Linear And Nonlinear Loudspeaker Characterization

Some issues, Aug. 1948-1954 are called Radio-electronic engineering edition and include a separately numbered and paged section: Radio-electronic engineering (issued separately Aug. 1954-May 1955).
Long-awaited update and expansion of a widely recognised classic in the field by pioneering acoustics expert, Leo L. Beranek Builds upon Beranek's 1954 Acoustics classic by incorporating recent developments, practical formulas and methods for effective simulation Uniquely, provides the detailed acoustic fundamentals which enable better understanding of complex design parameters, measurement methods and data Brings together topics currently scattered across a variety of books and sources into one valuable reference Includes relevant case studies, real-world examples and solutions to bring the theory to life Acoustics: Sound Fields and Transducers is a modern expansion and re-working of Acoustics, the 1954 classic reference written by Leo L. Beranek. Updated throughout and focused on electroacoustics with the needs of a broad range of acoustics engineers and scientists in mind, this new book retains and expands on the detailed acoustical fundamentals included in the original whilst adding practical formulas and simulation methods for practising professionals. Benefitting from Beranek's lifetime experience as a leader in the field and co-author Tim Mellow's cutting-edge industry experience, Acoustics: Sound Fields and Transducers is a modern classic to keep close to hand in the lab, office and design studio. Builds on Beranek's 1954 Acoustics classic by incorporating recent developments, practical formulas and methods for effective simulation Uniquely provides the detailed acoustic fundamentals, enabling better understanding of complex design parameters, measurement methods and data Brings together topics currently scattered across a variety of books and sources into one valuable reference Includes relevant case studies, real-world examples and solutions to bring the theory to life
This fourth volume, edited and authored by world leading experts, gives a review of the principles, methods and techniques of important and emerging research topics and technologies in Image, Video Processing and Analysis, Hardware, Audio, Acoustic and Speech Processing. With this reference source you will: Quickly grasp a new area of research Understand the underlying principles of a topic and its application Ascertain how a topic relates to other areas and learn of the research issues yet to be resolved Quick tutorial reviews of important and emerging topics of research in Image, Video Processing and Analysis, Hardware, Audio, Acoustic and Speech Processing Presents core principles and shows their application Reference content on core principles, technologies, algorithms and applications Comprehensive references to journal articles and other literature on which to build further, more specific and detailed knowledge Edited by leading people in the field who, through their reputation, have been able to commission experts to write on a particular topic
The parametric array exploits two highly collimated ultrasound beams interacting in a given volume producing a single beam with very high directivity and almost no side lobes. The high directivity of the difference frequency signal of the parametric array is due to the interaction of the waves in the volume effectively producing a virtual endfired array boosting pressure levels along the interaction region which is limited by the absorption coefficient. This thesis focuses on experiments conducted in an anechoic room using AS-18-B Audio Spotlight system from HolosonicTM. Furthermore, nonlinear theory was modeled by a linear discrete array. The beam pattern of the parametric loudspeaker, range dependence of primary and secondary signals and total harmonic distortion (THD) were measured and then compared to theory. Experimental data for the beam pattern of the parametric loudspeaker agreed with the theory. It was all shown that the parametric array had a very narrow beam width and almost no side lobes as opposed to conventional loudspeakers. Both primary waves and difference wave frequency signals were examined for their range dependence. Due to the complicated interference of the primary waves, it was impossible to compare experimental results with theoretical predictions. For the difference wave signals, experimental data was verified by theory, which was modified in order to accommodate both wave generation and spreading region. Finally, THD of the parametric loudspeaker was measured for different amplitude modulation depths. Experimental results showed that preprocessing should be applied in order to decrease THD and achieve clean audio signal reproduction.
Artificial neural networks possess several properties that make them particularly attractive for applications to modelling and control of complex non-linear systems. Among these properties are their universal approximation ability, their parallel network structure and the availability of on- and off-line learning methods for the interconnection weights. However, dynamic models that contain neural network architectures might be highly non-linear and difficult to analyse as a result. Artificial Neural Networks for Modelling and Control of Non-Linear Systems investigates the subject from a system theoretical point of view. However the mathematical theory that is required from the reader is limited to matrix calculus, basic analysis, differential equations and basic linear system theory. No preliminary knowledge of neural networks is explicitly required. The book presents both classical and novel network architectures and learning algorithms for modelling and control. Topics include non-linear system identification, neural optimal control, top-down model based neural control design and stability analysis of neural control systems. A major contribution of this book is to introduce NLq Theory as an extension towards modern control theory, in order to analyze and synthesize non-linear systems that contain linear together with static non-linear operators that satisfy a sector condition: neural state space control systems are an example. Moreover, it turns out that NLq Theory is unifying with respect to many problems arising in neural networks, systems and control. Examples show that complex non-linear systems can be modelled and controlled within NLq theory, including mastering chaos. The didactic flavor of this book makes it suitable for use as a text for a course on Neural Networks. In addition, researchers and designers will find many important new techniques, in particular NLq emTheory, that have applications in control theory, system theory, circuit theory and Time Series Analysis.
The prospect of writing a book on loudspeakers is a daunting one, since only a multivolume encyclopedia could truly do justice to the subject. Authors writing about this subject have generally concentrated on their own areas of expertise, often covering their own specific topics in great detail. This book is no exception; the author's background is largely in professional loudspeaker application and specification, and the emphasis in this book is on basic component design, operation, measurement, and system concepts. The book falls largely into two sections; the first (Chapters 1-9) emphasizing the building blocks of the art and the second (Chapters 10-16) emphasizing applications, measurements, and modeling. While a thorough understanding of the book requires a basic knowledge of complex algebra, much of it is understandable through referring to the graphics. Every attempt has been made to keep graphics clear and intuitive. Chapter 1 deals with the basic electro-mechano-acoustical chain between input to the loudspeaker and its useful output, with emphasis on the governing equations and equivalent circuits. Chapter 2 is a survey of cone and dome drivers, the stock-in-trade of the industry. They are discussed in terms of type, design, performance, and perfor

