

PREFACE: DIGITAL SIGNAL PROCESSING AND DIGITAL FILTER DESIGN*

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Digital signal processing (DSP) has existed as long as quantitative calculations have been systematically applied to data in Science, Social Science, and Technology. The set of activities started out as a collection of ideas and techniques in very different applications. Around 1965, when the fast Fourier transform (FFT) was rediscovered, DSP was extracted from its applications and became a single academic and professional discipline to be developed as far as possible.

One of the earliest books on DSP was by Gold and Rader [5], written in 1968, although there had been earlier books on sampled data control and time series analysis, and chapters in books on computer applications. In the late 60's and early 70's there was an explosion of activity in both the theory and application of DSP. As the area was beginning to mature, two very important books on DSP were published in 1975, one by Oppenheim and Schaffer [8] and the other by Rabiner and Gold [11]. These three books dominated the early courses in universities and self study in industry.

The early applications of DSP were in the defense, oil, and medical industries. They were the ones who needed and could afford the expensive but higher quality processing that digital techniques offered over analog signal processing. However, as the theory developed more efficient algorithms, as computers became more powerful and cheaper, and finally, as DSP chips became commodity items (e.g. the Texas Instruments TMS-320 series) DSP moved into a variety of commercial applications and the current digitization of communications began. The applications are now everywhere. They are tele-communications, seismic signal processing, radar and sonar signal processing, speech and music signal processing, image and picture processing, entertainment signal processing, financial data signal processing, medical signal processing, non-destructive testing, factory floor monitoring, simulation, visualization, virtual reality, robotics, and control. DSP chips are found in virtually all cell phones, digital cameras, high-end stereo systems, MP3 players, DVD players, cars, toys, the "Segway", and many other digital systems.

In a modern curriculum, DSP has moved from a specialized graduate course down to a general undergraduate course, and, in some cases, to the introductory freshman or sophomore EE course [6]. An exciting project is experimenting with teaching DSP in high schools and in colleges to non-technical majors [9].

Our reason for writing this book and adding to the already long list of DSP books is to cover the new results in digital filter design that have become available in the last 10 to 20 years and to make these results available on line in Connexions as well as print. Digital filters are important parts of a large number of systems and processes. In many cases, the use of modern optimal design methods allows the use of a less expensive DSP chip for a particular application or obtaining higher performance with existing hardware. The book should be useful in an introductory course if the students have had a course on discrete-time systems. It can be used in a second DSP course on filter design or used for self-study or reference in industry.

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We first cover the optimal design of Finite Impulse Response (FIR) filters using a least squared error, a maximally flat, and a Chebyshev criterion. A feature of the book is covering finite impulse response (FIR) filter design before infinite impulse response (IIR) filter design. This reflects modern practice and new filter design algorithms. The FIR filter design chapter contains new methods on constrained optimization, mixed optimization criteria, and modifications to the basic Parks-McClellan algorithm that are very useful. Design programs are given in MatLab and FORTRAN.

A brief chapter on structures and implementation presents block processing for both FIR and IIR filters, distributed arithmetic structures for multiplierless implementation, and multirate systems for filter banks and wavelets. This is presented as a generalization to sampling and to periodically time-varying systems. The bifrequency map gives a clearer explanation of aliasing and how to control it.

The basic notes that were developed into this book have evolved over 35 years of teaching and conducting research in DSP at Rice, Erlangen, and MIT. They contain the results of research on filters and algorithms done at those universities and other universities and industries around the world. The book tries to give not only the different methods and approaches, but also reasons and intuition for choosing one method over another. It should be interesting to both the university student and the industrial practitioner.

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We particularly thank Texas Instruments and Prentice Hall for returning the copyrights to me so that part of the material in **DFT/FFT and Convolution Algorithms**[4], **Design of Digital Filters**[10], and "Efficient Fourier Transform and Convolution Algorithms" in **Advanced Topics in Signal Processing**[2] could be included here under the Creative Commons Attribution copyright. I also appreciate IEEE policy that allows parts of my papers to be included here.

A rather long list of references is included to point to more background, to more advanced theory, and to applications. A book of Matlab DSP exercises that could be used with this book has been published through Prentice Hall [3], [7]. Some Matlab programs are included to aid in understanding the design algorithms and to actually design filters. LabView from National Instruments is a very useful tool to both learn with and use in application. All of the material in these notes is being put into "Connexions" [1] which is a modern web-based open-content information system www.cnx.org. Further information is available on our web site at www.dsp.rice.edu with links to other related work. We thank Richard Baraniuk, Don Johnson, Ray Wagner, Daniel Williamson, and Marcia Horton for their help.

This version of the book is a draft and will continue to evolve under Connexions. A companion FFT book is being written and is also available in Connexions and print form. All of these two books are in the repository of Connexions and, therefore, available to anyone free to use, reuse, modify, etc. as long as attribution is given.

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