robust Speech Recognition and Understanding brings together many different aspects of the current research on automatic speech recognition and language understanding. The first four chapters address the task of voice activity detection which is considered an important issue for all speech recognition systems. The next chapters give several extensions to state-of-the-art HMM methods. Furthermore, a number of chapters particularly address the task of robust ASR under noisy conditions. Two chapters on the automatic recognition of a speaker's emotional state highlight the importance of natural speech understanding and interpretation in voice-driven systems. The last chapters of the book address the application of conversational systems on robots, as well as the autonomous acquisition of vocalization skills.
Aspects of Speech Recognition by Computer Springer Science & Business Media
Automatic speech recognition (ASR) systems are finding increasing use in everyday life. Many of the commonplace environments where the systems are used are noisy, for example users calling up a voice search system from a busy cafeteria or a street. This can result in degraded speech recordings and adversely affect the performance of speech recognition systems. As the use of ASR systems increases, knowledge of the state-of-the-art in techniques to deal with such problems becomes critical to system and application engineers and researchers who work with or on ASR technologies. This book presents a comprehensive survey of the state-of-the-art in techniques used to improve the robustness of speech recognition systems to these degrading external influences. Key features: Reviews all the main noise robust ASR approaches, including signal separation, voice activity detection, robust feature extraction, model compensation and adaptation, missing data techniques and recognition of reverberant speech. Acts as a timely exposition of the topic in light of more widespread use in the future of ASR technology in challenging environments. Addresses robustness issues and signal degradation which are both key requirements for practitioners of ASR. Includes contributions from top ASR researchers from leading research units in the field
Voice Application Development for Android Springer Science & Business Media
Automatic speech recognition (ASR) systems are finding increasing use in everyday life. Many of the commonplace environments where the systems are used are noisy, for example users calling up a voice search system from a busy cafeteria or a street. This can result in degraded speech recordings and adversely affect the performance of speech recognition systems. As the use of ASR systems increases, knowledge of the state-of-the-art in techniques to deal with such problems becomes critical to system and application engineers and researchers who work with or on ASR technologies. This book presents a comprehensive survey of the state-of-the-art in techniques used to improve the robustness of speech recognition systems to these degrading external influences. Key features: Reviews all the main noise robust ASR approaches, including signal separation, voice activity detection, robust feature extraction, model compensation and adaptation, missing data techniques and recognition of reverberant speech. Acts as a timely exposition of the topic in light of more widespread use in the future of ASR technology in challenging environments. Addresses robustness issues and signal degradation which are both key requirements for practitioners of ASR. Includes contributions from top ASR researchers from leading research units in the field
Automatic Speech Recognition BoD – Books on Demand
Speech recognition technology is being increasingly employed in human-machine interfaces. A remaining problem however is the robustness of this technology to non-native accents, which still cause considerable difficulties for current systems. In this book, methods to overcome this problem are described. A speaker adaptation algorithm that is capable of adapting to the current speaker with just a few words of speaker-specific data based on the MLLR principle is developed and combined with confidence measures that focus on phone durations as well as on acoustic features. Furthermore, a specific pronunciation modelling technique that allows the automatic derivation of non-native pronunciations without using non-native data is described and combined with the previous techniques to produce a robust adaptation to non-native accents in an automatic speech recognition system.

Information Retrieval Techniques for Speech Applications Prentice Hall
Two Top Industry Leaders Speak Out Judith Markowitz When Amy asked me to co-author the foreword to her new book on advances in speech recognition, I was honored. Amy's work has always been infused with c- ative intensity, so I knew the book would be as interesting for established speech professionals as for readers new to the speech-processing industry. The fact that I would be writing the foreword with Bill Scholz made the job even more enjoyable. Bill and I have known each other since he was at UNISYS directing projects that had a profound impact on speech-recognition tools and applications. Bill Scholz The opportunity to prepare this foreword with Judith provides me with a rare oppor- nity to collaborate with a seasoned speech professional to identify numerous signi- cant contributions to the field offered by the contributors whom Amy has recruited. Judith and I have had our eyes opened by the ideas and analyses offered by this collection of authors. Speech recognition no longer needs be relegated to the ca- gory of an experimental future technology; it is here today with sufficient capability to address the most challenging of tasks. And the point-click-type approach to GUI control is no longer sufficient, especially in the context of limitations of mode-day hand held devices. Instead, VUI and GUI are being integrated into unified multimodal solutions that are maturing into the fundamental paradigm for comput- human interaction in the future.
Multilingual Phone Recognition in Indian Languages Springer Science & Business Media
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 Recognition presents a contemporary and comprehensive description of both

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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.