mance limits. Chapter 3 deals with magnetics. Once a source of difficulty in loudspeaker design, magnetics today yields easily to modeling techniques. Chapter 4 discusses low-frequency (LF) system performance, primarily from the viewpoint of Thiele-Small parameters. We also discuss some of the multi chamber LF systems that became popular during the eighties.

Need advice on which type of speaker to use and where? Very often the choice and positioning of loudspeakers is down to intuition, hearsay and chance. This practical guide explores the link between experience and the technology, giving you a better understanding of the tools you are using and why, leading to greatly improved results. Newell and Holland share years of experience in the design, application and use of loudspeakers for recording and reproducing music. Get practical advice on the applications of different loudspeakers to the different phases of the music recording and reproduction chain. If you are using loudspeakers in a recording studio, mastering facility, broadcasting studio, film post production facility, home or musician's studio, or you inspire to improve your music reproduction system this book will help you make the right decisions.

All the design and development inspiration and direction an audio engineer needs in one blockbuster book! Douglas Self has selected the very best sound engineering design material from the Focal and Newnes portfolio and complied it into this volume. The result is a book covering the gamut of sound engineering. The material has been selected for its timelessness as well as for its relevance to contemporary sound engineering issues.

Refereed postproceedings of the International Conference on Non-Linear Speech Processing, NOLISP 2005. The 30 revised full papers presented together with one keynote speech and 2 invited talks were carefully reviewed and selected from numerous submissions for inclusion in the book. The papers are organized in topical sections on speaker recognition, speech analysis, voice pathologies, speech recognition, speech enhancement, and applications.

This textbook provides a unified approach to acoustics and vibration suitable for use in advanced undergraduate and first-year graduate courses on vibration and fluids. The book includes thorough treatment of vibration of harmonic oscillators, coupled oscillators, isotropic elasticity, and waves in solids including the use of resonance techniques for determination of elastic moduli. Drawing on 35 years of experience teaching introductory graduate acoustics at the Naval Postgraduate School and Penn State, the author presents a hydrodynamic approach to the acoustics of sound in fluids that provides a uniform methodology for analysis of lumped-element systems and wave propagation that can incorporate attenuation mechanisms and complex media. This view provides a consistent and reliable approach that can be extended with confidence to more complex fluids and future applications. Understanding Acoustics opens with a mathematical introduction that includes graphing and statistical uncertainty, followed by five chapters on vibration and elastic waves that provide important results and highlight modern applications while introducing analytical techniques that are revisited in the study of waves in fluids covered in Part II. A unified approach to waves in fluids (i.e., liquids and gases) is based on a mastery of the hydrodynamic equations. Part III demonstrates extensions of this view to nonlinear acoustics. Engaging and practical, this book is a must-read for graduate students in acoustics and vibration as well as active researchers interested in a novel approach to the material.

