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Single Channel Speech Enhancement Using Kalman Filter - Sujan Kumar Roy - 2016
The quality and intelligibility of speech conversation are generally degraded by the surrounding noises. The main objective of speech enhancement (SE) is to eliminate or reduce such disturbing noises from the degraded speech. Various SE methods have been proposed in literature. Among them, the Kalman filter (KF) is known to be an efficient SE method that uses the minimum mean square error (MMSE). However, most of the conventional KF based speech enhancement methods need access to clean speech and additive noise information for
in noisy conditions. For LPC parameters, namely, the linear prediction coefficients (LPCs) and the additive noise variance estimation, which is impractical in the sense that in practice, we can access only the noisy speech. Moreover, it is quite difficult to estimate these model parameters efficiently in the presence of adverse environmental noises. Therefore, the main focus of this thesis is to develop single channel speech enhancement algorithms using Kalman filter, where the model parameters are estimated in noisy conditions. Depending on these parameter estimation techniques, the proposed SE methods are classified into three approaches based on non-iterative, iterative, and sub-band iterative KF. In the first approach, a non-iterative Kalman filter based speech enhancement algorithm is presented, which operates on a frame-by-frame basis. In this proposed method, the state-space model parameters, namely, the LPCs and noise variance, are estimated first estimation, a combined speech smoothing and autocorrelation method is employed. A new method based on a lower-order truncated Taylor series approximation of the noisy speech along with a difference operation serving as high-pass filtering is introduced for the noise variance estimation. The non-iterative Kalman filter is then implemented with these estimated parameters effectively. In order to enhance the SE performance as well as parameter estimation accuracy in noisy conditions, an iterative Kalman filter based single channel SE method is proposed as the second approach, which also operates on a frame-by-frame basis. For each frame, the state-space model parameters of the KF are estimated through an iterative procedure. The Kalman filtering iteration is first applied to each noisy speech frame, reducing the noise component to a certain degree. At the end of this first iteration, the LPCs and other
reconstructed HF sub-band are re-estimated using the processed speech frame and the Kalman filtering is repeated for the same processed frame. This iteration continues till the KF converges or a maximum number of iterations is reached, giving further enhanced speech frame. The same procedure will repeat for the following frames until the last noisy speech frame being processed. For further improving the speech enhancement performance, a sub-band iterative Kalman filter based SE method is also proposed as the third approach. A wavelet filter-bank is first used to decompose the noisy speech into a number of sub-bands. To achieve the best trade-off among the noise reduction, speech intelligibility and computational complexity, a partial reconstruction scheme based on consecutive mean squared error (CMSE) is proposed to synthesize the low-frequency (LF) and high-frequency (HF) sub-bands such that the iterative KF is employed only to the partially speech. Finally, the enhanced HF sub-band speech is combined with the partially reconstructed LF sub-band speech to reconstruct the full-band enhanced speech. Experimental results have shown that the proposed KF based SE methods are capable of reducing adverse environmental noises for a wide range of input SNRs, and the overall performance of the proposed methods in terms of different evaluation metrics is superior to some existing state-of-the-art SE methods.

