Blind Source Separation Advances In Theory Algorithms And Applications Signals And Communication Technology

MSEC2011 is an integrated conference concentrating its focus upon Multimedia ,Software Engineering, Computing and Education. In the proceeding, you can learn much more knowledge about Multimedia, Software Engineering ,Computing and Education of researchers all around the world. The main role of the proceeding is to be used as an exchange pillar for researchers who are working in the mentioned field. In order to meet high standard of Springer, AISC series ,the organization committee has made their efforts to do the following things. Firstly, poor quality paper has been refused after reviewing course by anonymous referee experts. Secondly, periodically review meetings have been held around the reviewers about five times for exchanging reviewing suggestions. Finally, the conference organization had several preliminary sessions before the conference. Through efforts of different people and departments, the conference will be successful and fruitful.

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.

Source Separation and Machine Learning presents the fundamentals in adaptive learning algorithms for Blind Source Separation (BSS) and emphasizes the importance of machine learning perspectives. It illustrates how BSS problems are tackled through adaptive learning algorithms and model-based approaches using the latest information on mixture signals to build a BSS model that is seen as a statistical model for a whole system. Looking at different models, including independent component analysis (ICA), nonnegative matrix factorization (NMF), nonnegative tensor factorization (NTF), and deep neural network (DNN), the book addresses how they have evolved to deal with multichannel and single-channel source separation. Emphasizes the modern model-based Blind Source Separation (BSS) which closely connects the latest research topics of BSS and Machine Learning Includes coverage of Bayesian learning, sparse learning, online learning, discriminative learning and deep learning Presents a number of case studies of model-based BSS (categorizing them into four modern models - ICA, NMF, NTF and DNN), using a variety of learning algorithms that provide solutions for the construction of BSS systems

Independent Component Analysis (ICA) is a fast developing area of intense research interest. Following on from Self-Organising Neural Networks: Independent Component Analysis and Blind Signal Separation, this book reviews the significant developments of the past year. It covers topics such as the use of hidden Markov methods, the independence assumption, and topographic ICA, and includes tutorial chapters on Bayesian and variational approaches. It also provides the latest approaches to ICA problems, including an investigation into certain "hard problems" for the very first time. Comprising contributions from the most respected and innovative researchers in the field, this volume will be of interest to students and researchers in computer science and electrical engineering; research and development personnel in disciplines such as statistical modelling and data analysis; bio-informatic workers; and physicists and chemists requiring novel data analysis methods.

Wavelets and Related Geometric Multiscale Analysis

First International Conference on Computing, ICC 2019, Riyadh, Saudi Arabia, December 10–12, 2019, Proceedings, Part I

Nonlinear Source Separation

Source Separation and Machine Learning

Learning Algorithms and Applications

Proceedings of the 2011 MESC International Conference on Multimedia, Software Engineering and Computing, November 26-27, Wuhan, China

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.

Thisvolumecollectsthepaperspresentedatthe9thInternationalConferenceon Latent Variable Analysis and Signal Separation,LVA/ICA 2010. The conference was organized by INRIA, the French National Institute for Computer Science and Control,and was held in Saint-Malo, France, September 27–30,2010,at the Palais du Grand Large. Tenyearsafterthe?rstworkshoponIndependent Component Analysis(ICA) in Aussois, France, the series of ICA conferences has shown the liveliness of the community of theoreticians and practitioners working in this ?eld. While ICA and blind signal separation have become mainstream topics, new approaches have emerged to solve problems involving signal mixtures or various other types of latent variables: semi-blind models, matrix factorization using sparse com- nent analysis, non-negative matrix factorization, probabilistic latent semantic indexing, tensor decompositions, independent vector analysis, independent s-space analysis, and so on. To re?ect this evolution towards more general latent variable analysis problems in signal processing, the ICA International Steering Committee decided to rename the 9th instance of the conference LVA/ICA. From more than a hundred submitted papers, 25 were accepted as oral p- sentationsand53 asposter presentations. Thecontent ofthis volumefollowsthe conference schedule, resulting in 14 chapters. The papers collected in this v- ume demonstrate that the research activity in the ?eld continues to range from abstract concepts to the most concrete and applicable questions and consid- ations. Speech and audio, as well as biomedical applications, continue to carry the mass of the applications considered.