Higher-Order Statistical Signal Processing brings together some most recent innovations in the field of higher-order statistical signal processing. It is structured to provide a comprehensive understanding of the fundamentals of the discipline, as well as a treatment of recent advances.

This is the definitive reference for microphones and loudspeakers, your one-stop reference covering in great detail all you could want and need to know about electroacoustics devises (microphones and loudspeakers). Covering both the technology and the practical set up and placement this guide explores and bridges the link between experience and the technology, giving you a better understanding of the tools to use and why, leading to greatly improved results.

Measured transfer functions of acoustic systems are often used to derive single-number parameters. The uncertainty analysis is commonly focused on the derived parameters but not on the transfer function as the primary quantity. This thesis presents an approach to assess the uncertainty contributions in these transfer functions by using analytic models. Uncertainties caused by the measurement method are analyzed with a focus on the underlying signal processing. In particular, the influence of nonlinearities in the acoustic measurement chain are modeled to predict artifacts in the measured signals and hence the calculated acoustic transfer function. Secondly, characterization methods commonly applied in the field of signal processing are linked to the acoustic scenarios and the main influencing parameters. Acoustic parameters are then derived analytically and by means of Monte Carlo simulations considering the uncertainty of these input parameters. In order to provide airborne applications, analytic models for sound barrier and room acoustic measurements are developed incorporating the directivity and the orientation of the sound source as well as the positions of sources and receivers. The simulated uncertainty contributions are validated by measurements. The same approach is also applied to structure-borne sound applications.

A digital filter can be pictured as a "black box" that accepts a sequence of numbers and emits a new sequence of numbers. In digital audio signal processing applications, such number sequences usually represent sounds. For example, digital filters are used to implement graphic equalizers and other digital audio effects. This book is a gentle introduction to digital filters, including mathematical theory, illustrative examples, some audio applications, and useful software starting points. The theory treatment begins at the high-school level, and covers fundamental concepts in linear systems theory and digital filter analysis. Various "small" digital filters are analyzed as examples, particularly those commonly used in audio applications. Matlab programming examples are emphasized for illustrating the use and development of digital filters in practice.

Modelling and simulation in acoustics is currently gaining importance. In fact, with the development and improvement of innovative computational techniques and with the growing need for predictive models, an impressive boost has been observed in several research and application areas, such as noise control, indoor acoustics, and industrial applications.

