Digital Signal Processing First Lab Solutions

1.1 Overview of Lab-on-Chip Laboratory-on-Chip (LoC) is a multidisciplinary approach used for the miniaturization, integration and automation of biological assays or procedures in analytical chemistry [1–3]. Biology and chemistry are experimental sciences that are continuing to evolve and develop new protocols. Each protocol offers step-by-step laboratory instructions, lists of the necessary equipments and required biological and/or chemical substances [4–7]. A biological or chemical laboratory contains various pieces of equipment used for performing such protocols and, as shown in Fig. 1.1, the engineering aspect of LoC design is aiming to embed all these components in a single chip for single-purpose applications. 1.1.1 Main Objectives of LoC Systems Several clear advantages of this technology over conventional approaches, including portability, full automation, ease of operation, low sample consumption and fast assays time, make LoC suitable for many applications including. 1.1.1.1 Highly Throughput Screening To conduct an experiment, a researcher fills a well with the required biological or chemical analytes and keeps the sample in an incubator for some time to allowing the sample to react properly. Afterwards, any changes can be observed using a microscope. In order to quickly conduct millions of biochemical or pharmacolo- cal tests, the researchers will require an automated highly throughput screening (HTS) [8], comprised of a large array of wells, liquid handling devices (e.g., mic-channel, micropump and microvalves [9–11]), a fully controllable incubator and an integrated sensor array, along with the appropriate readout system.

Designed for senior electrical engineering students, this textbook explores the theoretical concepts of digital signal processing and communication systems by presenting laboratory experiments using real-time DSP hardware. The experiments are designed for the Texas Instruments TMS320C6701 Evaluation Module or TMS320C6711 DSK but can easily be adapted to other DSP boards. Each chapter begins with a presentation of the required theory and concludes with instructions for performing experiments to implement the theory. In the process of performing the experiments, students gain experience in working with software tools and equipment commonly used in industry.

Mneney's text focuses on basic concepts of digital signal processing, MATLAB simulation, and implementation on selected DSP hardware.

Signal Processing for Neuroscientists introduces analysis techniques primarily aimed at neuroscientists and biomedical engineering students with a reasonable but modest background in mathematics, physics, and computer programming. The focus of this text is on what can be considered the ‘golden trio’ in the signal processing field: averaging, Fourier analysis, and filtering. Techniques such as convolution, correlation, coherence, and wavelet analysis are considered in the context of time and frequency domain analysis. The whole spectrum of signal analysis is covered, ranging from data acquisition to data processing; and from the mathematical background of the analysis to the practical application of processing algorithms. Overall, the approach to the mathematics is informal with a focus on basic understanding of the methods and their interrelationships rather than detailed proofs or derivations. One of the principle goals is to provide the reader with the background required to understand the principles of commercially available analyses software, and to allow him/her to construct his/her own analysis tools in an environment such as MATLAB®. Multiple color illustrations are integrated in the text Includes an introduction to biomedical signals, noise characteristics, and recording techniques Basics and background for more advanced topics can be found in extensive notes and appendices A Companion Website hosts the MATLAB scripts and several data files:
http://www.elsevierdirect.com/companion.jsp?ISBN=9780123708670

Real-time or applied digital signal processing courses are offered as follow-ups to conventional or theory-oriented digital signal processing courses in many engineering programs for the purpose of teaching students the technical know-how for putting signal processing algorithms or theory into practical use. These courses normally involve access to a teaching laboratory that is equipped with hardware boards, in particular DSP boards, together with their supporting software. A number of textbooks have been written discussing how to achieve real-time implementation on these hardware boards. This book discusses how to use smartphones as hardware boards for real-time implementation of signal processing algorithms as an alternative to the hardware boards that are used in signal processing laboratory courses. The fact that mobile devices, in particular smartphones, have become powerful processing platforms led to the development of this book enabling students to use their own smartphones to run signal processing algorithms in real-time considering that these days nearly all students possess smartphones. Changing the hardware platforms that are currently used in applied or real-time signal processing courses to smartphones creates a truly mobile laboratory experience or environment for students. In addition, it relieves the cost burden associated with using dedicated signal processing boards noting that the software development tools for smartphones are free of charge and are well-maintained by smartphone manufacturers. This book is written in such a way that it can be used as a textbook for real-time or applied digital signal processing courses offered at many universities. Ten lab experiments that are commonly encountered in such courses are covered in the book. This book is written primarily for those who are already familiar with signal processing concepts and are interested in their real-time and practical aspects. Similar to existing real-time courses, knowledge of C programming is assumed. This book can also be used as a self-study guide for those who wish to become familiar with signal processing app development on either Android or iPhone smartphones.

