Blind Speech Separation

This book constitutes the proceedings of the 9th International Conference on Latent Variable Analysis and Signal Separation, LVA/ICA 2010, held in St. Malo, France,

in September 2010. The 25 papers presented were carefully reviewed and selected from over hundred submissions. The papers collected in this volume demonstrate that the research activity in the field continues to gather theoreticians and practitioners, with contributions ranging range from abstract concepts to the most concrete and applicable questions and considerations. Speech and audio, as well as biomedical applications, continue to carry the mass of the

considered applications. Unsurprisingly the concepts of sparsity and non-negativity, as well as tensor decompositions, have become predominant, reflecting the strongactivity on these themes in signal and image

processing at large. Speech Dereverberation gathers together an overview, a mathematical formulation of the problem and the state-of-the-art solutions for dereverberation. Speech Dereverberation

presents current approaches to the problem of reverberation. It provides a review of topics in room acoustics and also describes performance measures for dereverberation. The algorithms are then explained with mathematical analysis and examples that enable the reader to see the strengths and weaknesses of the various techniques, as well as giving an understanding of the questions still to be addressed. Techniques rooted in speech enhancement

are included, in addition to a treatment of multichannel blind acoustic system identification and inversion. The TRINICON framework is shown in the context of dereverberation to be a generalization of the signal processing

for a range of analysis and enhancement techniques. Speech Dereverberation is suitable for students at masters and doctoral level, as well as established researchers.
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.

A Novel Beamforming Blind Source Separation Technique for the Separation and Localisation of Speech Signals

Blind Convolutive Speech Separation

and Dereverberation
On-line Blind Signal Separation to Speech Sources

Advances in Modern Blind Signal Separation Algorithms

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?ndingsan dsystemscouldbecollect edinthisbook.Comparing the contributions in both edited volumes progress in knowledge and technology becomesclea rlyvisible:Blindmethods andmultiinputsystemsre place 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. 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
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
Source Separation and Machine Learning
Audio Source Separation
Implementation and

Evaluation of a Real-time Blind Source Separation Algorithm for Speech
Fast Convolutive Blind Speech Separation Via Subband Adaptation
Advances in Theory, Algorithms and Applications
This handbook plays a fundamental role in sustainable progress in speech research and

development. With an accessible format and with accompanying DVD-Rom, it targets three categories of readers: graduate students, professors and active researchers in academia, and engineers in industry who need to understand or implement some specific algorithms for their speech-related products. It is a superb source of application-oriented, authoritative and comprehensive information about these technologies, this work combines the established knowledge derived from research in such fast evolving disciplines as Signal Processing and Communications, Acoustics, Computer

Science and Linguistics. 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.

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.

Latent Variable Analysis and Signal Separation
Blind Speech Separation
Speech Enhancement Technique Based on Blind Source Separation for Far-Field Noisy Speech Recognition
User Customizable Real-time Single and Dual Microphone Speech Enhancement and Blind Speech Separation for

Smartphone Hearing Aid Applications
Speech Enhancement
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.
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.

"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." -- International Conference, SIP 2009, Held as Part of the Future Generation

Information Technology Conference, FGIT 2009, Jeju Island, Korea, December 10-12, 2009. Proceedings
10th International Conference, LVA/ICA 2012, Tel Aviv, Israel, March 12-15, 2012,

Proceedings
Speech Dereverberation
Blind Signal Processing
Signal Processing, Image Processing and Pattern Recognition,
The need for speech enhancement is

very important, because of the acoustic environment we are living in, which is composed of noise and other atmospheric disturbances, and this makes it almost impossible to *record a speech signal in pure form. In most of the mixed signals there is usually no information about each source. In such situation the estimates of the original source signals* *is done based on the information of the received mixed signals, therefore the approach to be adopted in such cases to separate the signals must be one that does it blindly,* *thus the method Blind Source Separation is used in this work. Our thesis work focuses on Frequency domain Blind Source Separation (BSS) in which the received* *mixed signals are converted into the frequency domain and Independent Component Analysis (ICA) is applied at each frequency bin. Our main target in this project is to* *solve the permutation and scaling ambiguities in real time applications using the method proposed by Minje et al in [12]. Our results show that this method works*

better in an "offline" mixtures than in real time and lastly we gave some suggestions to improve the results.
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.
Blind Speech Se parationSpringe r Science & Business Media
Blind Source Separation Audio Signal Processing for

Next-Generation Multimedia Communication Systems Exploiting Second Order Statistics 7th International Conference, ICA 2007, London, UK, September 9-12, 2007,

Proceedings Independent Component Analysis and Signal Separation
As future generation information technology (FGIT) becomes specialized and fr-mented, it is easy to lose sight that many

topics in FGIT have common threads and, because of this, advances in one discipline may be transmitted to others. Presentation of recent results obtained in different disciplines encourages this interchange for the advancement of FGIT as a whole. Of particular interest are hybrid solutions that c- bine ideas taken from multiple disciplines in order to achieve something more signi- cant than the sum of the individual parts. Through such hybrid philosophy, a new principle can be discovered, which has the propensity to propagate throughout mul-faceted disciplines. FGIT 2009 was the first mega-conference that attempted to follow the above idea of hybridization in FGIT in a form of multiple events related to particular disciplines of IT, conducted by separate scientific committees, but coordinated in order to expose the most important contributions. It included the following international conferences: Advanced Software Engineering and Its Applications

(ASEA), Bio-Science and Bio-Technology (BSBT), Control and Automation (CA), Database Theory and Application (DTA), D- aster Recovery and Business Continuity (DRBC; published independently), Future G- eration Communication and Networking (FGCN) that was combined with Advanced Communication and Networking (ACN), Grid and Distributed Computing (GDC), M- timedia, Computer Graphics and Broadcasting (MulGraB), Security Technology (SecTech), Signal Processing, Image Processing and

Pattern Recognition (SIP), and- and e-Service, Science and Technology (UNESST).

