Book Review
Digital Signal Processing
by Li Tan
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With the fast growing technology development from the past decade, the industry has high demands of qualified technologist to fit for their new requirement. At the mean time, universities are also updating their curriculum to catch up with this change. Some courses traditionally belong to the engineering territory, such as digital signal processing (DSP), are now emerging in the technology degree plan. But the textbook development is far behind to fit in this revolution. It is realized by both authors of this paper the difficulty of teaching technology students DSP using popular textbooks chosen by engineering program. Technology students have their own features and different expectation in their future job position, so to find a textbook suitable for this group of students is a great task for the instructors of similar situation. As future technologists, engineering technology students are more hands-on and less theory-based comparing to other engineering students. Meeting both requirements, Dr. Tan's book is perfectly suitable for being a textbook for technology students. By learning the fundamental principles of the topics covered in this book, students will gain the DSP knowledge and apply them in their future career.

In the book by Li Tan, we find the combination of fundamentals and applications associate with digital signal processing. It summarizes the necessary mathematics background, and concentrates on the application part of DSP. The book is presented in 816 pages with 409 figures, 174 examples, 252 exercises, and 432 equations. A solution manual and many power-points slides are provided for instructor's convenience.

The book contains 13 chapters and 6 appendices. The first eight chapters cover the fundamental DSP concepts and theories, from signal sampling, digital signal systems, to basic transforms and filter design. These eight chapters are suitable for readers with no or little DSP experiences. Chapters 9 to 13 talk about DSP hardware and applications, including digital signal processor, adaptive filters and applications, waveform quantization and compression, and image processing. These chapters are appropriate for readers who are equipped with fundamental DSP knowledge. A summary of each chapter is presented next.

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Chapter 1–Introduction to Digital Signal Processing—introduces the basic concepts of DSP and presents a general DSP block diagram. Some basic DSP examples, such as digital filtering and signal spectrum analysis, are shown in block diagrams. Typical DSP real-world applications are also introduced. *Chapter 2–Signal Sampling and Quantization*—talks about the sampling theorem in both time domain and frequency domain. Signal reconstruction is also covered in this chapter. Practical considerations for signal sampling are presented. Finally, concepts of analog-to-digital conversion (ADC), digital-to-analog conversion (DAC), and quantization, are included. *Chapter 3–Digital Signals and Systems*—introduces notations for digital signals and common digital sequences. Linear time-invariant, causal systems are studied and the properties of such systems are presented. Then, difference equations and impulse responses are talked about for representation of digital systems using their impulse responses. The concept of constructing stable systems is also covered by introducing the bounded-in-and-bounded-out stability. The chapter ends with a section works with digital convolution, which plays an important role in digital filtering. *Chapter 4–Discrete Fourier Transform and Signal Spectrum*—investigates the discrete Fourier transform (DFT) and its properties. The most common DFT application of transformation of a finite-length digital signal into the spectrum in frequency domain is covered in the "Amplitude Spectrum and Power Spectrum" section. Then the chapter talks about how to reduce spectral leakage using window functions. Example application on speech signal spectral estimation is presented, followed by the introduction of fast Fourier Transform (FFT).

Chapter 5–The z-Transform—investigates z-transform and its properties. As a very important tool in describing and analyzing digital systems, z-transform possesses several important properties, such as linearity and convolution, which are studied in this chapter. Several inverse z-transform methods are also introduced. The chapter ends with the application of z-transform on solving difference equations. *Chapter 6–Digital Signal Processing Systems, Basic Filtering Types, and Digital Filter Realizations*—describes DSP systems with filtering concepts starting with the difference equation and digital filtering, then derives transfer function from difference equation. The chapter also analyzes DSP systems' stability using z-plane pole-zero plot and digital filter frequency response. Basic types of filtering are then described, followed by the discussion of basic realization methods for digital filters. Finally, the chapter presents digital filtering applications on speech enhancement. *Chapter 7–Finite Impulse Response Filter Design*—first introduces the finite impulse response (FIR) filter format. Then the chapter investigates some FIR filter design methods. The chapter also presents applications of FIR filters on noise reduction and digital crossover. *Chapter 8–Infinite Impulse Response Filter Design*—introduces the infinite impulse response (IIR) filter format, investigates bilinear transformation method for IIR filter design, then describes the procedures for design digital Butterworth and Chebyshev filters. The chapter also investigates other IIR filter design methods. Finally, the chapter presents IIR filter application on solving real-world problems.

Chapter 9–Hardware and Software for Digital Signal Processors—introduces digital signal processor architecture and hardware units, including multiplier and accumulator, shifters, and address generators. The chapter also investigates fixed-point and floating-point data formats. Then this chapter presents the implementations of FIR and IIR filters in fixed-point systems, as well as programming examples of DSP. *Chapter 10–Adaptive Filters and Applications*—introduces basics of adaptive FIR filters and basic wiener filter theory and least mean square algorithm. Then the chapter presents adaptive filter applications on noise cancellation, and etc. *Chapter 11–Waveform Quantization and Compression*—investigates speech and audio quantization and compression techniques. The chapter then introduces linear midtread quantization, followed by the description on speech and audio compression algorithms. Finally, the discrete cosine transform (DCT) and modified DCT, which is adopted in MP3, are studied. *Chapter 12–Multirate Digital Signal Processing, Oversampling of Analog-to-Digital Conversion, and Undersampling of Bandpass Signals*—studies basic concepts of multirate DSP. Polyphase filter structure and implementation are also investigated. Oversampling of ADC, undersampling of bandpass signals techniques, and their applications are then investigated. *Chapter 13–Image Processing Basics*—presents the most import DSP application, image and video processing. Digital images and their processing techniques are studied in this chapter, by first looking at the image processing notation and image data formats. Image filtering enhancement is also covered in this chapter. Image spectra are analyzed using DFT. JPEG color image compression with DCT are described. Finally, the chapter describes basic video signal processing techniques.

The book contains a total of six appendices labeled as A to F. They serve as valuable resources for students to obtain more hands-on experiments in order to have a better understanding of the contents of the book. Appendix A introduces the MATLAB environment. Appendix B provides functions and illustrates fundamental analog signal processing with detailed step by step examples. In appendices C, D, and E, Butterworth and Chebyshev filters, Sinusoidal steady-state response in digital filters, and FIR filter design equations are presented respectively. Appendix F reviews some useful mathematical formulas.

A very convenient feature of this book is the on-line support which can be found at http://books.elsevier.com/companions/9780123740908. The lab manuals, plus real-time C programs can be found from this link. Qualified instructors can also request all the solutions, and PowerPoint slides from the publisher and the author.

In summary, this book suffices the need of teaching technology students digital signal processing related concepts and practice. The step by step approach to the theories and plenty of examples to illustrate the applications assist students to acquire sufficient knowledge in the area of DSP. MATLAB simulation is used throughout the book with code provided. In the second half of the book, starting with chapter 9, varieties of DSP applications are introduced which include both hardware and software for digital signal processor from the current leading manufactures, image, audio and video processing under according standards. All those lit students up with up-to-date technology and strengthened the hands-on feature of this book.

With more than five years of teaching experience in Computer Engineering Technology program for both authors, we are confident that this book will benefit technology students in the learning of DSP.