Blind Speech Separation

On-line Blind Signal Separation to Speech Sources
Independent Component Analysis and Applications
12th International Conference, LVA/ICA 2015, Liberec, Czech Republic, August 25-28, 2015, Proceedings
Blind Convolutive Speech Separation and Dereverberation
Methods for Bilinear, Linear-quadratic and Polynomial Mixtures
Springer Handbook of Speech Processing
Blind Source Separation of Speech Signals
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2017 2nd International Conference on Communication Systems, Computing and IT Applications (CSCITA)
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Speech Separation by Humans and Machines
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Independent Component Analysis and Signal Separation

Independent Component Analysis and Blind Signal Separation
Blind Source Separation of Speech Signals Using Filter Banks
Nonnegative Matrix and Tensor Factorizations
Nonlinear Blind Source Separation and Blind Mixture Identification
Speech Enhancement Technique Based on Blind Source Separation for Far-Field Noisy Speech Recognition
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Independent Component Analysis for Audio and Biosignal Applications
Handbook of Blind Source Separation
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Time-Domain Beamforming and Blind Source Separation
User Customizable Real-time Single and Dual Microphone Speech Enhancement and Blind Speech Separation for Smartphone Hearing Aid Applications
Advances in Modern Blind Signal Separation Algorithms
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Noise Reduction in Speech Applications
Audio Signal Processing for Next-Generation Multimedia Communication Systems
Speech Input in the Car Environment

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Adaptive Blind Signal and Image Processing

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On-line Blind Signal Separation to Speech Sources John Wiley & Sons Audio Signal Processing for Next-Generation Multimedia Communication Systems presents cutting-edge digital signal processing theory and implementation techniques for problems including speech acquisition and enhancement using microphone arrays, new adaptive filtering algorithms, multichannel acoustic echo cancellation, sound source tracking and separation, audio coding, and realistic sound stage reproduction. This book's focus is almost exclusively on the processing, transmission, and presentation of audio and acoustic signals in multimedia communications for telecollaboration where immersive acoustics will play a great role in the near future.

Independent Component Analysis and Applications

John Wiley & Sons

Noise and distortion that degrade the quality of speech signals can come from any number of sources. The technology and techniques for dealing with noise are almost as numerous, but it is only recently, with the development of inexpensive digital signal processing hardware, that the implementation of the technology has become practical. Noise Reduction in Speech Applications provides a comprehensive introduction to modern techniques for removing or reducing background noise from a range of speech-related applications. Self-contained, it starts with a tutorial-style chapter of background material, then focuses on system aspects, digital algorithms, and implementation. The final section explores a variety of applications and demonstrates to potential users of the technology the results possible with the noise reduction techniques presented. The book offers chapters contributed by international experts, a practical, systems approach, and numerous references. For electrical, acoustics, signal

processing, communications, and bioengineers, Noise Reduction in Speech Applications is a valuable resource that shows you how to decide whether noise reduction will solve problems in your own systems and how to make the best use of the technologies available. 12th International Conference, LVA/ICA 2015, Liberec, Czech Republic, August 25-28, 2015, Proceedings Blind Speech Separation Blind Speech SeparationSpringer Science & Business Media *Blind Convolutive Speech Separation and Dereverberation* John Wiley & Sons
This book provides a broad survey of models and efficient algorithms for Nonnegative Matrix Factorization (NMF). This includes NMF's various extensions and modifications, especially Nonnegative Tensor Factorizations (NTF) and Nonnegative Tucker Decompositions (NTD). NMF/NTF and their extensions are increasingly used as tools in signal and image processing, and data analysis, having garnered interest due to their capability to provide new insights and relevant information about the complex latent relationships in experimental data sets. It is suggested that NMF can provide meaningful components with physical interpretations; for example, in bioinformatics, NMF and its extensions have been successfully applied to gene expression, sequence analysis, the

functional characterization of genes, clustering and text mining. As such, the authors focus on the algorithms that are most useful in practice, looking at the fastest, most robust, and suitable for large-scale models. Key features: Acts as a single source reference guide to NMF, collating information that is widely dispersed in current literature, including the authors' own recently developed techniques in the subject area. Uses generalized cost functions such as Bregman, Alpha and Beta divergences, to present practical implementations of several types of robust algorithms, in particular Multiplicative, Alternating Least Squares, Projected Gradient and Quasi Newton algorithms. Provides a comparative analysis of the different methods in order to identify approximation error and complexity. Includes pseudo codes and optimized MATLAB source codes for almost all algorithms presented in the book. The increasing interest in nonnegative matrix and tensor factorizations, as well as decompositions and sparse representation of data, will ensure that this book is essential reading for engineers, scientists, researchers, industry practitioners and graduate students across signal and image processing; neuroscience; data mining and data analysis; computer science; bioinformatics; speech processing; biomedical engineering; and multimedia.
Methods for Bilinear, Linear-quadratic and