Concisely covers all the important concepts in an easy-to-understand way Gaining a strong sense of signals and systems fundamentals is key for general proficiency in any electronic engineering discipline, and critical for specialists in signal processing, communication, and control. At the same time, there is a pressing need to gain mastery of these concepts quickly, and in a manner that will be immediately applicable in the real word. Simultaneous study of both continuous and discrete signals and systems presents a much easy path to understanding signals and systems analysis. In A Practical

Approach to Signals and Systems, Sundararajan details the discrete version first followed by the corresponding continuous version for each topic, as discrete signals and systems are more often used in practice and their concepts are relatively easier to understand. In addition to examples of typical applications of analysis methods, the author gives comprehensive coverage of transform methods, emphasizing practical methods of analysis and physical interpretations of concepts. Gives equal emphasis to theory and practice Presents methods that can be immediately applied Complete treatment of transform methods Expanded coverage of Fourier analysis Self-contained: starts from the basics and discusses applications Visual aids and examples makes the subject easier to understand End-of-chapter exercises, with a extensive solutions manual for instructors MATLAB software for readers to download and practice on their own Presentation slides with book figures and slides with lecture notes A Practical Approach to Signals and Systems is an excellent resource for the electrical engineering student or professional to quickly gain an understanding of signal analysis concepts - concepts which all electrical engineers will eventually encounter no matter what their specialization. For aspiring engineers in signal processing, communication, and control, the topics presented will form a sound foundation to their future study, while allowing them to quickly move on to more advanced topics in the area. Scientists in chemical, mechanical, and biomedical areas will also benefit from this book, as increasing overlap with electrical engineering solutions and applications will require a working understanding of signals. Compact and self contained, A Practical Approach to Signals and Systems be used for courses or self-study, or as a reference book.

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 updated edition gives readers hands-on experience in real-time DSP using a practical, step-by-step framework that also incorporates demonstrations, exercises, and problems, coupled with brief overviews of applicable theory and MATLAB applications. Organized in three sections that cover enduring fundamentals and present practical projects and invaluable appendices, this new edition provides support for the most recent and powerful of the inexpensive DSP development boards currently available from Texas Instruments: the OMAP-L138 LCDK. It includes two new real-time DSP projects, as well as three new appendices: an introduction to the Code Generation tools available with MATLAB, a guide on how to turn the LCDK into a portable battery-operated device, and a comparison of the three DSP boards directly supported by this edition.

In this supplementary text, 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. Important Notice: Media content referenced within the product description or the product text may not be available in the ebook version.

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.

For introductory courses (freshman and sophomore courses) in Digital Signal Processing and Signals and Systems. Text may be used before the student has taken a course in circuits. DSP First and its accompanying digital assets are the result of more than 20 years of work that originated from, and was guided by, the premise that signal processing is the best starting point for the study of electrical and computer engineering. The "DSP First" approach introduces the use of mathematics as the language for thinking about engineering problems, lays the groundwork for subsequent courses, and gives students hands-on experiences with MATLAB. The 2nd Edition features three new chapters on the Fourier Series, Discrete-Time Fourier Transform, and the The Discrete Fourier Transform as well as updated labs, visual demos, an update to the existing chapters, and hundreds of new homework problems and solutions. The full text downloaded to your computer With eBooks you can: search for key concepts, words and phrases make highlights and notes as you study share your notes with friends eBooks are downloaded to your computer and accessible either offline through the Bookshelf (available as a free download), available online and also via the iPad and Android apps. Upon purchase, you will receive via email the code and instructions on how to access this product. Time limit The eBooks products do not have an expiry date. You will continue to access your digital ebook products whilst you have your Bookshelf installed.