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Speech Enhancement has been extensively studied for many years and various speech enhancement methods have been developed during the past decades. One of the objectives of speech enhancement is to provide high-quality speech communication in the presence of background noise and concurrent interference signals. In the process of speech communication, the clean speech signal is inevitably corrupted by acoustic noise from the surrounding environment, transmission media, communication equipment, electrical noise, other speakers, and other sources of interference. These disturbances can significantly degrade the quality and intelligibility of the received speech signal. Therefore, it is of great interest to develop efficient speech enhancement techniques to recover the original speech from the noisy observation. In recent years, various techniques have been developed to tackle this problem, which can be classified into single channel and multi-channel enhancement approaches. Since single channel enhancement is easy to implement, it has been a significant field of research and various approaches have been developed. For example, spectral subtraction and Wiener filtering, are among the earliest single channel methods, which are based on estimation of the power spectrum of stationary noise. However, when the noise is non-stationary, or there exists music noise and ambient speech noise, the enhancement performance would degrade considerably. To overcome this disadvantage, this thesis focuses on single channel speech enhancement under adverse noise environment, especially the non-stationary noise environment. Recently, wavelet transform based methods have been widely used to reduce the undesired
Kalman filter (KF) methods offer competitive denoising results, especially in non-stationary environment. It has been used as a popular and powerful tool for speech enhancement during the past decades. In this regard, a single channel wavelet thresholding based Kalman filter (KF) algorithm is proposed for speech enhancement in this thesis. The wavelet packet (WP) transform is first applied to the noise corrupted speech on a frame-by-frame basis, which decomposes each frame into a number of subbands. A voice activity detector (VAD) is then designed to detect the voiced/unvoiced frames of the subband speech. Based on the VAD result, an adaptive thresholding scheme is applied to each subband speech followed by the WP based reconstruction to obtain the pre-enhanced speech. To achieve a further level of enhancement, an iterative Kalman filter (IKF) is used to process the pre-enhanced speech. The proposed adaptive filtering (AT-IKF) method is evaluated and compared with some existing methods under various noise conditions in terms of segmental SNR and perceptual evaluation of speech quality (PESQ) as two well-known performance indexes. Firstly, we compare the proposed adaptive thresholding (AT) scheme with three other thresholding schemes: the non-linear universal thresholding (U-T), the non-linear wavelet packet transform thresholding (WPT-T) and the non-linear SURE thresholding (SURE-T). The experimental results show that the proposed AT scheme can significantly improve the segmental SNR and PESQ for all input SNRs compared with the other existing thresholding schemes. Secondly, extensive computer simulations are conducted to evaluate the proposed AT-IKF as opposed to the AT and the IKF as standalone speech enhancement methods. It is shown that the AT-IKF method still performs the best. Lastly, the proposed ATIKF method is compared
Speech Enhancement with Adaptive Thresholding and Kalman Filtering - Mengjiao Zhao - 2018

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Wideband Speech Enhancement Approaches Using a Kalman Filter and a Perceptual Post-filter - Frédéric Delâge - 2007

Enhancement Approaches Using a Kalman Filter and a Perceptual Post-filter - Frédéric Delâge - 2007

Speech Enhancement - Philipos C. Loizou - 2013-02-25
With the proliferation of mobile devices and hearing devices, including hearing aids and cochlear implants, there is a growing and pressing need to design algorithms that can improve speech intelligibility without sacrificing quality. Responding to this need, Speech Enhancement: Theory and Practice, Second Edition introduces readers to the basic pr

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range from security, healthcare and epidemic control to urban computing, agriculture and robotics. In this book, researchers, graduate students and professionals will find innovative solutions to real-world problems in industry and society as a whole, together with inspirations for further research.

This volume gathers selected, peer-reviewed original contributions presented at the International Conference on Computational Vision and Bio-inspired Computing (ICCVBIC) conference which was held in Coimbatore, India, on November 29-30, 2018. The works included here offer a rich and diverse sampling of recent developments in the fields of Computational Vision, Fuzzy, Image Processing and Bio-inspired Computing. The topics covered include computer vision; cryptography and digital privacy; machine learning and artificial neural networks; genetic algorithms and computational intelligence; the Internet of Things; and biometric systems, to name but a few. The applications discussed
enhancement, describing the networks; genetic algorithms and computational intelligence; the Internet of Things; and biometric systems, to name but a few. The applications discussed range from security, healthcare and epidemic control to urban computing, agriculture and robotics. In this book, researchers, graduate students and professionals will find innovative solutions to real-world problems in industry and society as a whole, together with inspirations for further research.

Adaptive Beamformer for Speech Enhancement Using Second-Order Constrained Kalman Filter - - 2013

Speech Enhancement - Jacob Benesty - 2006-03-30
A strong reference on the problem of signal and speech

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.

