Digital Signal Processing Scilab

Signals, Systems, Transforms, and Digital Signal Processing with MATLAB® has as its principal objective simplification without compromise of rigor. Graphics, called by the author, "the language of scientists and engineers", physical interpretation of subtle mathematical concepts, and a gradual transition from basic to more advanced topics are meant to be among the important contributions of this book. After illustrating the analysis of a function through a step-by-step addition of harmonics, the book deals with Fourier and Laplace transforms. It then covers discrete time signals and systems, the z-transform, continuous- and discrete-time filters, active and passive filters, lattice filters, and continuous- and discrete-time state space models. The author goes on to discuss the Fourier transform of sequences, the discrete Fourier transform, and the fast Fourier transform, followed by Fourier-, Laplace, and z-related transforms, including Walsh–Hadamard, generalized Walsh, Hilbert, discrete cosine, Hartley, Hankel, Mellin, fractional Fourier, and wavelet. He also surveys the architecture and design of digital signal processors, computer architecture, logic design of sequential circuits, and random signals. He concludes with simplifying and demystifying the vital subject of distribution theory. Drawing on much of the author's own research work, this book expands the domains of existence of the most important transforms and thus opens the door to a new world of applications using novel, powerful mathematical tools.
Supplementary files run on UNIX and Windows 95/98/NT

The aim of this book is to introduce the general area of Digital Signal Processing from a practical point of view with a working minimum of mathematics. The emphasis is placed on the practical applications of DSP: implementation issues, tricks and pitfalls. Intuitive explanations and appropriate examples are used to develop a fundamental understanding of DSP theory, laying a firm foundation for the reader to pursue the matter further. The reader will develop a clear understanding of DSP technology in a variety of fields from process control to communications. * Covers the use of DSP in different engineering sectors, from communications to process control * Ideal for a wide audience wanting to take advantage of the strong movement towards digital signal processing techniques in the engineering world * Includes numerous practical exercises and diagrams covering many of the fundamental aspects of digital signal processing

This textbook provides a detailed introduction to the use of software in combination with simple and economical hardware (a sound level meter with calibrated AC output and a digital recording system) to obtain sophisticated measurements usually requiring expensive equipment. It emphasizes the use of free, open source, and multiplatform software. Many commercial acoustical measurement systems use software algorithms as an integral component; however the methods are not disclosed. This book enables the reader to develop useful algorithms and provides insight into the use of digital audio editing tools to document features in the signal. Topics covered include acoustical measurement principles, in-depth critical study of uncertainty applied to acoustical measurements, digital signal processing from the basics, and metrologically-oriented spectral and statistical analysis of signals. The student will gain a deep understanding of the use of software for measurement purposes; the ability to implement software-based measurement systems; familiarity with the hardware necessary to acquire and store signals; an appreciation for the key issue of long-term preservation of signals; and a full grasp of the often neglected issue of uncertainty in acoustical measurements. Pedagogical features include in-text worked-out examples, end-of-chapter problems, a glossary of metrology terms, and extensive appendices covering statistics, proofs, additional examples, file formats, and underlying theory.

Intended for a one-semester junior or senior level undergraduate course, this book provides a modern and self-contained introduction to digital signal processing (DSP). It is supplemented by a vast number of end-of-chapter problems such as worked examples, drill exercises, and application oriented problems that require the use of computational resources such as MATLAB. Also, many figures have been included to help the student grasp and visualize critical concepts. Results are tabulated and summarized for easy reference and access. It also attempts to provide a broader perspective by introducing useful applications and additional special topics in each chapter. These form the background for more advanced graduate courses, and also allow the book to be used as a source of basic reference for professionals across various disciplines interested in DSP.

