Digital Signal Processing In Rf Applications Uspas

A wireless communication system employs a radio frequency (RF) wave to transmit information bearing signals. In modern digital communication systems, sophisticated modulation techniques are developed to modulate information onto an RF carrier waveform, so as to transmit more information. This new book presents signal processing techniques for reducing impairments of analog and RF circuits in wireless communications systems. Engineers, researchers, and students will find full coverage of the topic, including vector modulators, power amplifiers, vector demodulators, group delay distortion in analog/RF filters, digital beamforming networks, and dual polarization systems. Several applications are discussed, including both single carrier and multi-carrier scenarios.

An engineer's introduction to concepts, algorithms, and advancements in Digital Signal Processing. This lucidly written resource makes extensive use of real-world examples as it covers all the important design and engineering references.

Previous space-to-ground, single-platform geolocation experiments exploiting time-difference-of arrival (TDOA) via interferometry were successful at separating and quantitatively characterizing interfering radio frequency (RF) signals from expected RF transmissions. Much of the success of these experiments rested on the use of embedded processors to perform the required signal processing. The experiments handled data in a 'snapshot' fashion: digitized data was collected, the data was processed via a digital signal processing (DSP) microprocessor to yield differential phase measurements, and these measurements were transmitted to the Earth for geolocation processing. With the utilization of FPGAs (field programmable gate arrays) for the intensive number-crunching algorithms, the processing of streaming real-time data is feasible for bandwidths on the order of 20 MHz. By partitioning the signal processing algorithm so there is a significant reduction in the data rate as data flows through the FPGA, a DSP microprocessor can now be employed to perform further decision-oriented processing on the FPGA output. This hybrid architecture, employing both FPGAs and DSPs, typically requires an expensive and lengthy development cycle. However, the use of graphical development environments with auto-code generation and hardware-in-the-loop testing can result in rapid prototyping for geolocation experiments, which enables adaptation to emerging signals of interest in a cost and time effective manner.

Considering the rapid evolution of digital signal processing (DSP), those studying this field require an easily understandable text that complements practical software and hardware applications with sufficient coverage of theory. Designed to keep pace with advancements in the field and elucidate lab work, Digital Signal Processing Laboratory, Second Edition was developed using material and student input from courses taught by the author. Contains a new section on digital filter structure Honed over the past several years, the information presented here reflects the experience and insight the author gained on how to convey the subject of DSP to senior undergraduate and graduate students coming from varied subject backgrounds. Using feedback from those students and faculty involved in these courses, this book integrates simultaneous training in both theory and practical software/hardware aspects of DSP. The practical component of the DSP course curriculum has proven to greatly enhance understanding of the basic theory and principles. To this end, chapters in the text contain sections on: Theory—Explaining the underlying mathematics and principles Problem solving—Offering an ample amount of workable problems for the reader Computer laboratory—Featuring programming examples and exercises in MATLAB® and Simulink® Hardware laboratory—Containing exercises that employ test and measurement equipment, as well as the Texas Instruments TMS320C6711DSP Starter Kit The text covers the progression of the Discrete and Fast Fourier transforms (DFT and FFT). It also addresses Linear Time-Invariant (LTI) discrete-time signals and systems, as well as the mathematical tools used to describe them. The author includes appendices that give detailed descriptions of hardware along with instructions on how to use the equipment featured in the book.

Digital Signal Processing in Modern Communication Systems takes you on a journey that starts with basic DSP principles and ends with a treatment of modern wireless modems like OFDM and single-tone transceivers. Throughout this journey, we will cover signal processing topics that are applicable not just to the field of communications but to many engineering disciplines. This text steps outside the often dry mathematical presentation of more traditional DSP books and provides a more intuitive approach to this fascinating topic. Some of this book's uniqueness can be summarized as follows: - An intuitive approach to the topic of digital signal processing. - Working in-book MatLab examples supporting all important concepts. - A large scope covering basic concepts (correlation, convolution, DFT, FIR filters ...) as well as advanced topics (optimization, adaptive signal processing, equalization, OFDM, MIMO ...). - MatLab modeling of analog/RF effects (multipath channel, thermal noise, phase noise, IQ imbalances, DC and frequency offsets) that must be addressed and solved in modern modem design. - Real world topics that go beyond the ordinary communication textbooks such as signal synchronization, modem rate management, and fixed-point effects. All in all, this book is a must-have for students and practicing engineers who want to build upon the principles of Digital Signal Processing, enrich their understanding with advanced topics, and then apply that knowledge to the design of modern wireless modems.