This textbook provides an introduction to the study of digital signal processing, employing a top-to-bottom structure to motivate the reader, a graphical approach to the solution of the signal processing mathematics, and extensive use of MATLAB. In contrast to the conventional teaching approach, the book offers a top-down approach which first introduces students to digital filter design, provoking questions about the mathematical tools required. The following chapters provide answers to these questions, introducing signals in the discrete domain, Fourier analysis, filters in the time domain and the Z-transform. The author introduces the mathematics in a conceptual manner with figures to illustrate the physical meaning of the equations involved. Chapter six builds on these concepts and discusses advanced filter design, and chapter seven discusses matters of practical implementation. This book introduces the corresponding MATLAB functions and programs in every chapter with examples, and the final chapter introduces the actual real-time filter from MATLAB. Aimed primarily at undergraduate students in electrical and electronic engineering, this book enables the reader to implement a digital filter using MATLAB.

Field Programmable Gate Arrays (FPGAs) are increasingly becoming the platform of choice to implement DSP algorithms. This book is designed to allow DSP students or DSP engineers to achieve FPGA implementation of DSP algorithms in a one-semester DSP laboratory course or in a short design cycle time based on the LabVIEW FPGA Module. Features: - The first DSP laboratory book that uses the FPGA platform instead of the DSP platform for implementation of DSP algorithms - Incorporating introductions to LabVIEW and VHDL - Lab experiments covering FPGA implementation of basic DSP topics including convolution, digital filtering, fixed-point data representation, adaptive filtering, frequency domain processing - Hardware FPGA implementation applications including wavelet transform, software-defined radio, and MP3 player - Website providing downloadable LabVIEW FPGA codes

LabVIEW (Laboratory Virtual Instrumentation Engineering Workbench) developed by National Instruments is a graphical programming environment. Its ease of use allows engineers and students to streamline the creation of code visually, leaving time traditionally spent on debugging for true comprehension of DSP. This book is perfect for practicing engineers, as well as hardware and software technical managers who are familiar with DSP and are involved in system-level design. With this text, authors Kehtarnavaz and Kim have also provided a valuable resource for students in conventional engineering courses. The integrated lab exercises create an interactive experience which supports development of the hands-on skills essential for learning to navigate the LabVIEW program. Digital Signal Processing System-Level Design Using LabVIEW is a comprehensive tool that will greatly accelerate the DSP learning process. Its thorough examination of LabVIEW leaves no question unanswered. LabVIEW is the program that will demystify DSP and this is the book that will show you how to master it. * A graphical programming approach (LabVIEW) to DSP system-level design * DSP implementation of appropriate components of a LabVIEW designed system * Providing system-level, hands-on experiments for DSP lab or project courses

A mathematically rigorous but accessible treatment of digital signal processing that intertwines basic theoretical techniques with hands-on laboratory instruction is provided by this book. The book covers various aspects of the digital signal processing (DSP) "problem". It begins with the analysis of discrete-time signals and explains sampling and the use of the discrete and fast Fourier transforms. The second part of the book — covering digital to analog and analog to digital conversion — provides a practical interlude in the mathematical content before Part III lays out a careful development of the Z-transform and the design and analysis of digital filters.

The TMS320C6x is Texas Instrument's next generation DSP found in over 60 percent of wireless devices from leading manufacturers such as Ericsson, Nokia, Sony, and Handspring Author has many years experience working with the TI line of TMS DSPs and his books are based on courses and seminars given at TI sponsored meetings All programs listed in the text will be available on the Wiley FTP site In addition to its wireless applications, the TMS DSP is tailored to enable a new generation of Internet media entertainment appliances

Digital Signal Processing System Design combines textual and graphical programming to form a hybrid programming approach, enabling a more effective means of building and analyzing DSP systems. The hybrid programming approach allows the use of previously developed textual programming solutions to be integrated into LabVIEW's highly interactive and visual environment, providing an easier and quicker method for building DSP systems. This book is an ideal introduction for engineers and students seeking to develop DSP systems in quick time. Features: The only DSP laboratory book that combines textual and graphical programming 12 lab experiments that incorporate C/MATLAB code blocks into the LabVIEW graphical programming environment via the MathScripting feature Lab experiments covering basic DSP implementation topics including sampling, digital filtering, fixed-point data representation, frequency domain processing Interesting applications using the hybrid programming approach, such as a software-defined radio system, a 4-QAM Modem, and a cochlear implant simulator The only DSP project book that combines textual and graphical programming 12 Lab projects that incorporate MATLAB code blocks into the LabVIEW graphical programming environment via the MathScripting feature Interesting applications such as the design of a cochlear implant simulator and a software-defined radio system

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.