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2020 14th International Conference on Signal Processing and Communication Systems (ICSPCS) - IEEE Staff - 2020-12-14
The Conference will be a forum for presenting research results dealing with all aspects of protocols and processes to improve performance of communication systems and the applications to utilize full
bioinformatics networking infrastructure including results related to multimedia signal processing, medical forensic applications, Bio Communications, Nano Networking, data security and integrity, as well as emerging areas of satellite communications and bioinformatics

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A Real Time Implementation of a Novel Speech Enhancement Scheme Based on a Kalman Filter - Jiefeng Chen - 2007

Artificial Intelligence and Evolutionary Computations in Engineering Systems - Subhransu Sekhar Dash - 2016-02-05
The book is a collection of high-quality peer-reviewed research papers presented in the first International Conference on Artificial Intelligence and Evolutionary Computations in Engineering Systems (ICAIECES -2015) held at Velammal Engineering College (VEC), Chennai, India during 22 – 23 April 2015. The book discusses wide variety of industrial, engineering and scientific applications of the emerging techniques. Researchers from academic
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Optimal Filtration in Signal Processing - Mohamed Najim - 2010-01-05
The purpose of this book is to provide graduate students and practitioners with traditional methods and more recent results for model-based approaches in signal processing. Firstly, discrete-time linear models such as AR, MA and ARMA models, their properties and their limitations are introduced. In addition, sinusoidal models are addressed. Secondly, estimation approaches based on least squares methods and instrumental variable techniques are presented. Finally, the book deals with optimal filters, i.e. Wiener and Kalman filtering, and adaptive filters such as the RLS, the LMS and their variants.

Modeling, Estimation and Optimal Filtration in Signal Processing - Mohamed Najim - 2010-01-05
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Kalman filtering, and adaptive
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LMS and their variants.

**Springer Handbook of Speech Processing** - Jacob
Benesty - 2007-11-28
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
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oriented, authoritative and
comprehensive information
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Computer Science and
Linguistics.
Applications (ICPECA) - work combines the established knowledge derived from research in such fast evolving disciplines as Signal Processing and Communications, Acoustics, Computer Science and Linguistics.

2021 IEEE International Conference on Power Electronics, Computer Applications (ICPECA) - IEEE Staff - 2021-01-22
2021 IEEE International Conference on Power, Electronics and Computer Applications (ICPECA 2021) will take place in Shenyang, China, on January 22-24, 2021 ICPECA 2021 seeks to provide a high level forum for experts, researchers, professionals, innovators and practitioners in the field of Power, Electronics and Computer Applications from industry and academia to present and discuss the wide spectrum of original and novel researches and contributions together

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Modern Approaches in Machine Learning and Cognitive Science: A Walkthrough - Vinit Kumar Gunjan - 2020-02-04
This book discusses various machine learning & cognitive science approaches, presenting high-throughput research by experts in this area. Bringing together machine learning, cognitive science and other aspects of artificial intelligence to help provide a roadmap for future research on intelligent
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valuable reference resource
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Speech and Computer
- Alexey Karpov - 2020-10-04
This book constitutes the
proceedings of the 22nd
International Conference on
Speech and Computer,
presented were carefully reviewed and selected from 160 submissions. The papers present current research in the area of computer speech processing including speech science, speech technology, natural language processing, human-computer interaction, language identification, multimedia processing, human-machine interaction, deep learning for audio processing, computational paralinguistics, affective computing, speech and language resources, speech translation systems, text mining and sentiment analysis, voice assistants, etc. Due to the Corona pandemic SPECOM 2020 was held as a virtual event.

Speech and Computer - Alexey Karpov - 2020-10-04

This book constitutes the proceedings of the 22nd International Conference on Speech and Computer, SPECOM 2020, held in St. Petersburg, Russia, in October 2020. The 65 papers reviewed and selected from 160 submissions. The papers present current research in the area of computer speech processing including speech science, speech technology, natural language processing, human-computer interaction, language identification, multimedia processing, human-machine interaction, deep learning for audio processing, computational paralinguistics, affective computing, speech and language resources, speech translation systems, text mining and sentiment analysis, voice assistants, etc. Due to the Corona pandemic SPECOM 2020 was held as a virtual event.

Speech Enhancement Using Kalman Filter - Alaa Kamal Satti Salih - 2009

Linear Prediction of Speech - J.D. Markel - 2013-03-12

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During the past ten years a new area in speech processing, generally referred to as linear prediction, has evolved. As with all scientific research, results did not always get published in a logical order and terminology was not always consistent. In mid-1974, we decided to begin an extra hours and weekends project of organizing the literature in linear prediction of speech and developing it into a unified presentation in terms of content and terminology. This effort was completed in November, 1975, with the contents presented herein. If there are two words which describe our goals in this book, they are unification and depth. Considerable effort has been spent on showing the interrelationships among various linear prediction formulations and solutions, and in developing extensions such as acoustic tube models and synthesis filter structures in a unified manner with consistent terminology. Topics are presented in such a manner that derivations and theoretical details are covered, along with Fortran subroutines and practical considerations. Using this approach we hope to have made the material useful for a wide range of backgrounds and interests.