The approach adopted in Digital Synthesizers and Transmitters for Software Radio will provide an understanding of key areas in the field of digital synthesizers and transmitters. It is easy to include different digital techniques in the digital synthesizers and transmitters by using digital signal processing methods, because the signal is in digital form. By programming the digital synthesizers and transmitters, adaptive channel bandwidths, modulation formats, frequency hopping and data rates are easily achieved. Techniques such as digital predistortion for power amplifier linearization, digital compensation methods for analog I/Q modulator nonlinearities and digital power control and ramping are presented in this book. The flexibility of the digital synthesizers and transmitters makes them ideal as signal generators for software radio. Software radios represent a major change in the design paradigm for radios in which a large portion of the functionality is implemented through programmable signal processing devices, giving the radio the ability to change its operating parameters to accommodate new features and capabilities. A software radio approach reduces the content of radio frequency (RF) and other analog components of traditional radios and emphasizes digital signal processing to enhance overall transmitter flexibility. Software radios are emerging in commercial and military infrastructure.

This hands-on, laboratory driven textbook helps readers understand principles of digital signal processing (DSP) and basics of software-based digital communication, particularly software-defined networks (SDN) and software-defined radio (SDR). In the book only the most important concepts are presented. Each book chapter is an introduction to computer laboratory and is accompanied by complete laboratory exercises and ready-to-go Matlab programs with figures and comments (available at the book webpage and running also in GNU Octave 5.2 with free software packages), showing all or most details of relevant algorithms. Students are tasked to understand programs, modify them, and apply presented concepts to recorded real RF signal or simulated received signals, with modelled transmission condition and hardware imperfections. Teaching is done by showing examples and their modifications to different real-world telecommunication-like applications. The book consists of three parts: introduction to DSP (spectral analysis and digital filtering), introduction to DSP advanced topics (multi-rate, adaptive, model-based and multimedia - speech, audio, video - signal analysis and processing) and introduction to software-defined modern telecommunication systems (SDR technology, analog and digital modulations, single- and multi-carrier systems, channel estimation and correction as well as synchronization issues). Many real signals are processed in the book, in the first part - mainly speech and audio, while in the second part - mainly RF recordings taken from RTL-SDR USB stick and ADALM-PLUTO module, for example captured IQ data of VOR avionics signal, classical FM radio with RDS, digital DAB/DAB+ radio and 4G-LTE digital telephony. Additionally, modelling and simulation of some transmission scenarios are tested in software in the book, in particular TETRA, ADSL and 5G signals. Provides an introduction to digital signal processing and software-based digital communication; Presents a transition from digital signal processing to software-defined telecommunication; Features a suite of pedagogical materials including a laboratory test-bed and computer exercises/experiments .

Real-time testing and simulation of open- and closed-loop radio frequency (RF) systems for signal generation, signal analysis and digital signal processing require deterministic, low-latency, high-throughput capabilities afforded by user reconfigurable field programmable gate arrays (FPGAs). This comprehensive book introduces LabVIEW FPGA, provides best practices for multi-FPGA solutions, and guidance for developing high-throughput, low-latency FPGA based RF systems. Written by a recognized expert with a wealth of real-world experience in the field, this is the first book written on the subject of FPGAs for radar and other RF applications.