A complete, one-stop reference on the state of the act of unsupervised adaptive filtering While unsupervised adaptive filtering has its roots in the 1960s, more recent advances in signal processing, information theory, imaging, and remote sensing have made this a hot area for research in several diverse fields. This book brings together cutting-edge information previously available only in disparate papers and articles, presenting a thorough and integrated treatment of the two major classes of algorithms used in the field, namely, blind signal separation and blind channel equalization algorithms. Divided into two volumes for ease of presentation, this important work shows how these algorithms, although developed independently, are closely related foundations of unsupervised adaptive filtering. Through contributions by the foremost experts on the subject, the book provides an up-to-date account of research findings, explains the underlying theory, and discusses potential applications in diverse fields. More than 100 illustrations as well as case studies, appendices, and references further enhance this excellent resource. Topics in Volume I include: Neural and information-theoretic approaches to blind signal separation Models, concepts, algorithms, and performance of blind source separation Blind separation of delayed and convolved sources Blind deconvolution of multipath mixtures Applications of blind source separation Volume II: Blind Deconvolution continues coverage with blind channel equalization and its relationship to blind source separation.

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.

Methods for Bilinear, Linear-quadratic and Polynomial Mixtures

Third International Symposium on Neural Networks, Isnn 2006, Chengdu, China, May 28 - June 1, 2006, Proceedings

Independent Component Analysis and Applications

Advances in Modern Blind Signal Separation Algorithms

Audio Source Separation and Speech Enhancement

9th International Conference, LVA/ICA 2010, St. Malo, France, September 27-30, 2010, Proceedings

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.

The 8th Ibero-American Conference on Artificial Intelligence, IBERAMIA 2002, took place in Spain for the second time in 14 years; the first conference was organized in Barcelona in January 1988. The city of Seville hosted this 8th conference, giving the participants the opportunity of enjoying the richness of its historical and cultural atmosphere. Looking back over these 14 years, key aspects of the conference, such as its structure, organization, the quantity and quality of submissions, the publication policy, and the number of attendants, have significantly changed. Some data taken from IBERAMIA'88 and IBERAMIA 2002 may help to illustrate these changes. IBERAMIA'88 was planned as an initiative of three Ibero-American AI associations: the Spanish Association for AI (AEPIA), the Mexican Association for AI (SMIA), and the Portuguese Association for AI

(APIA). The conference was organized by the AEPIA staff, including the AEPIA president, José Cuena, the secretary, Felisa Verdejo, and other members of the AEPIA board. The proceedings of IBERAMIA'88 contain 22 full papers grouped into six areas: knowledge representation and reasoning, learning, AI tools, expert systems, language, and vision. Papers were written in the native languages of the participants: Spanish, Portuguese, and Catalan. Twenty extended abstracts describing ongoing projects were also included in the proceedings.

"This book provides an updated overview of signal processing applications and recent developments in EMG from a number of diverse aspects and various applications in clinical and experimental research"--Provided by publisher.

Leading experts provide the theoretical underpinnings of the subject plus tutorials on a wide range of applications, from automatic code generation to robust broadband beamforming. Emphasis on cutting-edge research and formulating problems in convex form make this an ideal textbook for advanced graduate courses and a useful self-study guide.

Advances in Theory, Algorithms and Applications

Advances in Independent Component Analysis

Speech and Audio Signal Processing

Blind Source Separation

A Tutorial Introduction

Nonnegative Matrix and Tensor Factorizations

This book constitutes the refereed proceedings of the 7th International Conference on Independent Component Analysis and Blind Source Separation, ICA 2007, held in London, UK, in September 2007. It covers algorithms and architectures, applications, medical applications, speech and signal processing, theory, and visual and sensory processing.

