Speech Processing Solutions

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SHEPARD MELENDEZ

Springer Handbook of Speech Processing Springer Science & Business Media
Noise is everywhere and in most applications that are related to audio and speech, such as human-machine interfaces, hands-free communications, voice over IP (VoIP), hearing aids, teleconferencing/telepresence/telecollaboration systems, and so many others, the signal of interest (usually speech) that is picked up by a microphone is generally contaminated by noise. As a result, the microphone signal has to be cleaned up with digital signal processing tools before it is stored, analyzed, transmitted, or played out. This cleaning process is often called noise reduction and this topic has attracted a considerable amount of research and engineering attention for several decades. One of the objectives of this book is to present in a common framework an overview of the state of the art of noise reduction algorithms in the single-channel (one microphone) case. The focus is on the most useful approaches, i.e., filtering techniques (in different domains) and spectral enhancement methods. The other objective of Noise Reduction in Speech Processing is to derive all these well-known techniques in a rigorous way and prove many fundamental and intuitive results often taken for granted. This book is especially written for graduate students and research engineers who work on noise reduction for speech and audio applications and want to understand the subtle mechanisms behind each approach. Many new and interesting concepts are presented in this text that we hope the readers will find useful and inspiring.

Artificial Neural Networks and Machine Learning - ICANN 2011 Springer Science & Business Media
With the Internet, the proliferation of Big Data, and autonomous systems, mankind has entered into an era of 'digital obesity'. In this century, computational intelligence, such as thinking machines, have been brought forth to process complex human problems in a wide scope of areas — from social sciences, economics and biology, medicine and social networks, to cyber security.The Handbook of Computational Intelligence (in two volumes) prompts readers to look at these problems from a non-traditional angle. It takes a step by step approach, supported by case studies, to explore the issues that have arisen in the process. The Handbook covers many classic paradigms, as well as recent achievements and future promising developments to solve some of these very complex problems. Volume one explores the subjects of fuzzy logic and systems, artificial neural networks, and learning systems. Volume two delves into evolutionary computation, hybrid systems, as well as the applications of computational intelligence in decision making, the process industry, robotics, and autonomous systems.This work is a 'one-stop-shop' for beginners, as well as an inspirational source for more advanced researchers. It is a useful resource for lecturers and learners alike.

Image, Video Processing and Analysis, Hardware, Audio, Acoustic and Speech Processing Springer Science & Business Media
This two volume set (LNCS 6791 and LNCS 6792) constitutes the refereed proceedings of the 21th International Conference on Artificial Neural Networks, ICANN 2011, held in Espoo, Finland, in June 2011. The 106 revised full or poster papers presented were carefully reviewed and selected from numerous submissions. ICANN 2011 had two basic tracks: brain-inspired computing and machine learning research, with strong cross-disciplinary interactions and applications.

Speech & Language Processing John Wiley & Sons
Tanja Schultz and Katrin Kirchhoff have compiled a comprehensive overview of speech processing from a multilingual perspective. By taking this all-inclusive approach to speech processing, the editors have included theories, algorithms, and techniques that are required to support spoken input and output in a large variety of languages. Multilingual Speech Processing presents a comprehensive introduction to research problems and solutions, both from a theoretical as well as a practical perspective, and highlights technology that incorporates the increasing necessity for multilingual applications in our global community. Current challenges of speech processing and the feasibility of sharing data and system components across different languages guide contributors in their discussions of trends, prognoses and open research issues. This includes automatic speech recognition and speech synthesis, but also speech-to-speech translation, dialog systems, automatic language identification, and handling non-native speech. The book is complemented by an overview of multilingual resources, important research trends, and actual speech processing systems that are being deployed in multilingual human-human and human-machine interfaces. Researchers and developers in industry and academia with different backgrounds but a common interest in multilingual speech processing will find an excellent overview of research problems and solutions detailed from theoretical and practical perspectives. State-of-the-art research with a global perspective by authors from the USA, Asia, Europe, and South Africa The only comprehensive introduction to multilingual speech processing currently available Detailed presentation of technological advances integral to security, financial, cellular and commercial applications