Digital signal processing lies at the heart of the communications revolution and is an essential element of key technologies such as mobile phones and the Internet. This book covers all the major topics in digital signal processing (DSP) design and analysis, supported by MatLab examples and other modelling techniques. The authors explain clearly and concisely why and how to use digital signal processing systems; how to approximate a desired transfer function characteristic using polynomials and ratio of polynomials; why an appropriate mapping of a transfer function on to a suitable structure is important for practical applications; and how to analyse, represent and explore the trade-off between time and frequency representation of signals. An ideal textbook for students, it will also be a useful reference for engineers working on the development of signal processing systems.

This book presents theory, design methods and novel applications for integrated circuits for analog signal processing. The discussion covers a wide variety of active devices, active elements and amplifiers, working in voltage mode, current mode and mixed mode. This includes voltage operational amplifiers, current operational amplifiers, operational transconductance amplifiers, operational transresistance amplifiers, current conveyors, current differencing transconductance amplifiers, etc. Design methods and challenges posed by nanometer technology are discussed and applications described, including signal amplification, filtering, data acquisition systems such as neural recording, sensor conditioning such as biomedical implants, actuator conditioning, noise generators, oscillators, mixers, etc. Presents analysis and synthesis methods to generate all circuit topologies from which the designer can select the best one for the desired application; Includes design guidelines for active devices/elements with low voltage and low power constraints; Offers guidelines for selecting the right active devices/elements in the design of linear and nonlinear circuits; Discusses optimization of the active devices/elements for process and manufacturing issues of nanometer technology.

This book details design procedures used in creating a radio communication system using FPGA, ADC and DAC devices for high speed radio frequency signal processing. The procedures and calculation methodologies are applicable to a wide range of radio communication products and are demonstrated in a prototype that operates at 13.45 MHz, but can receive signals up to 450 MHz and transmit up to 40 MHz using various alias responses. Digital signal processing technology has migrated from initial audio applications and now offers cost effective, high performance signal processing directly at radio communication frequencies. Mixed signal devices, such as RF ADC and DAC's are available with resolutions up to 16 bits and some can sampling at 2 GHz. When combined with an FPGA, flexible radio architectures are realized. This book presents the reader with a real "walk through" design approach that covers essential concepts required for real world radio design using digital algorithms that process information conveyed by radio signals.This approach offers modulation format flexibility, programmable bandwidth and multiple channel processing.

This is the second volume in a trilogy on modern Signal Processing. The three books provide a concise exposition of signal processing topics, and a guide to support individual practical exploration based on MATLAB programs. This second book focuses on recent developments in response to the demands of new digital technologies. It is divided into two parts: the first part includes four chapters on the decomposition and recovery of signals, with special emphasis on images. In turn, the second part includes three chapters and addresses important data-based actions, such as adaptive filtering, experimental modeling, and classification.

Today digital signal processing systems use advanced CMOS technologies requiring the analog-to-digital converter to be implemented in the same (digital) technology. Such an implementation requires special circuit techniques. Furthermore the susceptibility of converters to ground bounce or digital noise is an important design criterion. In this part different converters and conversion techniques are described that are optimized for receiver applications. Part II, Sensor and Actuator Interfaces, interfaces for sensors and actuators shape the gates through which information is acquired from the real world into digital information systems, and vice versa. The interfaces should include analog signal conditioning, analog-to-digital conversion, digital bus interfaces and data-acquisition networks. To simplify the use of data-acquisition systems additional features should be incorporated, like self-test, and calibration

Wireless Receiver Architectures and Design presents the various designs and architectures of wireless receivers in the context of modern multi-mode and multi-standard devices. This one-stop reference and guide to designing low-cost low-power multi-mode, multi-standard receivers treats analog and digital signal processing simultaneously, with equal detail given to the chosen architecture and modulating waveform. It provides a complete understanding of the receiver's analog front end and the digital backend, and how each affects the other. The book explains the design process in great detail, starting from an analysis of requirements to the choice of architecture and finally to the design and algorithm development. The advantages and disadvantages of each wireless architecture and the suitability to a standard are given, enabling a better choice of design methodology, receiver lineup, analog block, and digital algorithm for a particular architecture. Whether you are a communications engineer working in system architecture and waveform design, an RF engineer working on noise and linearity budget and line-up analysis, a DSP engineer working on algorithm development, or an analog or digital design engineer designing circuits for wireless transceivers, this book is your one-stop reference and guide to designing low-cost low-power multi-mode multi-standard receivers. The material in this book is organized and presented to lead you from applied theory to practical design with plenty of examples and case studies drawn from modern wireless standards. Provides a complete description of receiver architectures together with their pros and cons, enabling a better choice of design methodology Covers the design trade-offs and algorithms between the analog front end and the digital modem – enabling an end-to-end design approach Addresses multi-mode multi-standard low-cost, low-power radio design – critical for producing the applications for Smart phones and portable internet devices