Edited by the people who were forerunners in creating the field, together with contributions from 34 leading international experts, this handbook provides the definitive reference on Blind Source Separation, giving a broad and comprehensive description of all the core principles and methods, numerical algorithms and major applications in the fields of telecommunications, biomedical engineering and audio, acoustic and speech processing. Going beyond a machine learning perspective, the book reflects recent results in signal processing and numerical analysis, and includes topics such as optimization criteria, mathematical tools, the design of numerical algorithms, convolutive mixtures, and time frequency approaches. This Handbook is an ideal reference for university researchers, R&D engineers and graduates wishing to learn the core principles, methods, algorithms, and applications of Blind Source Separation. Covers the principles and major techniques and methods in one book Edited by the pioneers in the field with contributions from 34 of the world's experts Describes the main existing numerical algorithms and gives practical advice on their design Covers the latest cutting edge topics: second order methods; algebraic identification of under-determined mixtures, time-frequency methods, Bayesian approaches, blind identification under non negativity approaches, semi-blind methods for communications Shows the applications of the methods to key application areas such as telecommunications, biomedical engineering, speech, acoustic, audio and music processing, while also giving a general method for developing applications

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 showi 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

A complete, one-stop reference on the state of the act of unsupervised adaptive filtering While unsupervised adaptive filtering has its roots in the 1960s, more recent advances in signal processing, information theory, imaging, and remote sensing have made th a hot area for research in several diverse fields. This book brings together cutting-edge information previously available only in disparate papers and articles, presenting a thorough and integrated treatment of the two major classes of algorithms used in the field, namely, blind signal separation and blind channel equalization algorithms. Divided into two volumes for ease of presentation, this important work shows how these algorithms, although developed independently, are closely related foundations of unsupervised adaptive filtering. Through contributions by the foremost experts on the subject, the book provides an up-to-date account of research findings, explains the underlying theory, and discusses potential applications in diverse fields. More than 100 illustrations as well as case studies, appendices, and references further enhance this excellent resource. Topics in Volume I include: * Neural and information-theoretic approaches to blind signal separation * Models, concepts, algorithms, and performance of blind source separation * Blind separation of delayed and convolved sources * Blind deconvolution of multipath mixtures * Applications of blind source separation Volume II: Blind Deconvolution continues coverage with blind channel equalization and its relationship to blind source separation.

11th international conference, ICONIP 2004, Calcutta, India, November 22-25, 2004 : proceedings

Advances in Blind Source Separation

Adaptive Blind Signal and Image Processing

Advances in Multimedia, Software Engineering and Computing Vol.1

6th International Conference, ICA 2006, Charleston, SC, USA, March 5-8, 2006, Proceedings

Advances in Variational Bayesian Nonlinear Blind Source Separation

Blind Source SeparationAdvances in Theory, Algorithms and ApplicationsSpringer

With solid theoretical foundations and numerous potential applications, Blind Signal Processing (BSP) is one of the hottest emerging areas in Signal Processing. This volume unifies and extends the theories of adaptive blind signal and image processing and provides practical and efficient algorithms for blind source separation: Independent, Principal, Minor Component Analysis, and Multichannel Blind Deconvolution (MBD) and Equalization. Containing over 1400 references and mathematical expressions Adaptive Blind Signal and Image Processing delivers an unprecedented collection of useful techniques for adaptive blind signal/image separation, extraction, decomposition and filtering of multi-variable signals and data. Offers a broad coverage of blind signal processing techniques and algorithms both from a theoretical and practical point of view Presents more than 50 simple algorithms that can be easily modified to suit the reader's specific real world problems Provides a guide to fundamental mathematics of multi-input, multi-output and multi-sensory systems Includes illustrative worked examples, computer simulations, tables, detailed graphs and conceptual models within self contained chapters to assist self study Accompanying CD-ROM features an electronic, interactive version of the book with fully coloured figures and text. C and MATLAB user-friendly software packages are also provided MATLAB is a registered trademark of The MathWorks, Inc. By providing a detailed introduction to BSP, as well as presenting new results and recent developments, this informative and inspiring work will appeal to researchers, postgraduate students, engineers and scientists working in biomedical engineering, communications, electronics, computer science, optimisations, finance, geophysics and neural networks.

Presents state-of-the-art sparse and multiscale image and signal processing with applications in astronomy, biology, MRI, media, and forensics.