Progress in Nonlinear Speech Processing Springer
In the past few years we have written and edited several books in the area of acousticandspeechsignalprocessing. Thereasonbehindthisendeavoristhat there were almost no books available in the literature when we ?rst started while there was (and still is) a real need to publish manuscripts summarizing the most useful ideas, concepts, results, and state-of-the-art algorithms in this important area of research. According to all the feedback we have received so far, we can say that we were right in doing this. Recently, several other researchers have followed us in this journey and have published interesting

books with their own visions and perspectives. The idea of writing a book on Microphone Array Signal Processing comes from discussions we have had with many colleagues and friends. As a c- sequence of these discussions, we came up with the conclusion that, again, there is an urgent need for a monograph that carefully explains the theory and implementation of microphone arrays. While there are many manuscripts on antenna arrays from a narrowband perspective (narrowband signals and narrowband processing), the literature is quite scarce when it comes to s- sor arrays explained from a truly broadband perspective. Many algorithms for speech applications were simply borrowed from narrowband antenna - rays. However, a direct application of narrowband ideas to broadband speech processing may not be necessarily appropriate and can lead to many m- understandings.

Algorithms and Case Studies Springer
The book discusses receiving signals that most electrical engineers detect and study. The vast majority of signals could never be detected due to random additive signals, known as noise, that distorts them or completely overshadows them. Such examples include an audio signal of the pilot communicating with the ground over the engine noise or a bioengineer listening for a fetus' heartbeat over the mother's. The text presents the methods for extracting the desired signals from the noise. Each new development includes examples and exercises that use MATLAB to provide the answer in graphic forms for the reader's comprehension and understanding.

Advancements in Domain Adaptation for Speaker Recognition and Effective Speaker De-identification Academic Press
With the introduction of WAP in Europe and I-mode in Japan, mobile terminals took their first steps out of the world of mobile telephony and into the world of mobile data. At the same time, the shift from 2nd generation to 3rd generation cellular technology has increased the potential data rate available to mobile users by tenfold as well as shifting data transport from circuit switched to packet data. These fundamental shifts in nature and the quantity of data available to mobile users has led to an explosion in the number of applications being developed for future digital terminal devices. Though these applications are diverse they share a common need for complex Digital Signal Processing (DSP) and in most cases benefit from the use of programmable DSPs (Digital Signal Processors). * Features contributions from experts who discuss the implementation and applications of programmable DSPs * Includes detailed introductions to speech coding, speech recognition, video and audio compression, biometric identification and their application for mobile communications devices * Discusses the alternative DSP technology which is attempting to unseat the programmable DSP from the heart of tomorrow's mobile terminals * Presents innovative new applications that are waiting to be discovered in the unique environment created when mobility meets signal processing The Application of Programmable DSPs in Mobile Communications provides an excellent overview for engineers moving into the area of mobile communications or entrepreneurs looking to understand state of the art in mobile terminals. It is also a must for students and professors looking for new application areas where DSP technology is being applied.

Advances in Modern Blind Signal Separation Algorithms CRC Press
An overview on the challenging new topic of phase-aware signal processing Speech communication technology is a key factor in human-machine interaction, digital hearing aids, mobile telephony, and automatic speech/speaker recognition. With the proliferation of these applications, there is a growing requirement for advanced methodologies that can push the limits of the conventional solutions relying on processing the signal magnitude spectrum. Single-Channel Phase-Aware Signal Processing in Speech Communication provides a comprehensive guide to phase signal processing and reviews the history of phase importance in the literature, basic problems in phase processing, fundamentals of phase estimation together with several applications to demonstrate the usefulness of phase processing. Key features: Analysis of recent advances demonstrating the positive impact of phase-based processing in pushing the limits of conventional methods. Offers unique coverage of the historical context, fundamentals of phase processing and provides several examples in speech communication. Provides a detailed review of many references and discusses the existing signal processing techniques required to deal with phase information in different applications involved with speech. The book supplies various examples and MATLAB® implementations delivered within the PhaseLab toolbox. Single-Channel Phase-Aware Signal Processing in Speech Communication is a valuable single-source for students, non-expert DSP engineers, academics and graduate students.

Synapseworld Academic Press
Speech Processing, Recognition and Artificial Neural Networks contains papers from leading researchers and selected students, discussing the experiments, theories and perspectives of acoustic phonetics as well as the latest techniques in the field of spe ech science and technology. Topics covered in this book include; Fundamentals of Speech Analysis and Perceptron; Speech Processing; Stochastic Models for Speech; Auditory and Neural Network Models for Speech; Task-Oriented Applications of Automatic Speech Recognition and Synthesis.

The Application of Programmable DSPs in Mobile Communications Springer Science & Business Media
Innovations and Advances in Computer Sciences and Engineering includes a set of rigorously reviewed world-class manuscripts addressing and detailing state-of-the-art research projects in the areas of Computer Science, Software Engineering, Computer Engineering, and Systems Engineering and Sciences. Innovations and Advances in Computer Sciences and Engineering includes selected papers form the conference proceedings of the International Conference on Systems, Computing Sciences and Software Engineering (SCSS 2008) which was part of the International Joint Conferences on Computer, Information and Systems Sciences and Engineering (CISSE 2008).