Summarizes cutting-edge physical layer technologies for multi-mode wireless RF transceivers. Includes original contributions from distinguished researchers and professionals. Covers cutting-edge physical layer technologies for multi-mode wireless RF transceivers. Contributors are all leading researchers and professionals in this field.

Digital Signal Processing 101: Everything You Need to Know to Get Started provides a basic tutorial on digital signal processing (DSP). Beginning with discussions of numerical representation and complex numbers and exponentials, it goes on to explain difficult concepts such as sampling, aliasing, imaginary numbers, and frequency response. It does so using easy-to-understand examples and a minimum of mathematics. In addition, there is an overview of the DSP functions and implementation used in several DSP-intensive fields or applications, from error correction to CDMA mobile communication to airborne radar systems. This book is intended for those who have absolutely no previous experience with DSP, but are comfortable with high-school-level math skills. It is also for those who work in or provide components for industries that are made possible by DSP. Sample industries include wireless mobile phone and infrastructure equipment, broadcast and cable video, DSL modems, satellite communications, medical imaging, audio, radar, sonar, surveillance, and electrical motor control. Dismayed when presented with a mass of equations as an explanation of DSP? This is the book for you! Clear examples and a non-mathematical approach gets you up to speed with DSP Includes an overview of the DSP functions and implementation used in typical DSP-intensive applications, including error correction, CDMA mobile communication, and radar systems

LabVIEW Digital Signal Processing teaches engineers how to use the graphical programming language to create virtual instruments to handle to most sophisticated DSP applications. From basic filters to complex sampling mechanisms to signal generators, LabVIEW virtual instruments (VIs) can make DSP work faster and much less expensive a particular boon to the many engineers working on cutting edge communications systems.

The use of present Digital Signal Processing (DSP) techniques can drastically reduce the residual rf amplitude and phase error in an accelerating rf cavity. Accelerator beam loading contributes greatly to this residual error, and the low-level rf field control loops cannot completely absorb the fast transient of the error. A feedforward technique using DSP is required to maintain the very stringent rf field amplitude and phase specifications. 7 refs.

RF and Digital Signal Processing for Software-Defined RadioA Multi-Standard Multi-Mode ApproachNewnes

This book discusses the trade-offs involved in designing direct RF digitization receivers for the radio frequency and digital signal processing domains. A system-level framework is developed, quantifying the relevant impairments of the signal processing chain, through a comprehensive system-level analysis. Special focus is given to noise analysis (thermal noise, quantization noise, saturation noise, signal-dependent noise), broadband non-linear distortion analysis, including the impact of the sampling strategy (low-pass, band-pass), analysis of time-interleaved ADC channel mismatches, sampling clock purity and digital channel selection. The system-level framework described is applied to the design of a cable multi-channel RF direct digitization receiver. An optimum RF signal conditioning, and some algorithms (automatic gain control loop, RF front-end amplitude equalization control loop) are used to relax the requirements of a 2.7GHz 11-bit ADC. A two-chip implementation is presented, using BiCMOS and 65nm CMOS processes, together with the block and system-level measurement results. Readers will benefit from the techniques presented, which are highly competitive, both in terms of cost and RF performance, while drastically reducing power consumption.