In many situations found both in Nature and in human-built systems, a set of mixed signals is observed (frequently also with noise), and it is of great scientific and technological relevance to be able to isolate or separate them so that the information in each of the signals can be utilized. Blind source separation (BSS) research is one of the more interesting emerging fields now a days in the field of signal processing. It deals with the algorithms that allow the recovery of the original sources from a set of mixtures only. The adjective "blind" is applied because the purpose is to estimate the original sources without any a priori knowledge about either the sources or the mixing system. Most of the models employed in BSS assume the hypothesis about the independence of the original sources. Under this hypothesis, a BSS problem can be considered as a particular case of independent component analysis(ICA), a linear transformation technique that, starting from a multivariate representation of the data, minimizes the statistical dependence between the components of the representation. It can be claimed that most of the advances in ICA have been motivated by the search for solutions to the BSS problem and, the other way around, advances in ICA have been immediately applied to BSS. ICA and BSS algorithms start from a mixture model, whose parameters are estimated from the observed mixtures. Separation is achieved by applying the inverse mixture model to the observed signals(separating or unmixing model). Mixturem- els usually fall into three broad categories: instantaneous linear models, convolutive models and nonlinear models, the?rstone being the simplest but, in general, not near realistic applications. The development and test of the algorithms can be accomplished through synthetic data or with real-world data. Obviously, the most important aim(and most difficult) is the separation of real-world mixtures. BSS and ICA have strong relations also, apart from signal processing, with other fields such as statistics and artificial neural networks. As long as we can find a system that emits signals propagated through a mean, andthosesignalsarereceivedbyasetofsensorsandthereisaninterestinrecovering the original sources, we have a potential field of application for BSS and ICA. Inside that wide range of applications we can find, for instance: noise reduction applications, biomedical applications, audio systems, telecommunications, and many others. This volume comes out just 20 years after the first contributions in ICA and BSS 1 appeared . Therein after, the number of research groups working in ICA and BSS has been constantly growing, so that nowadays we can estimate that far more than 100 groups are researching in these fields. As proof of the recognition among the scientific community of ICA and BSS developments there have been numerous special sessions and special issues in several well- 1 J. Herault, B. Ans, "Circuits neuronaux à synapses modi?ables: décodage de messages composites para apprentissage non supervise", C.R. de l'Académie des Sciences, vol. 299, no. III-13,pp.525-528,1984

Applications to Exploratory Multi-way Data Analysis and Blind Source Separation

Advances in Neural Networks-isnn 2006

Fifth International Conference, ICA 2004, Granada, Spain, September 22-24, 2004, Proceedings

Latent Variable Analysis and Signal Separation

Independent Component Analysis and Blind Signal Separation

Advances in Data Science, Cyber Security and IT Applications

"Blind Source Separation (BSS) is a classic problem that attempts to separate unknown sources from observed mixtures. A common framework of this is the Cocktail Party problem, where multiple individuals speaking in a noisy room are recorded by a configuration of microphones, with the challenge being to split the observed signals into individual conversations. This dissertation covers the evaluation of an existing algorithm known as Cascaded ICA with Intervention Alignment (CICAIA), the establishment of a new parameter for characterizing blind mixtures, and three new algorithms, of which two are capable of separating any number of sources given more microphones than sources. Evaluating the performance of the existing algorithm CICAIA reveals that the "reverberation time" parameter most commonly used to describe a mixing environment is actually insufficient to characterize the performance of a given algorithm. This leads to the new descriptor: sparseness. Its formulation is defined, and additional evaluations show its importance in algorithm characterization, while at the same time describing the strengths and weaknesses of CICAIA-namely, it has mediocre performance of about 13 dB SIRI in good conditions, but performs better than others in high noise or low microphone spacing. Modifying the trigger for intervention alignment to a more sensitive version creates the new algorithm Cascaded ICA with Demixing Intervention (CICADI). This proves to have better performance, roughly 20 dB SIRI, in good conditions, while still performing well in adverse conditions. With three or more sources, neither CICAIA nor CICADI produce more than 5 dB of separation. A new framework is developed to separate additional sources using redundant information present in overdetermined setups. The first approach, Inter-frequency

Correlation with Microphone Diversity (ICMD), efficiently chooses a set of microphones to produce a determined mixture that provides the best separation. The microphone set is selected at one frequency, and maintained for subsequent frequencies until a new set is necessary. The second approach, ICA with Triggered Principal component analysis (ITP), extracts the principal components of the overdetermined mixture, resulting in a determined mixture with minimal noise. Both approaches utilize an efficient detection algorithm to determine when separation has failed at a given frequency, and corrects the separation at that bin before proceeding to the next. Both algorithms produce roughly 15 dB of SIRI in good conditions"--Abstract, leaf iii.