Handbook On Computational Intelligence (In 2 Volumes) Plunkett Research, Ltd.
Based on a NATO Advanced Study Institute held in 1993, this book addresses recent advances in automatic speech recognition and speech coding. The book contains contributions by many of the most outstanding researchers from the best laboratories worldwide in the field. The contributions have been grouped into five parts: on acoustic modeling; language modeling; speech processing, analysis and synthesis; speech coding; and vector quantization and neural nets. For each of these topics, some of the best-known researchers were invited to give a lecture. In addition to these lectures, the topics were complemented with discussions and presentations of the work of those attending. Altogether, the reader is given a wide perspective on recent advances in the field and will be able to see the trends for future work.

Assistive Technology for Visually Impaired and Blind People John Wiley & Sons
This book constitutes the refereed

proceedings of the Second IFIP WG 5.5/SOCOLNET Doctoral Conference on Computing, Electrical and Industrial Systems, DoCEIS 2011, held in Costa de Caparica, Portugal, in February 2011. The 67 revised full papers were carefully selected from numerous submissions. They cover a wide spectrum of topics ranging from collaborative enterprise networks to microelectronics. The papers are organized in topical sections on collaborative networks, service-oriented systems, computational intelligence, robotic systems, Petri nets, sensorial and perceptional systems, sensorial systems and decision, signal processing, fault-tolerant systems, control systems, energy systems, electrical machines, and electronics.

Plunkett's Almanac of Middle Market Companies 2009 Pearson Education India

A business development tool for professionals, marketers, sales directors, consultants and strategists seeking to understand and reach middle market American companies. It covers important business sectors, from InfoTech to health care to telecommunications. Profiles of more than 500 leading US middle market companies. Includes business glossary, a listing of business contacts, indexes and database on CD-ROM.

Signals and Images World Scientific

Users of signal processing systems are never satis?ed with the system they currently use. They are constantly asking for higher quality, faster perf- mance, more comfort and lower prices. Researchers and developers should be appreciative for this attitude. It justi?es their constant e?ort for improved systems. Better knowledge about biological and physical interrelations c- ing along with more powerful technologies are their engines on the endless road to perfect systems. This book is an impressive image of this process. After “Acoustic Echo 1 and Noise Control” published in 2004 many new results lead to “Topics in 2 Acoustic Echo and Noise Control” edited in 2006 . Today – in 2008 – even morenew?ndingsandsystemscouldbecollectedinthisbook.Comparingthe contributions in both edited volumes progress in knowledge and technology becomesclearlyvisible:Blindmethodsandmultiinputsystemsreplace“h- ble” low complexity systems. The functionality of new systems is less and less limited by the processing power available under economic constraints. The editors have to thank all the authors for their contributions. They cooperated readily in our e?ort to unify the layout of the chapters, the ter- nology, and the symbols used. It was a pleasure to work with all of them. Furthermore, it is the editors concern to thank Christoph Baumann and the Springer Publishing Company for the encouragement and help in publi- ing this book.

Information, Computer and Application Engineering Springer Science & Business Media

Applied Speech Processing: Algorithms and Case Studies is concerned with supporting and enhancing the utilization of speech analytics in several systems and real-world activities, including sharing data analytics related information, creating collaboration networks between several participants, and the use of video-conferencing in different application areas. The book provides a well-standing forum to discuss the characteristics of the intelligent speech signal processing systems in different domains. The book is proposed for professionals, scientists, and engineers who are involved in new techniques of intelligent speech signal processing methods and systems. It provides an outstanding foundation for undergraduate and post-graduate students as well. Includes basics of speech data analysis and management tools with several applications, highlighting recording systems Covers different techniques of big data and Internet-of-Things in speech signal processing, including machine learning and data mining Offers a multidisciplinary view of current and future challenges in this field, with extensive case studies on the design, implementation, development and management of intelligent systems, neural networks, and related machine learning techniques for speech signal processing

Applied Speech Processing CRC Press

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.

