Linear And Nonlinear Loudspeaker Characterization

Understanding Acoustics
1977 IEEE International Conference on Acoustics, Speech, & Signal Processing, Held at the Sheraton-Hartford Hotel, Hartford, Connecticut, May 9-11, 1977
Time Delay Spectrometry
Signals and Systems for Speech and Hearing
5th International Conference on Nonlinear Speech Processing, NoLISP 2011, Las Palmas de Gran Canaria, Spain, November 7-9, 2011, Proceedings
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Journal of the Audio Engineering Society
Identification of Nonlinear Systems in Acoustics
Nonlinear Analysis, Differential Equations and Control
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Nonlinear Analyses and Algorithms for Speech Processing
Advances in Non-Linear Modeling for Speech Processing
An Experimentalist’s View of Acoustics and Vibration
Modeling, Measurement and Derivation of Parameters for Airborne and Structure-borne Sound
Advances in Nonlinear Speech Processing

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Uncertainties in Acoustical Transfer Functions
Loudspeakers
Electroacoustic Devices: Microphones and Loudspeakers
Design Criteria for Multiple Loudspeaker Sound System with Realism for Local Congregation
Topics in Acoustic Echo and Noise Control
The Journal of the Acoustical Society of America
Digital Signal Processing in Power Electronics Control Circuits
Selected Methods for the Cancellation of Acoustical Echoes, the Reduction of Background Noise, and Speech Processing

MILLS JIMENEZ

Understanding Acoustics Logos Verlag Berlin GmbH
LoudspeakersFor music recording and reproductionCRC Press

1977 IEEE International Conference on Acoustics, Speech, & Signal Processing, Held at the Sheraton-Hartford Hotel, Hartford, Connecticut, May 9-11, 1977 Taylor & Francis
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).

Time Delay Spectrometry Taylor & Francis

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.

Signals and Systems for Speech and Hearing Springer Science & Business Media

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 loudspeeker arrangement plays a significant role, as it was fou theat the decentralised loudspeeker 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.

5th International Conference on Nonlinear Speech Processing, NoLISP 2011, Las Palmas de Gran Canaria, Spain, November 7-9, 2011, Proceedings Springer Science & Business Media

Proceedings of the NATO Advanced Study Institute on Computational Models of Speech Pattern Processing, held in St. Helier, Jersey, UK, July 7-18, 1997

Academic Press Library in Signal Processing Audio Engineering Soc Incorporated

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.
The Acoustics and Psychoacoustics of Loudspeakers and Rooms Springer Science & Business Media
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.
For music recording and reproduction
Springer Science & Business Media
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.

Journal of the Audio Engineering Society CRC Press

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.

Identification of Nonlinear Systems in Acoustics Springer

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

Nonlinear Analysis, Differential Equations and Control Springer Science & Business Media

"Directory of members" published as pt. 2 of Apr. 1954- issue

Blue Book MDPI

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.

Sound Fields and Transducers BRILL

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.
Acoustics *Halsted Press
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.

International Conference on Non-Linear Speech Processing, NOLISP 2005, Barcelona, Spain, April 19-22, 2005, Revised Selected Papers Springer Science & Business Media 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.

Sound Reproduction Academic Press The theory of linear time-invariant (LTI) systems has been extensively studied over decades and the estimation of any unknown LTI system, knowing both the input and output of the system, is a solved problem. Nevertheless, almost all real-world devices exhibit more or less nonlinear behavior. In the case of very weak nonlinearities, a linear approximation can be used. If the nonlinearities are stronger, the linear approximation fails and systems have to be described using a nonlinear model. The goal of this thesis is to design and develop simple methods for nonlinear systems identification that would be accurate and robust enough to be applicable for analysis and identification of nonlinear systems in several domains, even if the main focus here is on the domain of audio and acoustics. The goal is to identify a nonlinear system and find its generic nonlinear model in such way that the response of the model to any input signal would be the same as the one of the real-world nonlinear system under test. Two methods are developed in the thesis. Both methods are based on Multiple Input - Single Output (MISO) model. The model consists of several parallel branches, each branch consisting of two separated blocks: a nonlinear static function and a linear dynamic filter. The first method uses a white Gaussian noise as the excitation signal for the identification. This method is successfully tested on several simulation examples, but fails when identifying real world nonlinear systems. The second method is based on the nonlinear convolution and uses swept sine excitation signal. This method is successfully tested on several simulation examples. Moreover, it is theoretically shown that it could be used for the identification of systems exhibiting specific dynamical hysteresis (called hysteresis with viscosity-type effect). Two well known real world nonlinear systems (an audio limiter and an acoustic waveguide) are used to validate the second method. The validation is based on the comparison between the output of these real world systems and the output of their estimated models, when excited with the same input signal. The comparison is performed both subjectively, using a simple visual comparison in time or frequency domains, and objectively, using a relative mean square error criterion. Once validated, the method is used in the general frame of the study of electrodynamic loudspeaker quality. Preliminary results show that this method could be used for the nonlinearities loudspeakers identification, and that an inverse filtering minimizing these nonlinearities could possibly be performed with the help of this method.

Electrical & Electronics Abstracts

Springer

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.

An Anthology of the Works of Richard C. Heyser on Measurement, Analysis, and Perception Academic Press
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.

Cumulative Index to Entire IEEE Group Transactions/journals, 1951-1971: Subject Springer

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

Talker Variability in Speech Processing
Routledge
"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.

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