This book constitutes the proceedings of the 12th International Conference on Latent Variable Analysis and Signal Separation, LVA/ICS 2015, held in Liberec, Czech Republic, in August 2015. The 61 revised full papers presented – 29 accepted as oral presentations and 32 accepted as poster presentations – were carefully reviewed and selected from numerous submissions. Five special topics are addressed: tensor-based methods for blind signal separation; deep neural networks for supervised speech separation/enhancement; joined analysis of multiple datasets, data fusion, and related topics; advances in nonlinear blind source separation; sparse and low rank modeling for acoustic signal processing.

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.

"Blind Signal Processing: Theory and Practice" not only introduces related fundamental mathematics, but also reflects the numerous advances in the field, such as probability density estimation-based processing algorithms, underdetermined models, complex value methods, uncertainty of order in the separation of convolutive mixtures in frequency domains, and feature extraction using Independent Component Analysis (ICA). At the end of the book, results from a study conducted at Shanghai Jiao Tong University in the areas of speech signal processing, underwater signals, image feature extraction, data compression, and the like are discussed. This book will be of particular interest to advanced undergraduate students, graduate students, university instructors and research scientists in related disciplines. Xizhi Shi is a Professor at Shanghai Jiao Tong University.

Speech Enhancement

Neural information processing [electronic resource]

Applications, Challenges, and Advancements in Electromyography Signal Processing

Independent Component Analysis

Blind Speech Separation

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.

When Speech and Audio Signal Processing published in 1999,it stood out from its competition in its breadth of coverage andits accessible, intutiont-based style. This book was aimed atindividual students and engineers excited about the broad span ofaudio processing and curious to understand the availabletechniques. Since then, with the advent of the iPod in 2001,the field of digital audio and music has exploded, leading to amuch greater interest in the technical aspects of audioprocessing. This Second Edition will update and revise the originalbook to augment it with new material describing both the enablingtechnologies of digital music distribution (most significantly theMP3) and a range of exciting new research areas in automatic musiccontent processing (such as automatic transcription, musicsimilarity, etc.) that have emerged in the past five years, drivenby the digital music revolution. New chapter topics include: Psychoacoustic Audio Coding, describing MP3 and relatedaudio coding schemes based on psychoacoustic masking ofquantization noise Music Transcription, including automatically derivingnotes, beats, and chords from music signals. Music Information Retrieval, primarily focusing

onaudio-based genre classification, artist/style identification, andsimilarity estimation. Audio Source Separation, including multi-microphonebeamforming, blind source separation, and the perception-inspiredtechniques usually referred to as Computational Auditory SceneAnalysis (CASA).
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.
This is the world’s first edited book on independent component analysis (ICA)-based blind source separation (BSS) of convolutive mixtures of speech. This book brings together a small number of leading researchers to provide tutorial-like and in-depth treatment on major ICA-based BSS topics, with the objective of becoming the definitive source for current, comprehensive, authoritative, and yet accessible treatment.
Convex Optimization in Signal Processing and Communications
7th International Conference, ICA 2007, London, UK, September 9-12, 2007, Proceedings
Blind Signal Processing
Audio Signal Processing for Next-Generation Multimedia Communication Systems
Advances in Compressive Sensing and Its Application in Blind Source Separation
Theory and Practice
This book grew out of the IEEE-EMBS Summer Schools on Biomedical Signal Processing, which have been held annually since 2002 to provide the participants state-of-the-art knowledge on emerging areas in biomedical engineering. Prominent experts in the areas of biomedical signal processing, biomedical data treatment, medicine, signal processing, system biology, and applied physiology introduce novel techniques and algorithms as well as their clinical or physiological applications. The book provides an overview of a compelling group of advanced biomedical signal processing techniques, such as multisource and multiscale integration of information for physiology and clinical decision; the impact of advanced methods of signal processing in cardiology and neurology; the integration of signal processing methods with a modelling approach; complexity measurement from biomedical signals; higher order analysis in biomedical signals; advanced methods of signal and data processing in genomics and proteomics; and classification and parameter enhancement.
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.
This book constitutes the refereed proceedings of the First International Conference on Intelligent Cloud Computing, ICC 2019, held in Riyadh, Saudi Arabia, in December 2019. The two-volume set presents 53 full papers, which were carefully reviewed and selected from 174 submissions. The papers are organized in topical sections on Cyber Security; Data Science; Information Technology and Applications; Network and IoT.
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.