Speech Recognition and Coding

Multilingual Speech Processing

Recent advancements in machine learning and artificial intelligence have significantly impacted the way humans interact with machines. Voice assistant based solutions are examples of emerging technology advancements that impact human-machine interaction. Since, speech is the most natural form of human communication, voice assistant devices have received wide user acceptance, and have become a pleasant way to facilitate and address everyday living needs, including access to the current news, events, etc. These voice-based technologies have been made possible through advanced robust processing of speech signals. Depending on the application, various speech processing techniques are required to achieve an effective overall robust solution. Speech recognition is required when text content of spoken words is needed; for example adding text captions to broadcast news or YouTube videos. If a service should become available based on who is interacting with the device, speaker recognition becomes a required step; for example, if an individual gains access to a data account (e.g., music, voice-mail, health or financial records), effective speaker recognition is needed for that service. Overall, a range of solutions in speech processing can be required to address an overall request. Other areas of speech processing that benefit the human-machine interaction include language/dialect recognition, speech enhancement, machine translation, speech synthesis, voice conversion, general diarization, etc. The environment where a person interacts with a device and input tools employed (such as phone or microphone) can impact performance. It is common to have intrinsic/extrinsic mismatch between train data and application data; in other words, data used for training the speech processing tasks is often different than those at the test time. These variations need to be considered while developing effective speech systems, especially when performance is impacted significantly due to mismatch conditions. In this dissertation, we study the problem of speaker recognition for domain mismatch. Recognizing the identity of a speaker is an important task in speaker-dependent applications, and providing robust performance regardless of how data is captured for model training and considering environmental/extrinsic changes within the application phase is very important. In this dissertation, we propose two categories of solutions to address the mismatch problem in speaker recognition: discriminant analysis based adaptation methods (generalized discriminant analysis-GDA, and support vector discriminant analysis-SVDA) and deep learning based adaptation technique (a-vector speaker embeddings). The proposed solutions are evaluated on NIST SRE-10, NIST SRE-16 and NIST SRE-18 tasks. The GDA and SVDA achieved 20% and 32% improvement in terms of EER for

SRE-10 task. A-Vectors with incorporating SVDA achieved up to 18% improvement over the previous best performing solution on SRE-16 task. In addition, we propose a solution for speaker de-identification task. In more detail, the first category of solutions we propose is based on domain mismatch compensation with discriminant analysis methods. Traditional speaker recognition use linear discriminant analysis to reduce the dimensionality of speaker embeddings and provide a better discriminant feature representations for speaker classes. We propose non-linear discriminant analysis to compensate for variabilities included during recording through generalized discriminant analysis. In addition, domain adaptation is also incorporated through our proposed support vector discriminant analysis method; which also provides improved discrimination by considering the boundary structure of speaker classes. The second category of solutions are based on domain mismatch compensation with deep learning approaches. We propose a deep learning based technique to compensate for unwanted directions and information included in speaker embeddings, and provide domaininvariant speaker representations. Finally, we address speaker de- identification advancements to help protect confidential speaker or text-content within a given audio stream. Taken collectively, these three domains highlight technological advancement, which strengthen and make speaker recognition more useful in commercial, personal, and governmental/society applications, which incorporate human-speech engagement. The environment where a person interacts with a device and input tools employed (such as phone or microphone) can impact performance. It is common to have intrinsic/extrinsic mismatch between train data and application data; in other words, data used for training the speech processing tasks is often different than those at the test time. These variations need to be considered while developing effective speech systems, especially when performance is impacted significantly due to mismatch conditions. In this dissertation, we study the problem of speaker recognition for domain mismatch. Recognizing the identity of a speaker is an important task in speaker-dependent applications, and providing robust performance regardless of how data is captured for model training and considering environmental/extrinsic changes within the application phase is very important. In this dissertation, we propose two categories of solutions to address the mismatch problem in speaker recognition: discriminant analysis based adaptation methods (generalized discriminant analysis-GDA, and support vector discriminant analysis-SVDA) and deep learning based adaptation technique (a-vector speaker embeddings). The proposed solutions are evaluated on NIST SRE-10, NIST SRE-16 and NIST SRE-18 tasks. The GDA and SVDA achieved 20% and 32% improvement in terms of EER for SRE-10 task. A-Vectors with incorporating SVDA achieved up to 18% improvement over the previous best performing solution on SRE-16 task. In addition, we propose a solution for speaker de-identification task. In more detail, the first category of proposed solutions we propose are based on domain mismatch compensation with discriminant analysis methods. Traditional speaker recognition use linear discriminant analysis to reduce the dimensionality of speaker embeddings and provide a better discriminant feature representations for speaker classes. We propose non-linear discriminant analysis to compensate for variabilities included during recording through generalized discriminant analysis. In addition, domain adaptation is also incorporated through our proposed support vector discriminant analysis method; which also provides improved discrimination by considering boundary structure of speaker classes. The second category of solutions are based on domain mismatch compensation with deep learning approaches. We propose a deep learning based technique to compensate for unwanted directions and information included in speaker embeddings, and provide domain-invariant speaker representations. Finally, we address speaker de-identification advancements to help protect confidential speaker or text-content within a given audio stream. Taken collectively, these three domains highlight technological advancement, which strengthen and make speaker recognition more useful in commercial, personal, and governmental/society applications, which incorporate human-speech engagement.