This led us to the proposal of a special issue about “Modelling, Simulation and Data Analysis in Acoustical Problems”, as we believe in the importance of these topics in modern acoustics’ studies. In total, 81 papers were submitted and 33 of them were published, with an acceptance rate of 37.5%. According to the number of papers submitted, it can be affirmed that this is a trending topic in the scientific and academic community and this special issue will try to provide a future reference for the research that will be developed in coming years.
This volume contains the proceedings of NOLISP 2009, an ISCA Tutorial and Workshop on Non-Linear Speech Processing held at the University of Vic (- talonia, Spain) during June 25-27, 2009. NOLISP2009wasprecededbythreeeditionsofthisbiannualeventheld2003 in Le Croisic (France), 2005 in Barcelona, and 2007 in Paris. The main idea of NOLISP workshops is to present and discuss new ideas, techniques and results related to alternative approaches in speech processing that may depart from the mainstream. In order to work at the front-end of the subject area, the following domains of interest have been de?ned for NOLISP 2009: 1. Non-linear approximation and estimation 2. Non-linear oscillators and predictors 3. Higher-order statistics 4. Independent component analysis 5. Nearest neighbors 6. Neural networks 7. Decision trees 8. Non-parametric models 9. Dynamics for non-linear systems 10. Fractal methods 11. Chaos modeling 12. Non-linear di?erential equations The initiative to organize NOLISP 2009 at the University of Vic (UVic) came from the UVic Research Group on Signal Processing and was supported by the Hardware-Software Research Group. We would like to acknowledge the ?nancial support obtained from the M- istry of Science and Innovation of Spain (MICINN), University of Vic, ISCA, and EURASIP. All contributions to this volume are original. They were subject to a doub- blind refereeing procedure before their acceptance for the workshop and were revised after being presented at NOLISP 2009.
Many digital control circuits in current literature are described using analog transmittance. This may not always be acceptable, especially if the sampling frequency and power transistor switching frequencies are close to the band of interest. Therefore, a digital circuit is considered as a digital controller rather than an analog circuit. This helps to avoid errors and instability in high frequency components. Digital Signal Processing in Power Electronics Control Circuits covers problems concerning the design and realization of digital control algorithms for power electronics circuits using digital signal processing (DSP) methods. This book bridges the gap between power electronics and DSP. The following realizations of digital control circuits are considered: digital signal processors, microprocessors, microcontrollers, programmable digital circuits. Discussed in this book is signal processing, starting from analog signal acquisition, through its conversion to digital form, methods of its filtration and separation, and ending with pulse control of output power transistors. The book is focused on two applications for the considered methods of digital signal processing: an active power filter and a digital class D power amplifier. The major benefit to readers is the acquisition of specific knowledge concerning discussions on the processing of signals from voltage or current sensors using a digital signal processor and to the signals controlling the output inverter transistors. Included are some Matlab examples for illustration of the considered problems.
Until now the criteria used in the design of a mosque sound reinforcement system are mainly based on criteria for religious building well accepted in the West. Accurateness and effectiveness of the theory and criteria being using cannot be upheld as the end users often could not accept the end product. it is appreciated that mosque and churches have fundamentally different acoustic requirements. This research was conducted primarily to identify design criteria for sound system that will be accepted by the local mosque congregation. The criteria investigated were the ambient noise disturbance level due to fan and pink noise, the most acceptable speech loudless level due to fanwith an optimum intelligibility at various ambient noise levels, the Haas (localization) effect and the percentage disturbance due to delay time and the difference in primary over secondary loudness level. In addition, the acoustic characteristics of the mosque of the UTM Kuala Lumpur and the characteristics of the sound system installed to enable this research to be conducted were elaborated on in this thesis. Analysis of the data gathered was done using the Statistical Analysis System (SAS) package available at UTM Computer Centre. The statistical analysis discussed includes varieties correlation coefficients, variates mean value, standard error, 95% confidence interval, Duncan multiple range test, T-test, coefficients of variations (CV), linear and nonlinear mathematical modelling of the variates under study. Based on the mathematical model obtained, prediction was made on the ambient boise level that would procedure peaceful and serenity environment inside the mosque. The most accepted speech loduness level with an optimum speech inteligibility for various ambient noise level with optimum speech intelligibility for various ambient noise level was also predicted. The results indicated that, for optimum intelligibility, speech level of at least 3 dB(A) above the most accepted speech loudness level is required. For the Haas effect, the loudspeaker arrangement plays a significant role, as it was fou theat the decentralised loudspeaker arrangement was able to provide realism more effectively. The percentage disturbance found in this study significantly indicated that higher level of disturbance as compared to the Haas findings. It was indicated also that, the exixtence of any echo cannot be tolerated. it is essential to ensure that the speech is heard to come mainly from the primary source. The implicit functions of the sound system design acceptance criteria also being contributed by the room acoustic characteristics. As such, further research is required to ascertain the actual implicit function for the sound system design acceptance criteria.
"Directory of members" published as pt. 2 of Apr. 1954- issue
This intriguing book constitutes the thoroughly refereed postproceedings of the International Conference on Non-Linear Speech Processing, NOLISP 2007, held in Paris, France, in May 2007. The 24 revised full papers presented were carefully reviewed and selected from numerous submissions. The papers are organized in topical sections on nonlinear and non-conventional techniques, speech synthesis, speaker recognition, speech recognition, and many other subjects.
Unique in its approach, Talker Variability in Speech Processing embraces the differences in speech patterns without