Many embedded engineers and programmers who need to implement basic process or motion control as part of a product design do not have formal training or experience in control system theory. Although some projects require advanced and very sophisticated control systems expertise, the majority of embedded control problems can be solved without resorting to heavy math and complicated control theory. However, existing texts on the subject are highly mathematical and theoretical and do not offer practical examples for embedded designers. This book is different;it presents mathematical background with sufficient rigor for an engineering text, but it concentrates on providing practical application examples that can be used to design working systems, without needing to fully understand the math and high-level theory operating behind the scenes. The author, an engineer with many years of experience in the application of control system theory to embedded designs, offers a concise presentation of the basics of control theory as it pertains to an embedded environment. Practical, down-to-earth guide teaches engineers to apply practical control theorems without needing to employ rigorous math Covers the latest concepts in control systems with embedded digital controllers

In its 40th year, “Principles of Electronics” remains a comprehensive and succinct textbook for students preparing for B. Tech, B. E., B.Sc., diploma and various other engineering examinations. It also caters to the requirements of those readers who wish to increase their knowledge and gain a sound grounding in the basics of electronics. Concepts fundamental to the understanding of the subject such as electron emission, atomic structure, transistors, semiconductor physics, gas-filled tubes, modulation and demodulation, semiconductor diode and regulated D.C. power supply have been included, added and updated in the book as full chapters to give the reader a well-rounded view of the subject.

This supplement to any standard DSP text is one of the first books to successfully integrate the use of MATLAB® in the study of DSP concepts. In this book, MATLAB® is used as a computing tool to explore traditional DSP topics, and solve problems to gain insight. This greatly expands the range and complexity of problems that students can effectively study in the course. Since DSP applications are primarily algorithms implemented on a DSP processor or software, a fair amount of programming is required. Using interactive software such as MATLAB® makes it possible to place more emphasis on learning new and difficult concepts than on programming algorithms. Interesting practical examples are discussed and useful problems are explored. This updated second edition includes new homework problems and revises the scripts in the book, available functions, and m-files to MATLAB® V7.

The Designer's Guide to the Cortex-M Microcontrollers gives you an easy-to-understand introduction to the concepts required to develop programs in C with a Cortex-M based microcontroller. The book begins with an overview of the Cortex-M family, giving architectural descriptions supported with practical examples, enabling you to easily develop basic C programs to run on the Cortex-M0/M0+/M3 and M4 and M7. It then examines the more advanced features of the Cortex architecture such as memory protection, operating modes, and dual stack operation. Once a firm grounding in the Cortex-M processor has been established the book introduces the use of a small footprint RTOS and the CMSIS-DSP library. The book also examines techniques for software testing and code reuse specific to Cortex-M microcontrollers. With this book you will learn: the key differences between the Cortex-M0/M0+/M3 and M4 and M7; how to write C programs to run on Cortex-M based processors; how to make the best use of the CoreSight debug system; the Cortex-M operating modes and memory protection; advanced software techniques that can be used on Cortex-M microcontrollers; how to use a Real Time Operating System with Cortex-M devices; how to optimize DSP code for the Cortex-M4; and how to build real time DSP systems. Includes an update to the latest version (5) of MDK-ARM, which introduces the concept of using software device packs and software components Includes overviews of the new CMSIS specifications Covers developing software with CMSIS-RTOS showing how to use RTOS in a real world design Provides a new chapter on the Cortex-M7 architecture covering all the new features Includes a new chapter covering test driven development for Cortex-M microcontrollers Features a new chapter on creating software components with CMSIS-Pack and device abstraction with CMSIS-Driver Features a new chapter providing an overview of the ARMv8-M architecture including the TrustZone hardware security model

Starts with an overview of today's FPGA technology, devices, and tools for designing state-of-the-art DSP systems. A case study in the first chapter is the basis for more than 30 design examples throughout. The following chapters deal with computer arithmetic concepts, theory and the implementation of FIR and IIR filters, multirate digital signal processing systems, DFT and FFT algorithms, and advanced algorithms with high future potential. Each chapter contains exercises. The VERILOG source code and a glossary are given in the appendices, while the accompanying CD-ROM contains the examples in VHDL and Verilog code as well as the newest Altera "Baseline" software. This edition has a new chapter on adaptive filters, new sections on division and floating point arithmetics, an up-date to the current Altera software, and some new exercises.

This book comprises selected peer-reviewed papers from the International Conference on VLSI, Signal Processing, Power Systems, Illumination and Lighting Control, Communication and Embedded Systems (VSPICE-2019). The contents are divided into five broad topics - VLSI and embedded systems, signal processing, power systems, illumination and control, and communication and networking. The book focuses on the latest innovations, trends, and challenges encountered in the different areas of electronics and communication, and electrical engineering. It also offers potential solutions and provides an insight into various emerging areas such as image fusion, bio-sensors, and underwater sensor networks. This book can prove to be useful for academics and professionals interested in the various sub-fields of electronics and communication engineering.