Digital signal processing plays a central role in the development of modern communication and information processing systems. The theory and application of signal processing is concerned with the identification, modelling and utilisation of patterns and structures in a signal process. The observation signals are often distorted, incomplete and noisy and therefore noise reduction, the removal of channel distortion, and replacement of lost samples are important parts of a signal processing system. The fourth edition of Advanced Digital Signal Processing and Noise Reduction updates and extends the chapters in the previous edition and includes two new chapters on MIMO systems, Correlation and Eigen analysis and independent component analysis. The wide range of topics covered in this book include Wiener filters, echo cancellation, channel equalisation, spectral estimation, detection and removal of impulsive and transient noise, interpolation of missing data segments, speech enhancement and noise/interference in mobile communication environments. This book provides a coherent and structured presentation of the theory and applications of statistical signal processing and noise reduction methods. Two new chapters on MIMO systems, correlation and Eigen analysis and independent component analysis Comprehensive coverage of advanced digital signal processing and noise reduction methods for communication and information processing systems Examples and applications in signal and information extraction from noisy data Comprehensive but accessible coverage of signal processing theory including probability models, Bayesian inference, hidden Markov models, adaptive filters and Linear prediction models Advanced Digital Signal Processing and Noise Reduction is an invaluable text for postgraduates, senior undergraduates and researchers in the fields of digital signal processing, telecommunications and statistical data analysis. It will also be of interest to professional engineers in telecommunications and audio and signal processing industries and network planners and implementers in mobile and wireless communication communities.

This thesis introduces a system that directly synthesizes the modulation signal at the intermediate frequency (IF) or radio frequency (RF) in the digital domain. In this system, the multi-bit digital signal is transformed into a 1bit stream binary sequence by using the digital parallel band pass Sigma-Delta (BPSD) modulator. The synthesized bit-stream sequence