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**Speech Dereverberation** - Patrick A. Naylor - 2010-07-27
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 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.

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**Noise Reduction in Speech Processing** - Jacob Benesty - 2009-04-28
Noise is everywhere and in most applications that are related to audio and speech, interfaces, hands-free communications, voice over IP (VoIP), hearing aids, teleconferencing/telepresence/telecollaboration systems, and so many others, the signal of interest (usually speech) that is picked up by a microphone is generally contaminated by noise. As a result, the microphone signal has to be cleaned up with digital signal processing tools before it is stored, analyzed, transmitted, or played out. This cleaning process is often called noise reduction and this topic has attracted a considerable amount of research and engineering attention for several decades. One of the objectives of this book is to present in a common framework an overview of the state of the art of noise reduction algorithms in the single-channel (one microphone) case. The focus is on the most useful approaches, i.e., filtering techniques (in different domains) and spectral enhancement methods. The other objective of Noise Reduction in Speech
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2020 IEEE Asia Pacific Conference on Computer Science and Data Engineering (CSDE) - IEEE Staff - 2020-12-16
The conference title is belonging 100 in the area of IEEE Computer Society This event would be a wonderful gathering between IEEE members in the area of South Pacific, Australia and the rest of the world to share the latest development in the area of computer science and data engineering The conference will be a forum for participants to discuss state of the art innovations in technologies which have been made available by the researchers & IT professionals and will feature plenary and panel sessions as well as technical paper presentations and poster sessions Workshop by international experts on ICT applications will also be available The conference theme for 2020 is Visualise the Future through Data CSDE 2020 is also aimed to promote discussion about the pedagogical potential of new sustainable technologies for the developing countries

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**Microelectronics, Electromagnetics and Telecommunications** - P. Satish Rama Chowdary - 2020-06-24
This book discusses the latest developments and outlines future trends in the fields of microelectronics, electromagnetics and telecommunication. It includes original research presented at the International Conference on Microelectronics, Electromagnetics and Telecommunication (ICMEET 2019), organized by the Department of ECE, Raghu Institute of Technology, Andhra Pradesh, India. Written by scientists, research scholars and practitioners from leading universities, engineering colleges and R&D institutes around the globe, the papers share the latest breakthroughs in and promising solutions to the most important issues facing today’s society.

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2020 Asia Pacific Signal and Information Processing Association Annual Summit and Conference (APSIPA ASC) - IEEE Staff - 2020-12-07

APSIPA ASC 2020 (www.apsipa2020.org) is the 12th annual conference organized by Asia Pacific Signal and Information Processing Association (APSIPA), which will be held on December 7-10, 2020, in Auckland, New Zealand. Founded in 2009, APSIPA aims to promote research and education in signal processing, information technology, and communications. The annual conferences have been held previously in Sapporo, Japan (2009), Singapore (2010), Xi’an, China (2011), Los Angeles, USA (2012), Kaohsiung, Taiwan (2013), Siem Reap, Cambodia (2014), Hong Kong, China (2015), Jeju, Korea (2016), and Kuala Lumpur, Malaysia (2017) and Hawaii, China (2019). APSIPA is interested in all aspects of signal and information processing theories, algorithms, securities, implementations, and applications. All accepted papers will be indexed by EI compendex and archived by IEEE Xplore.

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International Conference on Signal Processing, Communication, Power and Embedded System - - 2016
To bring together innovative academics and industrial experts in the field of communication, signal processing, power, intelligent embedded system and data analytic SCOPES 2016 will provide an Excellent international forum for sharing knowledge and results in Communication, Signal Processing, Power, Intelligent, Embedded System and Data Analytic The aim of the Conference is to provide a platform to the researchers and practitioners from both academia as well as industry to meet the share cutting edge development in the field.