Meant for students and practicing engineers, this book provides a clear, comprehensive and up-to-date introduction to Digital Image

Processing in a pragmatic style. An illustrative approach, practical examples and MATLAB applications given in the book help in bringing the theory to life.

This revised and extended second edition covers problems concerning the design and realization of digital control algorithms for power electronics circuits using digital signal processing (DSP) methods. This book discusses signal processing, starting from analog signal acquisition, through conversion to digital form, methods of 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, a three-phase shunt active power filter and a digital class-D audio power amplifier. The book bridges the gap between power electronics and digital signal processing. Many control algorithms and circuits for power electronics in the current literature are described using analog transmittances. This may not always be acceptable, especially if half of the sampling frequencies and half of the power transistor switching frequencies are close to the band of interest. Therefore in this book, a digital circuit is treated as a digital circuit with its own peculiar characteristics, rather than an analog circuit. This helps to avoid errors and instability. This edition includes a new chapter dealing with selected problems of simulation of power electronics systems together with digital control circuits. The book includes numerous examples using MATLAB and PSIM programs.

This book provides basic theories and implementations using SCILAB open-source software for digital images. The book simplifies image processing theories and well as implementation of image processing algorithms, making it accessible to those with basic knowledge of image processing. This book includes many SCILAB programs at the end of each theory, which help in understanding concepts. The book includes more than sixty SCILAB programs of the image processing theory. In the appendix, readers will find a deeper glimpse into the research areas in the image processing.

Briefly describes the physical characteristics, the habitat, and the behavior of the Alaskan brown bear.

This textbook provides comprehensive coverage for courses in the basics of design and implementation of digital filters. The book assumes only basic knowledge in digital signal processing and covers state-of-the-art methods for digital filter design and provides a simple route for the readers to design their own filters. The advanced mathematics that is required for the filter design is minimized by providing an extensive MATLAB toolbox with over 300 files. The book presents over 200 design examples with MATLAB code and over 300 problems to be solved by the reader. The students can design and modify the code for their use. The book and the design examples cover almost all known design methods of frequency-selective digital filters as well as some of the authors' own, unique techniques.

In addition to its thorough coverage of DSP design and programming techniques, Smith also covers the operation and usage of DSP chips. He uses Analog Devices' popular DSP chip family as design examples. Covers all major DSP topics Full of insider information and shortcuts Basic techniques and algorithms explained without complex numbers

The signal processing task is a very critical issue in the majority of new technological inventions and challenges in a variety of applications in both science and engineering fields. Classical signal processing techniques have largely worked with mathematical models that are linear, local, stationary, and Gaussian. They have always favored closed-form tractability over real-world accuracy. These constraints were imposed by the lack of powerful computing tools. During the last few decades, signal processing theories, developments, and applications have matured rapidly and now include tools from many areas of mathematics, computer science, physics, and engineering. This book is targeted primarily toward both students and researchers who want to be exposed to a wide variety of signal processing techniques and algorithms. It includes 27 chapters that can be categorized into five different areas depending on the application at hand. These five categories are ordered to address image processing, speech processing, communication systems, time-series analysis, and educational packages respectively. The book has the advantage of providing a collection of applications that are completely independent and self-contained; thus, the interested reader can choose any chapter and skip to another without losing continuity.

About the Book : - Digital Signal Processing Fundamentals Digital Signal Processing (DSP), as the term suggests, is the processing of signals using digital computers. These signals might be anything transferred from an analog domain to a digital form (e.g., temperature and pressure sensors, voices over a telephone, images from a camera, or data transmittal though computes). As a result, understanding the whole spectrum of DSP technology can be a daunting task for electrical engineering professionals and students alike. Digital Signal Processing Fundamentals provides a comprehensive look at DSP by introducing the important mathematical processes and then providing several application-specific tutorials for practicing the techniques learned. Beginning with general theory, including Fourier Analysis, the mathematics of complex numbers, Fourier transforms, differential equations, analog and digital filters, and much more; the book then delves into Matlab and Scilab tutorials with examples on solving practical engineering problems, followed by software applications on image processing and audio processing - complete with all the algorithms and source code. This is an invaluable resource for anyone seeking to understand how DSP works. Features: Provides a comprehensive overview and introduction of digital signal processing technology. Provides application with software algorithms Explains the concept of Nyquist frequency, orthogonal functions and method of finding Fourier coefficients Includes a CD-ROM with the source code for the projects plus Matlab and Scilab that generate graphs, figures in the book, and third party application software Discusses the techniques of digital filtering and windowing of input data, including: Butterwoth, Chebyshev, and elliptic filter formulation. Table Of Contents : Fourier Analysis Complex Number Arithmetic The Fourier Transform Solutions of Differential Equations Laplace Transforms and z-Tranforms Filter Design Digital Filters The FIR Filters Appendix A : Matlab Tutorial Appendix B : Scilab Tutorial Appendix C : Digital Filter Applications Appendix D : About the CD-ROM Appendix E : Software Licenses Appendix F : Bibliography Index About Author :- Ashfaq A. Khan (Baton Rouge, LA) is a senior software engineer for LIGO Livingston Observatory, with over 20 years of experience in system design. He has conducted several workshop and is the author of Practical Linux Programming: Device Drivers, Embedded Systems, and the Internet.