can drive a simple switch (i.e., a 1bit DAC) or a switch-mode power amplifier (PA) to improve efficiency by reducing the complexity of the analog circuits. The desired signal can finally be reconstructed by the linear passive filter and thus, the linearity is not affected by the non-linearity of the devices, i.e., the switch mode PA. This system also has advantages in two other aspects: 1) the multiplier-less structure implemented in this system improves hardware efficiency and speed performance; 2) mismatches between the conversion elements in the conventional multi-bit DAC are avoided. On the other hand, this implementation introduces new problems that need to be addressed. At the system level, the interleaved timing relationship between the in-phase (I) and quadrature (Q) paths in the parallel BPSD modulator can raise undesired image components in the synthesized signal. This thesis addresses this problem by combining the use of the CIC and the Lagrange filter to suppress the image at the mirror frequency with hardware efficiency. To increase the center frequency and the bandwidth of the synthesized signal, this thesis also introduces the carry-save coding technique to improve the speed at the register transfer level in addition to the parallel structure at the system level. The structure and methodology proposed in this thesis have been validated by intensive simulations and experiments done on reconfigurable CPLD and FPGA boards. The experimental results presented in this thesis show that the system implemented by using the proposed methodology can successfully synthesize the linear modulated signals at t.
Now available in a three-volume set, this updated and expanded edition of the bestselling Digital Signal Processing Handbook continues to provide the engineering community with authoritative coverage of the fundamental and specialized aspects of information-bearing signals in digital form. Encompassing essential background material, technical details, standards, and software, The Digital Signal Processing Handbook, Second Edition reflects cutting-edge information on signal processing algorithms and protocols related to speech, audio, multimedia, and video processing technology associated with standards ranging from WiMax to MP3 audio, low-power/high-performance DSPs, color image processing, and chips on video. The three-volume set draws on the experience of leading engineers, researchers, and scholars and includes 29 new chapters that address multimedia and Internet technologies, tomography, radar systems, architecture, standards, and future applications in speech, acoustics, video, radar, and telecommunications. Each volume in the set is also available individually ... Emphasizing theoretical concepts, Digital Signal Processing Fundamentals (Catalog no. 46063) provides comprehensive coverage of the basic foundations of DSP. Coverage includes: Signals and Systems, Signal Representation and Quantization, Fourier Transforms, Digital Filtering, Statistical Signal Processing, Adaptive Filtering, Inverse Problems and Signal Reconstruction, and Time–Frequency and Multirate Signal Processing. Wireless, Networking, Radar, Sensor Array Processing, and Nonlinear Signal Processing (Catalog no. 46047) thoroughly covers the foundations of signal processing related to wireless, radar, space–time coding, and mobile communications together with associated applications to networking, storage, and communications. Video, Speech, and Audio Signal Processing and Associated Standards, (Catalog no. 4608X) details the basic foundations of speech, audio, image, and video processing and associated applications to broadcast, storage, search and retrieval, and communications.
The digital front-end (DFE) is the most critical stage in a wireless base-station. The DFE along with the analog to digital converter (ADC) is responsible for bridging the analog RF and IF processing on one side and the digital baseband processing on the other side. The most important reason for replacing analog with digital signal processing is the ability to softly reconfigure the channels in the base station RF in real time, thus allowing for the implementation of various signal conditioning, compensation and mitigation channel non-linear responses. Once tested, these algorithms can be implemented on a proprietary CMOS vector processor and commercial FPGA hardware platforms. In this thesis, we attempt to minimize the design efforts and lower the cost involved in the usage of analog electronics by using sophisticated digital signal processing (DSP) for restoring and enhancing the quality of the wireless channels. This thesis presents a versatile Digital Front-End architecture, which has been simulated using MATLAB/Simulink. The architecture includes the design of robust Digital Up-Conversion (DUC) blocks in the transmit downlink and Digital Down-Conversion (DDC) blocks present in the receiver uplink paths in a wireless base station RF. Crest factor reduction (CFR) schemes help reduce the Peak to Average Power Ratio (PAPR)of the signal entering the base-station and have been implemented widely for code division multiple access (CDMA) and Long Term Evolution (LTE) systems, this is important because if the signal with the high PAPR is allowed to pass through the power amplifier(PA) it will result in the amplifier operating in its nonlinear region creating non-linear distortions in amplitude and phase, and the only other way to avoid this is to back off the signal to the linear region of the amplifier thus reducing its efficiency. The selection and design of the DUC and DDC filters has been compared and optimized to match to the spectral mask requirements mentioned in the 3GPP standards. Crest factor reduction has also been studied in detail and a computationally efficient algorithm for meeting the desired PAPR in accordance with the 3GPP standards will be presented. By using the CFR algorithm, the PAPR of the LTE signal was reduced from 10.8 dB to 7 dB and from 10.5 dB to 8 dB for a WCDMA signal. Finally, we implement Digital Predistortion (DPD) which is a method by which one first stimulates a non-linear power amplifier (PA) with baseband samples and then observes the result of that stimulus at its output. Without this process we will need to use a power amplifier with a higher input power rating which needs to be backed off to operate in its linear region thus reducing the efficiency of the PA used and increasing its cost. The process involves the use of a digital predistorter which creates an expanding nonlinearity which when used in cascade with the PA nullifies the compressing nonlinear characteristics of the PA thus enabling its use in its linear region up to its saturating point. A Look-Up Table (LUT) type Adaptive Digital Pre-Distortion (ADPD) is presented; here we developed an algorithm where the output signal of the PA is used as a reference signal. This reference signal is then used to update the coefficients of the LUT, so that the non-linear responses of the PA will not the affect the amplified signals. In addition, we investigated methods such as the nonlinear auto-regressive moving average (NARMA) and the memory polynomial models. In the latter, the predistorter parameters are calculated

from the output signal obtained from the PA through the adaptive functions obtained using the memory polynomial. From these parameters, the predistorted signal is reconstructed and fed to the input of the PA. By using the DPD algorithm the nonlinear distortions of the PA came down by 60 dB when a WCDMA signal was used and by around 40 dB when LTE signal was used. As the PA is the heart of the base-station RF, we show that the main function of the DFE is to ensure a PA linearized output with a high efficiency.