Robustness in Automatic Speech Recognition Springer Nature

Research in the field of automatic speech and speaker recognition has made a number of significant advances in the last two decades, influenced by advances in signal processing, algorithms, architectures, and hardware. These advances include: the adoption of a statistical pattern recognition paradigm; the use of the hidden Markov modeling framework to characterize both the spectral and the temporal variations in the speech signal; the use of a large set of speech utterance examples from a large population of speakers to train the hidden Markov models of some fundamental speech units; the organization of speech and language knowledge sources into a structural finite state network; and the use of dynamic, programming based heuristic search methods to find the best word sequence in the lexical network corresponding to the spoken utterance. Automatic Speech and Speaker Recognition: Advanced Topics groups together in a single volume a number of important topics on speech and speaker recognition, topics which are of fundamental importance, but not yet covered in detail in existing textbooks. Although no explicit partition is given, the book is divided into five parts: Chapters 1-2 are devoted to technology overviews; Chapters 3-12 discuss acoustic modeling of fundamental speech units and lexical modeling of words and pronunciations; Chapters 13-15 address the issues related to flexibility and robustness; Chapter 16-18 concern the theoretical and practical issues of search; Chapters 19-20 give two examples of algorithm and implementational aspects for recognition system realization. Audience: A reference book for speech researchers and graduate students interested in pursuing potential research on the topic. May also be used as a text for advanced courses on the subject.

Advances in Speech Recognition Springer Science & Business Media

This book covers language modeling and automatic speech recognition for inflective languages (e.g. Slavic languages), which represent roughly half of the languages spoken in Europe. These languages do not perform as well as English in speech recognition systems and it is therefore harder to develop an application with sufficient quality for the end user. The authors describe the most important language features for the development of a speech recognition system. This is then presented through the analysis of errors in the system and the development of language models and their inclusion in speech recognition systems, which specifically address the errors that are relevant for targeted applications. The error analysis is done with regard to morphological characteristics of the word in the recognized sentences. The book is oriented towards speech recognition with large vocabularies and continuous and even spontaneous speech. Today such applications work with a rather small number of languages compared to the number of spoken languages.

Techniques for Noise Robustness in Automatic Speech Recognition Springer Science & Business Media

Proceedings of the NATO Advanced Study Institute on New Systems and Architecture for Automatic Speech Recognition and Synthesis, held at Bonas, Gers, France, 2-14 July 1984

Mastering Voice Interfaces Springer

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.

Automatic Speech Recognition on Mobile Devices and over Communication Networks John Wiley & Sons

This volume is based on a workshop held on September 13, 2001 in New Orleans, LA, USA as part of the 24th Annual International ACM SIGIR conference on Research and Development in Information Retrieval. The title of the workshop was: "Information Retrieval Techniques for Speech Applications." Interest in speech applications dates back a number of decades. However, it is only in the last few years that automatic speech recognition has left the con?nes of the basic research lab and become a viable commercial application. Speech recognition technology has now matured to the point where speech can be used to interact with automated phone systems, control computer programs, and even create memos and documents. Moving beyond computer control and dictation, speech recognition has the potential to dramatically change the way we create, capture, and store knowledge. Advances in speech recognition technology combined with ever decreasing storage costs and processors that double in power every eighteen months have set the stage for a whole new era of applications that treat speech in the same way that we currently treat text. The goal of this workshop was to explore the technical issues involved in a- lying information retrieval and text analysis technologies in the new application domain enabled by automatic speech recognition. These possibilities bring with them a number of issues, questions, and problems. Speech-based user interfaces create di?erent expectations for the end user, which in turn places di?erent - mands on the back-end systems that must interact with the user and interpret the user's commands. Speech recognition will never be perfect, so analyses ap- plied to the resulting transcripts must be robust in the face of recognition errors. The ability to capture speech and apply speech recognition on smaller, more - werful, pervasive devices suggests that text analysis and mining technologies can be applied in new domains never before considered.