Cryptography is hard, but it’s less hard when it’s filled with adorable Japanese manga. The latest addition to the Manga Guide series, The Manga Guide to Cryptography, turns the art of encryption and decryption into plain, comic illustrated English. As you follow Inspector Jun Meguro in his quest to bring a cipher-wielding thief to justice, you’ll learn how cryptographic ciphers work. (Ciphers are the algorithms at the heart of cryptography.) Like all books in the Manga Guide series, The Manga Guide to Cryptography is illustrated throughout with memorable Japanese manga as it dives deep into advanced cryptography topics, such as classic substitution, polyalphabetic, and transposition ciphers; symmetric-key algorithms like block and DES (Data Encryption Standard) ciphers; and how to use public key encryption technology. It also explores practical applications of encryption such as digital signatures, password security, and identity fraud countermeasures. The Manga Guide to Cryptography is the perfect introduction to cryptography for programmers, security professionals, aspiring cryptographers, and anyone who finds cryptography just a little bit hard.

This book provides comprehensive, graduate-level treatment of analog and digital signal analysis suitable for course use and self-guided learning. This expert text guides the reader from the basics of signal theory through a range of application tools for use in acoustic analysis, geophysics, and data compression. Each concept is introduced and explained step by step, and the necessary mathematical formulae are integrated in an accessible and intuitive way. The first part of the book explores how analog systems and signals form the basics of signal analysis. This section covers Fourier series and integral transforms of analog signals, Laplace and Hilbert transforms, the main analog filter classes, and signal modulations. Part II covers digital signals, demonstrating their key advantages. It presents z and Fourier transforms, digital filtering, inverse filters, deconvolution, and parametric modeling for deterministic signals. Wavelet decomposition and reconstruction of non-stationary signals are also discussed. The third part of the book is devoted to random signals, including spectral estimation, parametric modeling, and Tikhonov regularization. It covers statistics of one and two random variables and the principles and methods of spectral analysis. Estimation of signal properties is discussed in the context of ergodicity conditions and parameter estimations, including the use of Wiener and Kalman filters. Two appendices cover the basics of integration in the complex plane and linear algebra. A third appendix presents a basic Matlab toolkit for computer signal analysis. This expert text provides both a solid theoretical understanding and tools for real-world applications.

Digital Signal Processing, Second Edition enables electrical engineers and technicians in the fields of biomedical, computer, and electronics engineering to master the essential fundamentals of DSP principles and practice. Many instructive worked examples are used to illustrate the material, and the use of mathematics is minimized for easier grasp of concepts. As such, this title is also useful to undergraduates in electrical engineering, and as a reference for science students and practicing engineers. The book goes beyond DSP theory, to show implementation of algorithms in hardware and software. Additional topics covered include adaptive filtering with noise reduction and echo cancellations, speech compression, signal sampling, digital filter realizations, filter design, multimedia applications, over-sampling, etc. More advanced topics are also covered, such as adaptive filters, speech compression such as PCM, u-law, ADPCM, and multi-rate DSP and over-sampling ADC. New to this edition: MATLAB projects dealing with practical applications added throughout the book New chapter (chapter 13) covering sub-band coding and wavelet transforms, methods that have become popular in the DSP field New applications included in many chapters, including applications of DFT to seismic signals, electrocardiography data, and vibration signals All real-time C programs revised for the TMS320C6713 DSK Covers DSP principles with emphasis on communications and control applications Chapter objectives, worked examples, and end-of-chapter exercises aid the reader in grasping key concepts and solving related problems Website with MATLAB programs for simulation and C programs for real-time DSP

Scilab and its Scicos block diagram graphical editor, with a special emphasis on modeling and simulation tools. The first part is a detailed Scilab tutorial, and the second is dedicated to modeling and simulation of dynamical systems in Scicos. The concepts are illustrated through numerous examples, and all code used in the book is available to the reader.