This book discusses the fundamentals of RFID and the state-of-the-art research results in signal processing for RFID, including MIMO, blind source separation, anti-collision, localization, covert RFID and chipless RFID. Aimed at graduate students as well as academic and professional researchers/engineers in RFID technology, it enables readers to become conversant with the latest theory and applications of signal processing for RFID. Key Features: Provides a systematic and comprehensive insight into the application of modern signal processing techniques for RFID systems Discusses the operating principles, channel models of RFID, RFID protocols and analog/digital filter design for RFID Explores RFID-oriented modulation schemes and their performance Highlights research fields such as MIMO for RFID, blind signal processing for RFID, anti-collision of multiple RFID tags, localization with RFID, covert RFID and chipless RFID Contains tables, illustrations and design examples

This book describes the design of CMOS circuits for ultra-low power consumption including analog, radio frequency (RF), and digital signal processing circuits (DSP). The book addresses issues from circuit and system design to production design, and applies the ultra-low power circuits described to systems for digital hearing aids and capsule endoscope devices. Provides a valuable introduction to ultra-low power circuit design, aimed at practicing design engineers; Describes all key building blocks of ultra-low power circuits, from a systems perspective; Applies circuits and systems described to real product examples such as hearing aids and capsule endoscopes.

Aperture-arrays combined with multi-dimensional (MD) digital signal processing (DSP) techniques provide the capability of synthesizing arbitrarily steered beams enabling directional enhancement of radio frequency (RF) signals. This is particularly desirable in radar applications, where electronically-scanned beams are needed to obtain measurements over range, angle/direction, polarization and doppler domains. Digital aperture arrays employing two-dimensional (2-D) infinite impulse response (IIR) beam filters combined with radar systems can lead to rapidly steerable RF beams with modulated radar signatures which carry essential information for critical real time electromagnetic (EM) sensing requirements. This thesis introduces novel MD DSP algorithms for radar signal extraction that find applications in emerging RF technologies. Directional sensing and remote localization architectures comprising of 2-D IIR digital filters, continuous wave (CW) doppler radars and frequency modulated continuous wave (FMCW) radars are proposed along with simulation results. Detection and classification of miniaturized unmanned aerial systems (UAS) using doppler radar signatures is presented with experimental results. Radar signal extraction algorithms in applications of 3-D body tracking and remote vital sign detection is also reviewed through a system study on recent advances of radar systems.

Understand the RF and Digital Signal Processing Principles Driving Software-defined Radios! Software-defined radio (SDR) technology is a configurable, low cost, and power efficient solution for multimode and multistandard wireless designs. This book describes software-defined radio concepts and design principles from the perspective of RF and digital signal processing as performed within this system. After an introductory overview of essential SDR concepts, this book examines signal modulation techniques, RF and digital system analysis and requirements, Nyquist and oversampled data conversion techniques, and multirate digital signal processing.. KEY TOPICS •Modulation techniques Master analog and digital modulation schemes •RF system-design parameters Examine noise and link budget analysis and Non-linear signal analysis and design methodology •Essentials of baseband and bandpass sampling and gain control IF sampling architecture compared to traditional quadrature sampling, Nyquist zones, automatic gain control, and filtering •Nyquist sampling converter architectures Analysis and design of various Nyquist data converters •Oversampled data converter architectures Analysis and design of continuous-time and discrete-time Delta-Sigma converters •Multirate signal processing Gain knowledge of interpolation, decimation, and fractional data rate conversion *Offers readers a powerful set of analytical and design tools *Details real world designs *Comprehensive coverage makes this a must have in the RF/Wireless industry

Theory, Analysis and Design of RF Interferometric Sensors presents the theory, analysis and design of RF interferometric sensors. RF interferometric sensors are attractive for various sensing applications that require every fine resolution and accuracy as well as fast speed. The book also presents two millimeter-wave interferometric sensors realized using RF integrated circuits. The developed millimeter-wave homodyne sensor shows sub-millimeter resolution in the order of 0.05 mm without correction for the non-linear phase response of the sensor's quadrature mixer. The designed millimeter-wave double-channel homodyne sensor provides a resolution of only 0.01 mm, or 1/840th of the operating wavelength, and can inherently suppress the non-linearity of the sensor's quadrature mixer. The experimental results of displacement and velocity measurement are presented as a way to demonstrate the sensing ability of the RF interferometry and to illustrate its many possible applications in sensing. The book is succinct, yet the material is very much self-contained, enabling readers with an undergraduate background in electrical engineering or physics with some experiences or graduate courses in RF circuits to understand easily.