Business Memo from Belgium Springer Science & Business Media

Synapseworld leverages on the creative use of cutting-edge technology such as Voice Recognition and SMS technology to provide cost effective solutions to facilitate communication, information delivery and commerce exchange in both the vertical and horizontal markets. Its key products include the patent pending MediVoice for information retrieval and drug procurement solution using speech recognition technology and the Healthaxon web based claim processing solution.

Recent Advances in Nonlinear Speech Processing Springer Science & Business Media

With human-computer interactions and hands-free communications becoming overwhelmingly important in the new millennium, recent research efforts have been increasingly focusing on state-of-the-art multi-microphone signal processing solutions to improve speech intelligibility in adverse environments. One such prominent statistical signal processing technique is blind signal separation (BSS). BSS was first introduced in the early 1990s and quickly emerged as an area of intense research activity showing huge potential in numerous applications. BSS comprises the task of 'blindly' recovering a set of unknown signals, the so-called sources from their observed mixtures, based on very little to almost no prior knowledge about the source characteristics or the mixing structure. The goal of BSS is to process multi-sensory observations of an inaccessible set of signals in a manner that reveals their individual (and original) form, by exploiting the spatial and temporal diversity, readily accessible through a multi-microphone configuration. Proceeding blindly exhibits a number of advantages, since assumptions about the room configuration and the source-to-sensor geometry can be relaxed without affecting overall efficiency. This booklet investigates one of the most commercially attractive applications of BSS, which is the simultaneous recovery of signals inside a reverberant (naturally echoing) environment, using two (or more) microphones. In this paradigm, each microphone captures not only the direct contributions from each source, but also several reflected copies of the original signals at different propagation delays. These recordings are referred to as the convolutive mixtures of the original sources. The goal of this booklet in the lecture series is to provide insight on recent advances in algorithms, which are ideally suited for blind signal separation of convolutive speech mixtures. More importantly, specific emphasis is given in practical applications of the developed BSS algorithms associated with real-life scenarios. The developed algorithms are put in the context of modern DSP devices, such as hearing aids and cochlear implants, where design requirements dictate low power consumption and call for portability and compact size. Along these lines, this booklet focuses on modern BSS algorithms which address (1) the limited amount of processing power and (2) the small number of microphones available to the end-user. Table of Contents: Fundamentals of blind signal separation / Modern blind signal separation algorithms / Application of blind signal processing strategies to noise reduction for the hearing-impaired / Conclusions and future challenges / Bibliography

Privacy-Preserving Machine Learning for Speech Processing Springer Science & Business Media

Robust Speech Recognition in Embedded Systems and PC Applications provides a link between the technology and the application worlds. As speech recognition technology is now good enough for a number of applications and the core technology is well established around hidden Markov models many of the differences between systems found in the field are related to implementation variants. We distinguish between embedded systems and PC-based applications. Embedded applications are usually cost sensitive and require very simple and optimized methods to be viable. 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 while providing some perspectives, and goes over the complementary technologies that are necessary to build an application, such as dialog and user interface technologies. Robust Speech Recognition in Embedded Systems and PC Applications is divided into five chapters. The first one reviews the main difficulties encountered in automatic speech recognition when the type of communication is unknown. The second chapter focuses on environment-independent/adaptive speech recognition approaches and on the mainstream methods applicable to noise robust speech recognition. The third chapter discusses several critical technologies that contribute to making an application usable. It also provides some design recommendations on how to design prompts, generate user feedback and develop speech user interfaces. The fourth chapter reviews several techniques that are particularly useful for embedded systems or to decrease computational complexity. It also presents some case studies for embedded applications and PC-based systems. Finally, the fifth chapter provides a future outlook for robust speech recognition, emphasizing the areas that the author sees as the most promising for the future. Robust Speech Recognition in Embedded Systems and PC Applications serves as a valuable reference and although not intended as a formal University textbook, contains some material that can be used for a course at the graduate or undergraduate level. It is a good complement for the book entitled Robustness in Automatic Speech Recognition: Fundamentals and Applications co-authored by the same author.