Digital Image Processing using SCILABSpringer

Combining clear explanations of elementary principles, advanced topics and applications with step-by-step mathematical derivations, this textbook provides a comprehensive yet accessible introduction to digital signal processing. All the key topics are covered, including discrete-time Fourier transform, z-transform, discrete Fourier transform and FFT, A/D conversion, and FIR and IIR filtering algorithms, as well as more advanced topics such as multirate systems, the discrete cosine transform and spectral signal processing. Over 600 full-color illustrations, 200 fully worked examples, hundreds of end-of-chapter homework problems and detailed computational examples of DSP algorithms implemented in MATLAB® and C aid understanding, and help put knowledge into practice. A wealth of supplementary material accompanies the book online, including interactive programs for instructors, a full set of solutions and MATLAB® laboratory exercises, making this the ideal text for senior undergraduate and graduate courses on digital signal processing.

Signals and Systems by Nahvi is intended for use in a signals and systems course at the undergraduate junior level. The book covers the analysis of signals and linear systems in the time and frequency domains and is organized into 18 chapters. The chapters are modular with sections and there are no sub-sections. The modular structure of the chapters provides a quick and direct approach to each topic within the chapters and makes the book a convenient tool for instructional needs in a wide range of teaching scenarios and at various levels of complexity. Continuous-time and discrete-time domains are treated separately in two parts. This allows the book to be used for instructions on either domain separately. It may also be used for courses teaching the two domains simultaneously, as the chapters in part one and two provide parallel presentations of each subject.

In recent years, there has been considerable interest in highly integrated, low power, portable wireless devices. This monograph focuses on the problem of low power GFSK/GMSK modulation and presents an architectural approach for improved performance. Including several valuable tools for the practicing engineer.

If you understand basic mathematics and know how to program with Python, you're ready to dive into signal processing. While most resources start with theory to teach this complex subject, this practical book introduces techniques by showing you how they're applied in the real world. In the first chapter alone, you'll be able to decompose a sound into its harmonics, modify the harmonics, and generate new sounds. Author Allen Downey explains techniques such as spectral decomposition, filtering, convolution, and the Fast Fourier Transform. This book also provides exercises and code examples to help you understand the material. You'll explore: Periodic signals and their spectrums Harmonic structure of simple waveforms Chirps and other sounds whose spectrum changes over time Noise signals and natural sources of noise The autocorrelation function for estimating pitch The discrete cosine transform (DCT) for compression The Fast Fourier Transform for spectral analysis Relating operations in time to filters in the frequency domain Linear time-invariant (LTI) system theory Amplitude modulation (AM) used in radio Other books in this series include Think Stats and Think Bayes, also by Allen Downey.

In the past few years Biomedical Engineering has received a great deal of attention as one of the emerging technologies in the last decade and for years to come, as witnessed by the many books, conferences, and their proceedings. Media attention, due to the applications-oriented advances in Biomedical Engineering, has also increased. Much of the excitement comes from the fact that technology is rapidly changing and new technological adventures become available and feasible every day. For many years the physical sciences contributed to medicine in the form of expertise in radiology and slow but steady contributions to other more diverse fields, such as computers in surgery and diagnosis, neurology, cardiology, vision and visual prosthesis, audition and hearing aids, artificial limbs, biomechanics, and biomaterials.

The list goes on. It is therefore hard for a person unfamiliar with a subject to separate the substance from the hype. Many of the applications of Biomedical Engineering are rather complex and difficult to understand even by the not so novice in the field. Much of the hardware and software tools available are either too simplistic to be useful or too complicated to be understood and applied. In addition, the lack of a common language between engineers and computer scientists and their counterparts in the medical profession, sometimes becomes a barrier to progress.