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This is one of the first books on the emerging research topic of digital compensation of RF imperfections. The book presents a new multidisciplinary vision on the design of wireless communication systems. In this approach the imperfections of the RF front-ends are accepted and digital signal processing algorithms are designed to suppress their impact on system performance. The book focuses on multiple-antenna orthogonal frequency division multiplexing (MIMO OFDM).

Addresses a wide selection of multimedia applications, programmable and custom architectures for the implementations of multimedia systems, and arithmetic architectures and design methodologies. The book covers recent applications of digital signal processing algorithms in multimedia, presents high-speed and low-priority binary and finite field arithmetic architectures, details VHDL-based implementation approaches, and more.

Master the basic concepts and methodologies of digital signal processing with this systematic introduction, without the need for an extensive mathematical background. The authors lead the reader through the fundamental mathematical principles underlying the operation of key signal processing techniques, providing simple arguments and cases rather than detailed general proofs. Coverage of practical implementation, discussion of the limitations of particular methods and plentiful MATLAB illustrations allow readers to better connect theory and practice. A focus on algorithms that are of theoretical importance or useful in real-world applications ensures that students cover material relevant to engineering practice, and equips students and practitioners alike with the basic principles necessary to apply DSP techniques to a variety of applications. Chapters include worked examples, problems and computer experiments, helping students to absorb the material they have just read. Lecture slides for all figures and solutions to the numerous problems are available to instructors.

PSpice is a software package that provides robust, advanced circuit analysis tools to improve design performance, yield, and reliability. Its capabilities enable engineers to create virtual prototypes of designs and maximize circuit performance automatically. This book is the fifth of a five-part series of books covering PSpice 10.5 and all of its applications. This book examines linear time invariant systems starting with the difference equation and applying the z-transform to produce a range of filter type i.e. low-pass, high-pass, and bandpass. Convolution is examined, followed by digital oscillators, including quadrature carrier generation, are then examined. Several filter design methods are considered and include the bilinear transform, impulse invariant, and window techniques. A range of DSP applications are then considered and include the Hilbert transform, single sideband modulator using the Hilbert transform and quad oscillators, integrators and differentiators. Decimation and interpolation are simulated to demonstrate the usefulness of the multi-sampling environment. Decimation is also applied in a treatment on digital receivers. Lastly, we look at some musical applications for DSP such as reverberation/echo using real-world signals imported into PSpice using the program Wav2Ascii. The zero-forcing equalizer is dealt with in a simplistic manner and illustrates the effectiveness of equalizing signals in a receiver after transmission.Other books in the series:PSpice for Circuit Theory and Electronic Devices (9781598291568)PSpice for Filters and Transmission Lines (9781598291582)PSpice for Analog Communications Engineering (9781598291605)PSpice for Digital Communications Engineering (9781598291629)

This book describes medical imaging systems, such as X-ray, Computed tomography, MRI, etc. from the point of view of digital signal processing. Readers will see techniques applied to medical imaging such as Radon transformation, image reconstruction, image rendering, image enhancement and restoration, and more. This book also outlines the physics behind medical imaging required to understand the techniques being described. The presentation is designed to be accessible to beginners who are doing research in DSP for medical imaging. Matlab programs and illustrations are used wherever possible to reinforce the concepts being discussed.

This guide to radio engineering covers every technique DSP and RF engineers need to build software radios for a wide variety of wireless systems using DSP techniques. Included are practical guidelines for choosing DSP microprocessors, and systematic, object-oriented software design techniques.