Advances in Digital Speech Transmission - Prof Rainer Martin - 2008-02-28
Speech processing and
technology and applications technology are expanding fields of active research. New challenges arise from the 'anywhere, anytime' paradigm of mobile communications, the ubiquitous use of voice communication systems in noisy environments and the convergence of communication networks toward Internet based transmission protocols, such as Voice over IP. As a consequence, new speech coding, new enhancement and error concealment, and new quality assessment methods are emerging. Advances in Digital Speech Transmission provides an up-to-date overview of the field, including topics such as speech coding in heterogeneous communication networks, wideband coding, and the quality assessment of wideband speech. Provides an insight into the latest developments in speech processing and speech transmission, making it an essential reference to those working in these fields Offers a balanced overview of

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**2020 IEEE International Symposium on Circuits and Systems (ISCAS)** - IEEE Staff - 2020-10-12
The International Symposium on Circuits and Systems (ISCAS) is the flagship conference of the IEEE Circuits and Systems (CAS) Society and the world’s premier networking and exchange forum for researchers in the highly active fields of theory, design and implementation of circuits and systems. ISCAS2020 focuses on the deployment of CASS knowledge towards Society Grand Challenges and highlights the strong foundation in methodology and the integration of multidisciplinary approaches which are the distinctive features of CAS contributions. The worldwide CAS community is exploiting such CASS knowledge to change the way in which devices and circuits are understood, optimized, and leveraged in a variety of systems and applications.

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Advanced Digital Signal Processing and Noise Reduction - Saeed V. Vaseghi - 2000-09-20
A young man begins a journey from Saudi Arabia, believing it will end with his death in England. If his mission
London's safety, his role is no martyr - and many innocents will die with him. For David Banks, an armed protection officer, charged with neutralizing the threat to London's safety, his role is no longer clear-cut: one man's terrorist is another man's freedom fighter: dangerous distinctions to a police officer with his finger on the trigger. Soon the two men's paths will cross. Before then, their commitment will be shaken by the journeys that take them there. The suicide bomber and the policeman will have cause to question the roads they've chosen. Win or lose, neither will be the same again.

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**Sequential-joint Estimation of Signal and Parameters Using the Unscented Kalman Filter with Application to Single-and Multi-microphone Speech Enhancement** - Sharon Gannot - 2002

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Microphone Arrays - Michael Brandstein - 2013-04-17
This is the first book to provide a single complete reference on microphone arrays. Top researchers in this field contributed articles documenting the current state of the art in microphone array research, development and technological application.

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Introduction and Implementations of the Kalman Filter - Felix Govaers - 2019-05-22
Sensor data fusion is the process of combining error-prone, heterogeneous, incomplete, and ambiguous data to gather a higher level of situational awareness. In principle, all living creatures are fusing information from their complementary senses to coordinate their actions and to detect and localize danger. In sensor data fusion, this process is transferred to electronic systems, which rely on some "awareness" of what is happening in certain areas of interest. By means of probability theory and statistics, it is possible to model the relationship between the state space and the sensor data. The number of ingredients of the resulting Kalman filter is limited, but its applications are not.
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This work addresses this problem in the short-time Fourier transform (STFT) domain. We divide the general problem into five basic categories depending on the number of microphones being used and whether the interframe or interband correlation is considered. The first category deals with the single-channel problem where STFT coefficients at different frames and frequency bands are assumed to be independent. In this case, the noise reduction filter in each frequency band is basically a real gain. Since a gain does not improve the signal-to-noise ratio (SNR) for any given subband and frame, the noise reduction is basically achieved by liftering the subbands and frames that are less noisy while weighing down on those that are more noisy. The second category also concerns the single-channel problem. The difference is that now the interframe correlation is taken into account and a filter is applied in each subband instead of just a gain. The
investigates the use of interframe correlation is that we can improve not only the long-time fullband SNR, but the frame-wise subband SNR as well. The third and fourth classes discuss the problem of multichannel noise reduction in the STFT domain with and without interframe correlation, respectively. In the last category, we consider the interband correlation in the design of the noise reduction filters. We illustrate the basic principle for the single-channel case as an example, while this concept can be generalized to other scenarios. In all categories, we propose different optimization cost functions from which we derive the optimal filters and we also define the performance measures that help analyzing them.