This book provides the know-how for the implementation and optimization of computationally intensive signal processing algorithms on the Texas Instruments family of TMS320C6000 digital signal processors.

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.

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.

For introductory courses (freshman and sophomore courses) in Digital Signal Processing and Signals and Systems. Text may be used before the student has taken a course in circuits. DSP First and it's accompanying digital assets are the result of more than 20 years of work that originated from, and was guided by, the premise that signal processing is the best starting point for the study of electrical and computer engineering. The "DSP First" approach introduces the use of mathematics as the language for thinking about engineering problems, lays the groundwork for subsequent courses, and gives students hands-on experiences with MATLAB. The Second Edition features three new chapters on the Fourier Series, Discrete-Time Fourier Transform, and the The Discrete Fourier Transform as well as updated labs, visual demos, an update to the existing chapters, and hundreds of new homework problems and solutions.

Now in a new edition—the most comprehensive, hands-on introduction to digital signal processing The first edition of Digital Signal Processing and Applications with the TMS320C6713 and TMS320C6416 DSK is widely accepted as the most extensive text available on the hands-on teaching of Digital Signal Processing (DSP). Now, it has been fully updated in this valuable Second Edition to be compatible with the latest version (3.1) of Texas Instruments Code Composer Studio (CCS) development environment. Maintaining the original's comprehensive, hands-on approach that has made it an instructor's favorite, this new edition also features: Added program examples that illustrate DSP concepts in real-time and in the laboratory Expanded coverage of analog input and output New material on frame-based processing A revised chapter on IIR, which includes a number of floating-point example programs that explore IIR filters more comprehensively More extensive coverage of DSP/BIOS All programs listed in the text—plus additional applications—which are available on a companion CD-ROM No other book provides such an extensive or comprehensive set of program examples to aid instructors in teaching DSP in a laboratory using audio frequency signals—making this an ideal text for DSP courses at the senior undergraduate and postgraduate levels. It also serves as a valuable resource for researchers, DSP developers, business managers, and technology solution providers who are looking for an overview and examples of DSP algorithms implemented using the TMS320C6713 and TMS320C6416 DSK.

Field Programmable Gate Arrays (FPGAs) are on the verge of revolutionising digital signal processing. Novel FPGA families are increasingly replacing ASICs and PDSPs for front-end digital signal processing algorithms. The efficient implementation of these algorithms is the main goal of this book. It starts with an overview of todays FPGA technology, devices and tools for designing DSP systems. A case study in the first chapter is the basis for more than 30 design examples. The following chapters deal with topics such as computer arithmetic concepts and the theory and the implementation of FIR and IIR filters. The VERILOG source code and a glossary are contained in the appendices. The accompanying CD-ROM contains examples in VHDL and Verilog code as well as the newest Altera 'Baseline' software.

"Digital Signal Processing: A Computer-Based Approach" is intended for a two-semester course on digital signal processing for seniors or first-year graduate students. Based on user feedback, a number of new topics have been added to the second edition, while some excess topics from the first edition have been removed. The author has taken great care to organize the chapters more logically by reordering the sections within chapters. More worked-out examples have also been included. The book contains more than 500 problems and 150 MATLAB exercises. New topics in the second edition include: finite-dimensional discrete-time systems, correlation of signals, inverse systems, system identification, matched filter, design of analog and IIR digital highpass, bandpass and bandstop filters, more on FIR filters, spectral analysis of random signals and sparse antenna array design. A corrected version of the main text is now packaged with Digital Signal Processing Laboratory Using MATLAB, which is intended for a computer-based DSP laboratory course that supplements a lecture course on Digital Signal Processing. The lab book includes 11 laboratory exercises, with each exercise containing a number of projects to be carried out on a computer. The book assumes that the reader has no background in MATLAB and teaches the reader, through tested programs in the first half of the book, the basics of this powerful language in solving important problems in signal processing. In the second half of the book, the student is asked to write the necessary MATLAB programs to carry out the projects.