Polynomial Mixtures
Springer Science & Business Media
"An implementation of a subband-based BSS system using DFT filter banks is described, and an adaptive algorithm tailored for subband separation is developed. Aliasing present in the filter bank (due to the non-ideal frequency response of the filters) is reduced by using an oversampled scheme. Experiments, conducted with two-input two-output BSS systems, using both subband and fullband adaptation, indicate that separation and distortion rates are similar for both systems. However, the proposed 32-subband system is approximately 10 times computationally faster than the fullband system." --
Springer Handbook of Speech Processing
Springer
A comprehensive introduction to ICA for students andpractitioners Independent Component Analysis (ICA) is one of the most excitingnew topics in fields such as neural networks, advanced statistics,and signal processing. This is the first book to provide acomprehensive introduction to this new technique complete with thefundamental mathematical background needed to understand andutilize it. It offers a general overview of the basics of ICA,important solutions and algorithms, and in-depth coverage of newapplications in image processing, telecommunications, audio signalprocessing, and more. Independent Component Analysis is

divided into four sections thatcover: * General mathematical concepts utilized in the book * The basic ICA model and its solution * Various extensions of the basic ICA model * Real-world applications for ICA models Authors Hyvarinen, Karhunen, and Oja are well known for theircontributions to the development of ICA and here cover all therelevant theory, new algorithms, and applications in variousfields. Researchers, students, and practitioners from a variety ofdisciplines will find this accessible volume both helpful andinformative.

Blind Source Separation of Speech Signals Springer Science & Business Media Independent Component Analysis (ICA) is a signal-processing method to extract independent sources given only observed data that are mixtures of the unknown sources. Recently, Blind Source Separation (BSS) by ICA has received considerable attention because of its potential signal-processing applications such as speech enhancement systems, image processing, telecommunications, medical signal processing and several data mining issues. This book brings the state-of-the-art of some of the most important current research of ICA related to Audio and Biomedical signal processing applications. The book is partly a textbook and partly a monograph. It is a textbook because it gives a detailed introduction to ICA applications. It is simultaneously a monograph because it presents several new results, concepts and further developments, which are brought together and published in the book.

Blind Speech Separation
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

2017 2nd International Conference on Communication Systems, Computing and IT Applications (CSCITA) John Wiley & Sons

This book provides the first comprehensive overview of the fascinating topic of audio source separation based on non-negative matrix factorization, deep neural networks, and sparse component analysis. The first section of the book covers single channel source separation based on non-negative matrix factorization (NMF). After an introduction to the technique, two further chapters describe separation of known sources using non-negative spectrogram factorization, and temporal NMF models. In section two, NMF methods are extended to multi-channel source separation. Section three introduces deep neural network (DNN) techniques, with chapters on multichannel and single channel separation, and a further chapter on DNN based mask estimation for monaural speech separation. In section four, sparse component analysis (SCA) is discussed, with chapters on source separation using audio directional statistics modelling, multi-microphone MMSE-based techniques and diffusion map methods. The book brings together leading researchers to provide tutorial-like and in-depth treatments on major audio source separation topics, with the objective of becoming the definitive source for a comprehensive, authoritative, and accessible treatment. This book is written for graduate students and researchers who are interested in audio source separation techniques based on NMF, DNN and SCA.

8th International Conference, ICA 2009, Paraty, Brazil, March 15-18, 2009, Proceedings

CRC Press

This book addresses the problem of separating spontaneous multi-party speech by way of microphone arrays (beamformers) and adaptive signal processing techniques. It is written is a concise manner and an effort has been made such that all presented algorithms can be straightforwardly implemented by the reader. All experimental results have been obtained with real in-car microphone recordings involving simultaneous speech of the driver and the co-driver.