Fractional Fourier Transform Techniques for Speech Enhancement - Prajna Kunche - 2020
This book explains speech enhancement in the Fractional Fourier Transform (FRFT) domain and different FRFT algorithms in both single channel and multi-channel enhancement systems, which has proven to be an ideal time frequency analysis tool in many speech signal processing applications. The authors discuss the complexities involved in the highly non-stationary signal processing and the concepts of FRFT for speech enhancement applications. The book explains the fundamentals of FRFT as well as its implementation in speech enhancement. Theories of different FRFT methods are also discussed. The book lets readers understand the new fractional domains to prepare them to develop new algorithms. A comprehensive literature survey regarding the topic is also made available to the reader. Analyzes FRFT techniques in speech enhancement applications; Presents new approaches for speech enhancement using FRFT; Suggests the future directions of research in this emerging area.
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Bayesian Filtering and Smoothing - Simo Särkkä - 2013-09-05
A unified Bayesian treatment of the state-of-the-art filtering, smoothing, and parameter estimation algorithms for non-linear state space models.

Speech Enhancement - Jacob Benesty - 2014-01-04
Speech enhancement is a classical problem in signal processing, yet still largely unsolved. Two of the conventional approaches for solving this problem are...
linear filtering, like the classical Wiener filter, and subspace methods. These approaches have traditionally been treated as different classes of methods and have been introduced in somewhat different contexts. Linear filtering methods originate in stochastic processes, while subspace methods have largely been based on developments in numerical linear algebra and matrix approximation theory. This book bridges the gap between these two classes of methods by showing how the ideas behind subspace methods can be incorporated into traditional linear filtering. In the context of subspace methods, the enhancement problem can then be seen as a classical linear filter design problem. This means that various solutions can more easily be compared and their performance bounded and assessed in terms of noise reduction and speech distortion. The book shows how various filter designs can be obtained in this framework, including the maximum SNR, Wiener, LCMV, and MVDR filters, and how these can be applied in various contexts, like in single-channel and multichannel speech enhancement, and in both the time and frequency domains. First short book treating subspace approaches in a unified way for time and frequency domains, single-channel, multichannel, as well as binaural, speech enhancement Bridges the gap between optimal filtering methods and subspace approaches Includes original presentation of subspace methods from different perspectives

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Progress in Nonlinear
Speech Processing - Yannis
Stylianou - 2007-05-24
This book constitutes of the
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(European Cooperation in the
field of Scientific and
Technical Research) Action
277: NSP, Nonlinear Speech
Processing, running from
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Coverage includes such areas
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**Robust Automatic Speech Recognition** - Jinyu Li - 2015-10-30

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

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**Data Engineering and Intelligent Computing** - Suresh Chandra Satapathy - 2017-05-31
The book is a compilation of high-quality scientific papers presented at the 3rd International Conference on Computer & Communication Technologies (IC3T 2016). The individual papers address cutting-edge technologies and applications of soft computing, artificial intelligence and communication. In addition, a variety of further topics are discussed, which include data mining, machine intelligence, fuzzy computing, sensor networks, signal and image processing, human-computer interaction, web intelligence, etc. As such, it offers readers a valuable and unique resource.

This two-volume book presents an unusually diverse selection of research papers, covering all major topics in the fields of information and communication technologies and related sciences. It provides a wide-angle snapshot of current themes in information and power engineering, pursuing a cross-disciplinary approach to do so. The book gathers revised contributions that were presented at the 2018 International Conference: Sciences of Electronics, Technologies of Information and Telecommunication (SETIT’18), held on 20–22 December 2018 in Hammamet, Tunisia. This eighth installment of the event attracted a wealth of submissions, and the papers presented here were selected underwent additional, painstaking revision. Topics covered include: · Information Processing · Human-Machine Interaction · Computer Science · Telecommunications and Networks · Signal Processing · Electronics · Image and Video This broad-scoped approach is becoming increasingly popular in scientific publishing. Its aim is to encourage scholars and professionals to overcome disciplinary barriers, as demanded by current trends in the industry and in the consumer market, which are rapidly leading toward a convergence of data-driven applications, computation, telecommunication, and energy awareness. Given its coverage, the book will benefit graduate students, researchers and practitioners who need to keep up with the latest technological advances.
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