Processing and Perception of Speech and Music

Sparse Image and Signal Processing

Audio Source Separation

Advances in Artificial Intelligence - IBERAMIA 2002

Advanced Methods of Biomedical Signal Processing

Independent Component Analysis and Signal Separation

Annotation This book constitutes the refereed proceedings of the 11th International Conference on Neural Information Processing, ICONIP 2004, held in Calcutta, India in November 2004. The 186 revised papers presented together with 24 invited contributions were carefully reviewed and selected from 470 submissions. The papers are organized in topical sections on computational neuroscience, complex-valued neural networks, self-organizing maps, evolutionary computation, control systems, cognitive science, adaptive intelligent systems, biometrics, brain-like computing, learning algorithms, novel neural architectures, image processing, pattern recognition, neuroinformatics, fuzzy systems, neuro-fuzzy systems, hybrid systems, feature analysis, independent component analysis, ant colony, neural network hardware, robotics, signal processing, support vector machine, time series prediction, and bioinformatics.

tionsalso,apartfromsignalprocessing,withother?eldssuchasstatisticsandarti?cial neuralnetworks. As long as we can ?nd a system that emits signals propagated through a mean, andthosesignalsarereceivedbyasetofsensorsandthereisaninterestinrecovering the originalsources,we have a potential?eld ofapplication forBSS and ICA. Inside thatwiderangeofapplicationswecan?nd,forinstance:noisereductionapplications, biomedicalapplications,audiosystems,telecommunications,andmanyothers. This volume comes out just 20 years after the ?rst contributionsin ICA and BSS 1 appeared . Thereinafter,the numberof research groupsworking in ICA and BSS has been constantly growing, so that nowadays we can estimate that far more than 100 groupsareresearchinginthese?elds. Asproofoftherecognitionamongthescienti?ccommunityofICAandBSSdev- opmentstherehavebeennumerousspecialsessionsandspecialissuesinseveralwell- 1 J.Herault, B.Ans, “ Circuits neuronaux à synapses modi?ables: d é codage de messages c- posites para apprentissage non supervise ” , C.R. de l'Acad é mie des Sciences, vol. 299, no. III-13,pp.525 – 528,1984.

The purpose of this lecture book is to present the state of the art in nonlinear blind source separation, in a form appropriate for students, researchers and developers. Source separation deals with the problem of recovering sources that are observed in a mixed condition. When we have little knowledge about the sources and about the mixture process, we speak of blind source separation. Linear blind source separation is a relatively well studied subject, however nonlinear blind source separation is still in a less advanced stage, but has seen several significant developments in the last few years. This publication reviews the main nonlinear separation methods, including the separation of post-nonlinear mixtures, and the MISEP, ensemble learning and kTDSEP methods for generic mixtures. These methods are studied with a significant depth. A historical overview is also presented, mentioning most of the relevant results, on nonlinear blind source separation, that have been presented over the years.

A fundamental problem in neural network research, as well as in many other disciplines, is finding a suitable representation of multivariate data, i.e. random vectors. For reasons of computational and conceptual simplicity, the representation is often sought as a linear transformation of the original data. In other words, each component of the representation is a linear combination of the original variables. Well-known linear transformation methods include principal component analysis, factor analysis, and projection pursuit. Independent component analysis (ICA) is a recently developed method in which the goal is to find a linear representation of nongaussian data so that the components are statistically independent, or as independent as possible. Such a representation seems to capture the essential structure of the data in many applications, including feature extraction and signal separation.

12th International Conference, LVA/ICA 2015, Liberec, Czech Republic, August 25-28, 2015, Proceedings

Theory and Applications

Unervised Adaptive Filtering, Blind Source Separation

Handbook of Blind Source Separation

Nonlinear Blind Source Separation and Blind Mixture Identification

8th Ibero-American Conference on AI, Seville, Spain, November 12-15, 2002, Proceedings

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.