Speech Separation by Humans and Machines

IntechOpen

Speech Enhancement (SE) is a vital algorithmic component in the Hearing Aid pipeline. Over the years, several algorithms have been developed to work in real-time and to improve the quality and intelligibility of speech. However, noise suppression with minimal distortion to speech is still a prime challenge that needs to be addressed. In this work, a new single microphone SE is introduced that is implemented on a smartphone to work as an assistive device to Hearing Aids via wireless connectivity. The uniqueness of the developed method is in the introduction of varying parameters that allow the smartphone user to control the amount of noise suppression and speech distortion in real-time, which allows the user to customize the perceptual audio to their preference. A super-Gaussian extension of this approach is explored and analyzed. With the recent accessibility of the two microphones on the smartphones, doors were opened to use beamformer as a pre-filtering stage to the proposed single microphone SE. Real-time blind speech separation technique is also proposed to yield superior quality for speech. Objective and subjective results show that the developed methods outperform traditional SE techniques.

Audio Source Separation
Springer Verlag
Blind source separation is a popular technique which is used in the fields of signal processing, audio, video and image processing. BSS is used to separate the mixed signals with only knowing the mixed signals and knowing very little about original signal characteristics. The separated signals should be very good approximations of the

source signals. In particular, the blind source separation algorithm tries to estimate the Mixing Matrix. In my thesis, I have studied the blind source separation of signals based on its second order statistics. The problem of blind source separation is studied considering the following cases: when the signal is modelled as non-stationary, cyclo-stationary and quasi-stationary. A closed form solution to the blind source separation of speech signals considering speech to be a quasi-stationary source is studied and implemented.
Blind Convolutive Stereo Speech Separation and Dereverberation Springer Science & Business Media
A systematic exploration of both classic and contemporaryalgorithms in blind source separation with practical casestudies The book presents an overview of Blind Source Separation, arelatively new signal processing method. Due to themultidisciplinary nature of the subject, the book has been writtenso as to appeal to an audience from very different backgrounds.Basic mathematical skills (e.g. on matrix algebra and foundationsof probability theory) are essential in order to understand thealgorithms, although the book is written in an introductory,accessible style. This book offers a general overview of the basics of BlindSource Separation, important solutions and algorithms, and in-depthcoverage of

applications in image feature extraction, remotesensing image fusion, mixed-pixel decomposition of SAR images,image object recognition fMRI medical image processing, geochemicaland geophysical data mining, mineral resources prediction andgeoanomalies information recognition. Firstly, the background andtheory basics of blind source separation are introduced, whichprovides the foundation for the following work. Matrix operation,foundations of probability theory and information theory basics areincluded here. There follows the fundamental mathematical model andfairly new but relatively established blind source separationalgorithms, such as Independent Component Analysis (ICA) and itsimproved algorithms (Fast ICA, Maximum Likelihood ICA, OvercompleteICA, Kernel ICA, Flexible ICA, Non-negative ICA, Constrained ICA,Optimised ICA). The last part of the book considers the very recentalgorithms in BSS e.g. Sparse Component Analysis (SCA) andNon-negative Matrix Factorization (NMF). Meanwhile, in-depth casesare presented for each algorithm in order to help the readerunderstand the algorithm and its application field. A systematic exploration of both classic and contemporaryalgorithms in blind source separation with practical casestudies Presents new improved algorithms aimed at differentapplications, such as image feature extraction, remote sensingimage fusion, mixed-pixel decomposition of SAR images, image objectrecognition, and MRI medical image processing With applications in geochemical and geophysical data mining,mineral resources prediction and geoanomalies informationrecognition Written by an expert team with accredited innovations in blindsource separation and its applications in natural science Accompanying website includes a software system providing codesfor most of the algorithms mentioned in the book, enhancing thelearning experience Essential reading for postgraduate students and researchersengaged in the area of signal processing, data mining, imageprocessing and recognition, information, geosciences, lifesciences. Independent Component Analysis and Signal Separation Springer Science & Business Media A strong reference on the problem of signal and speech enhancement, describing the newest developments in this exciting field. The general emphasis is on noise reduction, because of the large number of applications that can benefit from this technology. *Independent Component Analysis and Blind Signal Separation* Springer Science & Business Media Blind Source Separation intends to report the new results of the efforts on the study of Blind Source Separation (BSS). The book collects novel research ideas and some training in BSS, independent component analysis (ICA), artificial intelligence and signal processing applications. Furthermore, the research results previously scattered in many journals and conferences worldwide are methodically edited and presented in a unified form. The book is likely to be of interest to university researchers, R&D engineers and graduate students in computer science and electronics who wish to learn the core principles, methods, algorithms and applications of BSS. Dr. Ganesh R. Naik works at University of Technology, Sydney, Australia; Dr. Wenwu Wang works at University of Surrey, UK.