Quickly Engages in Applying Algorithmic Techniques to Solve Practical Signal Processing Problems With its active, hands-on learning approach, this text enables readers to master the underlying principles of digital signal processing and its many applications in industries such as digital television, mobile and broadband communications, and medical/scientific devices. Carefully developed MATLAB® examples throughout the text illustrate the mathematical concepts and use of digital signal processing algorithms. Readers will develop a deeper understanding of how to apply the algorithms by manipulating the codes in the examples to see their effect. Moreover, plenty of exercises help to put knowledge into practice solving real-world signal processing challenges. Following an introductory chapter, the text explores: Sampled signals and digital processing Random signals Representing signals and systems Temporal and spatial signal processing Frequency analysis of signals Discrete-time filters and recursive filters Each chapter begins with chapter objectives and an introduction. A summary at the end of each chapter ensures that one has mastered all the key concepts and techniques before progressing in the text. Lastly, appendices listing selected web resources, research papers, and related textbooks enable the investigation of individual topics in greater depth. Upon completion of this text, readers will understand how to apply key algorithmic techniques to address practical signal processing problems as well as develop their own signal processing algorithms. Moreover, the text provides a solid foundation for evaluating and applying new digital processing signal techniques as they are developed.

The rapid development in various fields of Digital Audio Effects, or DAFX, has led to new algorithms and this second edition of the popular book, DAFX: Digital Audio Effects has been updated throughout to reflect progress in the field. It maintains a unique approach to DAFX with a lecture-style introduction into the basics of effect processing. Each effect description begins with the presentation of the physical and acoustical phenomena, an explanation of the signal processing techniques to achieve the effect, followed by a discussion of musical applications and the control of effect parameters. Topics covered include: filters and delays, modulators and demodulators, nonlinear processing, spatial effects, time-segment processing, time-frequency processing, source-filter processing, spectral processing, time and frequency warping musical signals. Updates to the second edition include: Three completely new chapters devoted to the major research areas of: Virtual Analog Effects, Automatic Mixing and Sound Source Separation, authored by leading researchers in the field . Improved presentation of the basic concepts and explanation of the related technology. Extended coverage of the MATLABTM scripts which demonstrate the implementation of the basic concepts into software programs. Companion website (www.dafx.de) which serves as the download source for MATLABTM scripts, will be updated to reflect the new material in the book. Discussing DAFX from both an introductory and advanced level, the book systematically introduces the reader to digital signal processing concepts, how they can be applied to sound and their use in musical effects. This makes the book suitable for a range of professionals including those working in audio engineering, as well as researchers and engineers involved in the area of digital signal processing along with students on multimedia related courses.

In this book the reader will find a collection of chapters authored/co-authored by a large number of experts around the world, covering the broad field of digital signal processing. This book intends to provide highlights of the current research in the digital signal processing area, showing the recent advances in this field. This work is mainly destined to researchers in the digital signal processing and related areas but it is also accessible to anyone with a scientific background desiring to have an up-to-date overview of this domain. Each chapter is self-contained and can be read independently of the others. These nineteenth chapters present methodological advances and recent applications of digital signal processing in various domains as communications, filtering, medicine, astronomy, and image processing.

New edition of a text intended primarily for the undergraduate courses on the subject which are frequently found in electrical engineering curricula--but the concepts and techniques it covers are also of fundamental importance in other engineering disciplines. The book is structured to develop in parallel the methods of analysis for continuous-time and discrete-time signals and systems, thus allowing exploration of their similarities and differences. Discussion of applications is emphasized, and numerous worked examples are included. Annotation copyrighted by Book News, Inc., Portland, OR

Artificial Neural Networks for Engineering Applications presents current trends for the solution of complex engineering problems that cannot be solved through conventional methods. The proposed methodologies can be applied to modeling, pattern recognition, classification, forecasting, estimation, and more. Readers will find different methodologies to solve various problems, including complex nonlinear systems, cellular computational networks, waste water treatment, attack detection on cyber-physical systems, control of UAVs, biomechanical and biomedical systems, time series forecasting, biofuels, and more. Besides the real-time implementations, the book contains all the theory required to use the proposed methodologies for different applications. Presents the current trends for the solution of complex engineering problems that cannot be solved through conventional methods Includes real-life scenarios where a wide range of artificial neural network architectures can be used to solve the problems encountered in engineering Contains all the theory required to use the proposed methodologies for different applications