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.
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. Independent Component Analysis for Audio and Biosignal Applications Speech and Audio Processing in Adverse Environments Efficient Blind Speech Signal

Separation Using Independant Component Analysis
Independent Component Analysis and Applications
Multichannel Blind Separation of Speech Signals in a Reverberant Environment
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.
This book constitutes the proceedings of the 10th International Conference on Latent Variable Analysis and Signal Separation, LVA/ICA 2012, held in Tel Aviv, Israel, in March 2012. The 20 revised full

papers presented together with 42 revised poster papers, 1 keynote lecture, and 2 overview papers for the regular, as well as for the special session were carefully reviewed and selected from numerous *submissions. Topics addressed are ranging from theoretical issues such as causality analysis and measures, through novel methods for employing the well-established concepts of sparsity and non-negativity for*

matrix and tensor factorization, down to a variety of related applications ranging from audio and biomedical signals to precipitation analysis.
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.
Filtering, Segmentation, and Depth
Blind Speech Separation in Distant Speech

Recognition Front-end Processing
Blind Source Separation of Speech Signals Using Filter Banks
Blind Source Separation Using Frequency Independent Component Analysis
Blind Convolutive

Stereo Speech Separation and Dereverberation

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 "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 conto urs."--PUBLISHER'S WEBSITE.
Extraction of a target speech signal from the

convolutive mixture of multiple sources observed in a cocktail party environment is a challenging task, especially when the room acoustic effects and background noise are present in the environment. Such acoustic distortions may further degrade the separation performance of many existing source separation algorithms. Algorithmic solutions to this problem are likely to have strong impact on many applications including automatic speech recognition, hearing aids and cochlear implants, and human-machine interaction. In such applications, to extract the target speech, it is usually required to deal with not only the interfering sound, but also the room reverberations and background noise. To address this problem, several methods are developed in this thesis. For the blind separation of a target speech signal from the convolutive mixture, a multistage algorithm is proposed in which a convolutive independent component analysis (leA) algorithm is applied to the mixture, followed by the estimation of an ideal binary mask (IBM) from the separated sources obtained with the convolutive leA algorithm. In the last step, the errors introduced due to estimation of the IBM are reduced by cepstral smoothing. The separation performance of the above algorithm, however, deteriorates with the increase in surface reflections and background noise within the room environment. Two different methods are therefore developed to reduce such effects. In the first method which is also a multistage method, acoustic effects and background' noise are treated together using an empirical-mode-decomposition (EMD) based algorithm. The noisy reverberant speech is decomposed adaptively into oscillatory components called intrinsic mode functions (IMFs) via an EMD algorithm. Denoising is then applied to selected high frequency IMFs using an EMD- based minimum mean squared error (MMSE) filter, followed by spectral subtraction of the resulting denoised high and low-frequency IMFs. The second method is a two-stage dereverberation algorithm in which the smoothed spectral subtraction mask based on a frequency dependent model is derived and then applied to the reverberant speech to reduce the effects of late reverberations. Wiener filtering is then applied such that the early reverberations are attenuated. Finally, an algorithm is developed for joint blind separation and blind dereverberation. The proposed method consists of a step for the blind estimation of reverberation time (RT). The method is employed in three different ways. Firstly, the available mixture signals are used to estimate blindly the RT, followed by the dereverberation of the mixture signals. Then, the separation algorithm is applied to these resultant mixtures. Secondly, the separation algorithm is applied first to the mixtures, followed by the blind dereverberation of the segregated speech signals. In the third scheme, the separation algorithm is split such that the convolutive leA is first applied to the

mixtures, followed by the blind dereverberation of the signals obtained from convolutive leA. Then, the T-F representation of the dereverberated signals is used to estimate the IBM followed by cepstral smoothing.
Semi-blind Identification Methods for Source Separation of

Speech Signals
8th International Conference, ICA 2009, Paraty, Brazil, March 15-18, 2009, Proceedings
Speech Input in the Car Environment
Speech Separation by Humans and Machines
Theory and Practice
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 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.
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.
Springer Handbook of Speech Processing
Audio Source Separation and Speech Enhancement
9th International Conference, LVA/ICA 2010, St. Malo, France, September 27-30, 2010, Proceedings
Time-Domain Beamforming and

Blind Source Separation
Integrating Blind Source Separation and Subspace Speech Enhancement for Ubiquitous Voice Control System
This book constitutes the refereed proceedings of the 8th International Conference on Independent

Component Analysis and Signal Separation, ICA 2009, held in Paraty, Brazil, in March 2009. The 97 revised papers presented were carefully reviewed and selected from 137 submissions. The papers are organized in topical sections on theory, algorithms and architectures,

biomedical applications, image processing, speech and audio processing, other applications, as well as a special session on evaluation. 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.

Handbook of Blind Source Separation

A Study of Frequency-domain Blind Source Separation for Speech Enhancement

Theory and Applications

Blind Source Separation of Speech Signals