Robust Speech Springer Science & Business Media

The book presents current research and developments in multilingual speech recognition. The author presents a Multilingual Phone Recognition System (Multi-PRS), developed using a common multilingual phone-set derived from the International Phonetic Alphabets (IPA) based transcription of six Indian languages - Kannada, Telugu, Bengali, Odia, Urdu, and Assamese. The author shows how the performance of Multi-PRS can be improved using tandem features. The book compares Monolingual Phone Recognition Systems (Mono-PRS) versus Multi-PRS and baseline versus tandem system. Methods are proposed to predict Articulatory Features (AFs) from spectral features using Deep Neural Networks (DNN). Multitask learning is explored to improve the prediction accuracy of AFs. Then, the AFs are explored to improve the performance of Multi-PRS using lattice rescoring method of combination and tandem method of combination. The author goes on to develop and evaluate the Language Identification followed by Monolingual phone recognition (LID-Mono) and common multilingual phone-set based multilingual phone recognition systems.

Computer Speech John Wiley & Sons

Automatic speech recognition (ASR) is a very attractive means for human-machine interaction. The degree of maturity reached by speech recognition technologies during recent years allows the development of applications that use them. In particular, ASR shows an enormous potential in mobile environments, where devices such as mobile phones or PDAs are used, and for Internet Protocol (IP) applications. Speech Recognition Over Digital Channels is the first book of its kind to offer a complete system comprehension, addressing the topics of distributed and network-based speech recognition issues and standards, the concepts of speech processing and transmission, and system architectures and robustness. Describes the different client/server architectures for remote speech recognition systems, by means of which the client transmits speech parameters through a digital channel to a remote recognition server Focuses on robustness against both adverse acoustic environments (in the front-end) and bit errors/packet loss Discusses four ETSI standards for distributed speech recognition; the understanding of the standards and the technologies behind them Provides the necessary background for the comprehension of remote speech recognition technologies This book will appeal to a wide-ranging audience: engineers using speech recognition systems, researchers involved in ASR systems and those interested in processing and transmitting speech such as signal processing and communications communities. It will also be of interest to technical experts requiring an understanding of recognition over mobile and IP networks, and postgraduate students working on robust speech processing.

Language Modeling for Automatic Speech Recognition of Inflective Languages Academic Press

Build great voice apps of any complexity for any domain by learning both the how's and why's of voice development. In this book you'll see how we live in a golden age of voice technology and how advances in automatic speech recognition (ASR), natural language processing (NLP), and related technologies allow people to talk to machines and get reasonable responses. Today, anyone with computer access can build a working voice app. That democratization of the technology is great. But, while it's fairly easy to build a voice app that runs, it's still remarkably difficult to build a great one, one that users trust, that understands their natural ways of speaking and fulfills their needs, and that makes them want to return for more. We start with an overview of how humans and machines produce and process conversational speech, explaining how they differ from each other and from other modalities. This is the background you need to understand the consequences of each design and implementation choice as we dive into the core principles of voice interface design. We walk you through many design and development techniques, including ones that some view as advanced, but that you can implement today. We use the Google development platform and Python, but our goal is to explain the reasons behind each technique such that you can take what you learn and implement it on any platform. Readers of Mastering Voice Interfaces will come away with a solid understanding of what makes voice interfaces special, learn the core voice design principles for building great voice apps, and how to actually implement those principles to create robust apps. We've learned during many years in the voice industry that the most successful solutions are created by those who understand both the human and the technology sides of speech, and that both sides affect design and development. Because we focus on developing task-oriented voice apps for real users in the real world, you'll learn how to take your voice apps from idea through scoping, design, development, rollout, and post-deployment performance improvements, all illustrated with examples from our own voice industry experiences. What You Will Learn Create truly great voice apps that users will love and trust See how voice differs from other input and output modalities, and why that matters Discover best practices for designing conversational voice-first applications, and the consequences of design and implementation choices Implement advanced voice designs, with real-world examples you can use immediately. Verify that your app is performing well, and what to change if it doesn't Who This Book Is For Anyone curious about the real how's and why's of voice interface design and development. In particular, it's aimed at teams of developers, designers, and product owners who need a shared understanding of how to create successful voice interfaces using today's technology. We expect readers to have had some exposure to voice apps, at least as users.

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