Blind Source Separation of Speech Signals Using Filter Banks Springer

This book is appropriate for those specializing in speech science, hearing science, neuroscience, or computer science and engineers working on applications such as automatic speech recognition, cochlear implants, hands-free telephones, sound recording, multimedia indexing and retrieval.

Nonnegative Matrix and Tensor Factorizations Springer

This book constitutes the refereed proceedings of the 6th International Conference on Independent Component Analysis and Blind Source Separation, ICA 2006, held in Charleston, SC, USA, in March 2006. The

120 revised papers presented were carefully reviewed and selected from 183 submissions. The papers are organized in topical sections on algorithms and architectures, applications, medical applications, speech and signal processing, theory, and visual and sensory processing.
Nonlinear Blind Source Separation and Blind Mixture Identification
Springer Science & Business Media
This book provides a detailed survey of the methods that were recently developed to handle advanced versions of the blind source separation problem, which involve several types of nonlinear mixtures. Another attractive feature of the book is that it is based on a coherent framework. More precisely, the authors first present a general procedure for developing blind source separation methods. Then, all reported methods are defined with respect to this procedure. This allows the reader not only to more easily follow the description of each method but also to see how these methods relate to one another. The coherence of this book also results from the fact that the same notations are used throughout the chapters for the quantities (source signals and so on) that are used in various methods. Finally, among the quite varied types of processing methods that are presented in this book, a significant part of this description is dedicated to methods based on artificial neural

networks, especially recurrent ones, which are currently of high interest to the data analysis and machine learning community in general, beyond the more specific signal processing and blind source separation communities.

Speech Enhancement Technique Based on Blind Source Separation for Far-Field Noisy Speech Recognition Springer Science & Business Media Learn the technology behind hearing aids, Siri, and Echo Audio source separation and speech enhancement aim to extract one or more source signals of interest from an audio recording involving several sound sources. These technologies are among the most studied in audio signal processing today and bear a critical role in the success of hearing aids, hands-free phones, voice command and other noise-robust audio analysis systems, and music post-production software. Research on this topic has followed three convergent paths, starting with sensor array processing, computational auditory scene analysis, and machine learning based approaches such as independent component analysis, respectively. This book is the first one to provide a comprehensive overview by presenting the common foundations and the differences between these techniques in a unified setting. Key features: Consolidated perspective on audio source separation and speech enhancement. Both historical perspective and latest advances in the field, e.g. deep neural networks. Diverse disciplines: array processing, machine learning, and statistical signal processing. Covers the most important techniques for both single-channel and multichannel processing. This book provides both introductory and advanced material suitable for people with basic knowledge of signal processing and machine learning. Thanks to its comprehensiveness, it will help students select a promising research track, researchers leverage the acquired cross-domain knowledge to design improved techniques, and engineers and developers choose the right technology for their target application scenario. It will also be useful for practitioners from other fields (e.g., acoustics, multimedia, phonetics, and musicology) willing to exploit audio source separation or speech enhancement as pre-processing tools for their own needs.

7th International Conference, ICA 2007, London, UK, September 9-12, 2007, Proceedings John Wiley & Sons

"Computer vision seeks a process that starts with a noisy, ambiguous signal from a TV camera and ends with a high-level description of discrete objects located in 3-dimensional space and identified in a human classification. This book addresses the process at several levels. First to be treated are the low-level image-processing issues of noise removaland smoothing while preserving important lines and singularities in an image. At a slightly higher level, a robust contour tracing algorithm is described that produces a cartoon of the important lines in the image. Thirdis the high-level task of reconstructing the geometry of objects in the scene. The book has two aims: to give the computer vision community a new approach to early visual processing, in the form of image segmentation that incorporates occlusion at a low level, and to introduce real computer algorithms that do a better job than what most vision programmers use currently. The algorithms are: - a nonlinear filter that reduces noise and enhances edges, - an edge detector that also finds corners and produces smoothed contours rather than bitmaps, - an algorithm for filling gaps in contours."--PUBLISHER'S WEBSITE.

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