treating them as unwanted variables. The editors take on the difficult task of converting the mapping of speech patterns into mental representations. They cover theories of perception and cognition, issues in clinical speech pathology, and the practical concerns of speech technology. A radical departure from traditional approaches to speech processing, this text will strike a major chord for those surrounded by the dissonance of speech perception and language processing issues.
Proceedings of the NATO Advanced Study Institute on Computational Models of Speech Pattern Processing, held in St. Helier, Jersey, UK, July 7-18, 1997
Sound Reproduction: The Acoustics and Psychoacoustics of Loudspeakers and Rooms, Third Edition explains the physical and perceptual processes that are involved in sound reproduction and demonstrates how to use the processes to create high-quality listening experiences in stereo and multichannel formats. Understanding the principles of sound production is necessary to achieve the goals of sound reproduction in spaces ranging from recording control rooms and home listening rooms to large cinemas. This revision brings new science-based perspectives on the performance of loudspeakers, room acoustics, measurements and equalization, all of which need to be appropriately used to ensure the accurate delivery of music and movie sound tracks from creators to listeners. The robust website (www.routledge.com/cw/toole) is the perfect companion to this necessary resource.
This book constitutes of the major results of the EU COST (European Cooperation in the field of Scientific and Technical Research) Action 277: NSP - Nonlinear Speech Processing - running from April 2001 to June 2005. The results were presented at the last meeting of the management committee of COST Action 277, held in Heraklion, Crete, Greece on September 20-23, 2005 during the Workshop on Nonlinear Speech Processing, WNSP 2005. The 13 revised full papers in this state-of-the-art survey were carefully reviewed and selected for inclusion in the book and are preceded with an introductory leading-in by the editors. The articles present overviews of the four years research combining linear and non linear approaches for processing the speech signal. The aim of this book is to provide an additional and/or an alternative way to the traditional approach of linear speech processing and be mainly used by the researcher working in the domain. The papers cover areas such as speech analysis for speech synthesis, speech recognition, speech-non speech discrimination and voice quality assessment, speaker recognition/verification from a natural or modified speech signal, speech recognition, speech enhancement, and emotional state detection.
Advances in Non-Linear Modeling for Speech Processing includes advanced topics in non-linear estimation and modeling techniques along with their applications to speaker recognition. Non-linear aeroacoustic modeling approach is used to estimate the important fine-structure speech events, which are not revealed by the short time Fourier transform (STFT). This aeroacostic modeling approach provides the impetus for the high resolution Teager energy operator (TEO). This operator is characterized by a time resolution that can track rapid signal energy changes within a glottal cycle. The cepstral features like linear prediction cepstral coefficients (LPCC) and mel frequency cepstral coefficients (MFCC) are computed from the magnitude spectrum of the speech frame and the phase spectra is neglected. To overcome the problem of neglecting the phase spectra, the speech production system can be represented as an amplitude modulation-frequency modulation (AM-FM) model. To demodulate the speech signal, to estimation the amplitude envelope and instantaneous frequency components, the energy separation algorithm (ESA) and the Hilbert transform demodulation (HTD) algorithm are discussed. Different features derived using above non-linear modeling techniques are used to develop a speaker identification system. Finally, it is shown that, the fusion of speech production and speech perception mechanisms can lead to a robust feature set.
"Signals and Systems for Speech and Hearing, 2nd Edition" provides the reader with a thorough introduction to the concepts of signals and systems analysis that play a role in the speech and hearing sciences. Few equations are used, and an informal, friendly and informative style is maintained throughout. Because much of the story is told through figures, the authors have gone to great lengths to provide clear and truthful figures that show what the text says they do. It is hoped the reader will come away with a strong visual understanding of the concepts involved. This book can be used at many levels, from the student who hasn't heard of a spectrum before, to the experienced worker who has only a fuzzy understanding of the notion of an impulse response. The authors have tried to keep the underlying conceptual structure of signals and systems analysis explicit, in the hope that even some readers with advanced technical training might find clarification of the basic principles. Notable features include over 300 figures integrated closely with the text, all drawn specifically. Exercises are provided at the end of most chapters.
The near field seismic propagation medium was characterized using Wiener's nonlinear identification techniques. The system stimulus was a white noise signal generated by an audio system and measured by a microphone placed directly in front of the speaker. The output signal was measured by a geophone placed at predetermined intervals down range from the speaker. The first and second order Wiener kernels of the system were determined, and it was conclusively shown that the system exhibits nonlinearities. The calculated kernels were then used to predict an output for a given input. Comparison of the predicted output with that of the measured output indicates that the physical system can be more accurately characterized by the first and second order Wiener kernels than by use of linear models. (Author).
This book treats important topics in "Acoustic Echo and Noise Control" and reports the latest developments. Methods for enhancing the quality of transmitted speech signals are gaining growing attention in universities and in industrial development laboratories. This book, written by an international team of highly qualified experts, concentrates on the modern and advanced methods.
This book constitutes of the major results of the EU COST (European Cooperation in the field of Scientific and Technical Research) Action 277: NSP, Nonlinear Speech Processing, running from April 2001 to June 2005. Coverage includes such areas as speech analysis for speech synthesis, speech recognition, speech-non speech discrimination and voice quality assessment, speech enhancement, and emotional state detection.
LoudspeakersFor music recording and reproductionCRC Press