Discrete-Time Signal Processing covers the information that the electrical computing and engineering student needs to know about DSP.

Digital Signal Processing has undergone enormous growth in usage/implementation in the last 20 years and many engineering schools are now offering real-time DSP courses in their undergraduate curricula. Our everyday lives involve the use of DSP systems in things such as cell phones and high-speed modems; Texas Instruments has introduced the TMS320C6000 DSP processor family to meet the high performance demands of today's signal processing applications. This book provides the know-how for the implementation and optimization of computationally intensive signal processing algorithms on the Texas Instruments family of TMS320C6000 DSP processors. It is organized in such a way that it can be used as the textbook for DSP lab courses offered at many engineering schools or as a self-study/reference for those familiar with DSP but not this family of processors. This book provides a restructured, modified, and condensed version

of the information in more than twenty TI manuals so that one can learn real-time DSP implementations on the C6000 family in a structured course, within one semester. Each chapter is followed by an appropriate lab exercise to provide the hands-on lab material for implementing appropriate signal processing functions. Each chapter is followed by an appropriate lab exercise Provides the hands-on lab material for implementing appropriate signal processing functions
DIGITAL SIGNAL PROCESSING LABORATORY USING MATLAB is intended for a computer-based DSP laboratory course that supplements a lecture course on Digital Signal Processing. The book can be used either as a stand-alone text or in conjunction with Mitra's Digital Signal Processing: A Computer-Based Approach. The book includes 11 laboratory exercises, with each exercise containing a number of projects to be carried out on a computer. The book assumes that the reader has no background in MATLAB and teaches the reader, through tested programs in the first half of the book, the basics of this powerful language in solving important problems in signal processing. In the second half of the book, the student is asked to write the necessary MATLAB programs to carry out the projects.

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.

This book/lab manual allows readers to actually "implement" and optimize computationally intensive signal processing algorithms and examine their performance on the TMS320C6x DSP platform. Information from the TI reference manuals for the TMS3206x has been restructured, condensed, and modified for self-study, and seven lab exercises take readers through the entire process of C6x code writing and optimization. Requires knowledge of C programming. TMS320C6x Architecture; Software Tools (with lab on Code Composer Studio Tutorial); Sampling (with lab on Audio Signal Sampling); Fixed-Point vs. Floating-Point (with lab on Q-Format and Overflow); Code Optimization (with lab on Real-Time Filtering); Frame Processing (with lab on Fast Fourier Transform); Circular Buffering (with lab on Adaptive Filtering); Application Examples. For those who are already familiar with DSP concepts and are interested in real-time and efficient algorithm implementation on the TMS320C6x.

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??.

This textbook introduces readers to digital signal processing fundamentals using Arm Cortex-M based microcontrollers as demonstrator platforms. It covers foundational concepts, principles and techniques such as signals and systems, sampling, reconstruction and anti-aliasing, FIR and IIR filter design, transforms, and adaptive signal processing.

Digital Signal Processing: A Computer-Based Approach is intended for a two-semester course on digital signal processing for seniors or first-year graduate students. Based on user feedback, a number of new topics have been added to the third edition, while some excess topics from the second edition have been removed. The author has taken great care to organize the chapters more logically by reordering the sections within chapters. More worked-out examples have also been included. The book contains more than 500 problems and 150 MATLAB exercises. New topics in the third edition include: short-time characterization of discrete-time signals, expanded coverage of discrete-time Fourier transform and discrete Fourier transform, prime factor algorithm for DFT computation, sliding DFT, zoom FFT, chirp Fourier transform, expanded coverage of z-transform, group delay equalization of IIR digital filters, design of computationally efficient FIR digital filters, semi-symbolic analysis of digital filter structures, spline interpolation, spectral factorization, discrete wavelet transform.