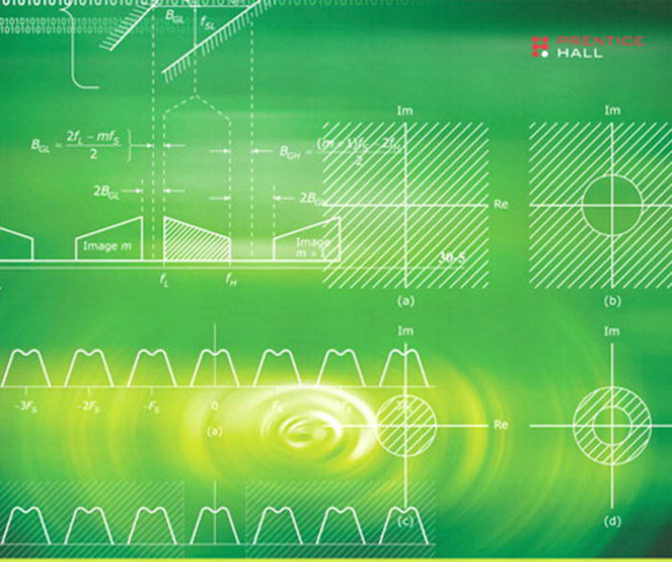

NOTES ON DIGITAL SIGNAL PROCESSING

Practical Recipes for Design, Analysis, and Implementation

C. BRITTON RORABAUGH

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Library of Congress Cataloging-in-Publication Data

Rorabaugh, C. Britton.
Notes on digital signal processing : practical recipes for design, analysis, and implementation / C. Britton Rorabaugh.
p. cm.
Includes bibliographical references and index.
ISBN 0-13-158334-4 (hardcover : alk. paper)
1. Signal processing—Digital techniques. I. Title.
TK5102.9.R673 2011
621.382'2--dc22

2010040954

ISBN-13: 978-0-13-158334-4
ISBN-10: 0-13-158334-4

Text printed in the United States on recycled paper at Edwards Brothers in Ann Arbor, Michigan
First printing, November 2010

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Preface

Standard advice for writing a preface tells the author to begin by answering the question, "Why did you write this book?" The published answers almost always include an explanation of how something is still missing in the already vast body of existing literature, and how the book in question represents a valiant attempt to fill the void at least partially.

This book is no exception. There still is a dearth of *good* collections of step-by-step procedures, or *recipes,* for design and implementation of anything beyond just the most elementary DSP procedures. This book *is* an attempt to fill this void—at least partially. However, the tagline for this book is most definitely not meant to be, "Get a result without really gaining much understanding along the way." Here, the focus is clearly on the recipes, but supporting explanations and mathematical material are also provided. This supporting material is set off in such a way so that it is easily bypassed if the reader so desires.

This book provides an opportunity to delve deeper into the nuances of certain interesting topics within DSP. A good alternative title might be *Exploring the Nooks and Crannies of Digital Signal Processing*. As with all books, every reader will not resonate with every topic, but I'm confident that each reader will share an interest in a large subset of the topics presented.

Note 1, Navigating the DSP Landscape, provides diagrams that map the relationships among all the book's various topics. One diagram is dedicated to processing techniques that operate on real-valued digital signals to modify in some way the properties of those signals while leaving their fundamental real-valued and digital natures intact. A second diagram is dedicated to processing techniques that are concerned primarily with conversion between real-valued digital signals and other entities such as analog signals, complex-valued signals, and estimated spectra.

Many of the Notes include examples that demonstrate an actual application of the technique being presented. Most sections use MATLAB tools for routine tasks such as designing the digital filters that are used in the reference designs. When appropriate, the use of these tools is discussed in the text. The results provided in Note 66, Generating I and Q Channels Digitally: Generalization of Rader's Approach, were generated by a modified version of the `PracSim` simulation package that is described in *Simulating Wireless Communication Systems* (Prentice Hall, 2004). However, the filter coefficients used in the simulation were generated using MATLAB. The examples make heavy use of MATLAB as a convenience. However, it is not my intent to make this a MATLAB "workbook" with projects and exercises that *require* the reader to use MATLAB, because I want the book to remain useful and attractive to readers who do not have access to MATLAB. The m-files for the MATLAB programs discussed in the book, as well as for programs used to generate some of the illustrations, can be found at the website www.informit.com/ph.

This book is not the best choice for a first book from which to learn DSP if you're starting from scratch. For this task, I recommend *Understanding Digital Signal Processing* by Richard Lyons (Prentice Hall, 2004). This book is, however, a good *N*+1st book for anyone—from novice to expert—with an interest in DSP. Its contents comprise an assortment of interesting tidbits, unique insights, alternative viewpoints, and rarely published techniques. The following are some examples.

- The set of five techniques for generating analytic signals presented in Notes 60 through 64 do not appear together in any single text.
- The visualization techniques used in Note 22 probably are not discussed anywhere else, because I came up with them while writing this book. These techniques follow directly from first principles, but I've never seen them explicitly presented elsewhere.
- Natural sampling, as discussed in Note 6, usually can be found only in older texts that cover traditional (that is, analog) communication theory.

My overarching goal was to write an easy-to-read book loaded with easy-to-access information and easy-to-use recipes. I hope I have succeeded.

Note 1

Navigating the DSP Landscape

Digital signal processing (DSP) is based on the notions that an analog signal can be digitized and that mathematical operations can effectively take the place of (or even surpass) electronic manipulations performed on the original analog signal. In the earliest days of DSP, its applications were limited to sonar and seismology because these fields utilized low-bandwidth signals that could be sampled at adequate rates using the available technology. As digital processing circuits and analog-to-digital converters have become faster and faster, the number of applications for DSP has exploded.

Hundreds of techniques (and variations thereof) are used in DSP, and it can be difficult to see the big picture—how all these various techniques relate to each other and to a particular application at hand. Rather than a comprehensive treatment of all the possible topics within DSP, this book is an attempt to document in-depth explorations of some of the "nooks and crannies" in DSP that often are glossed over in traditional texts. Figures 1.1 and 1.2 show diagramatically the realtionships among the various processing techniques explored in this book. The topic areas are arbitrarily split into two groups

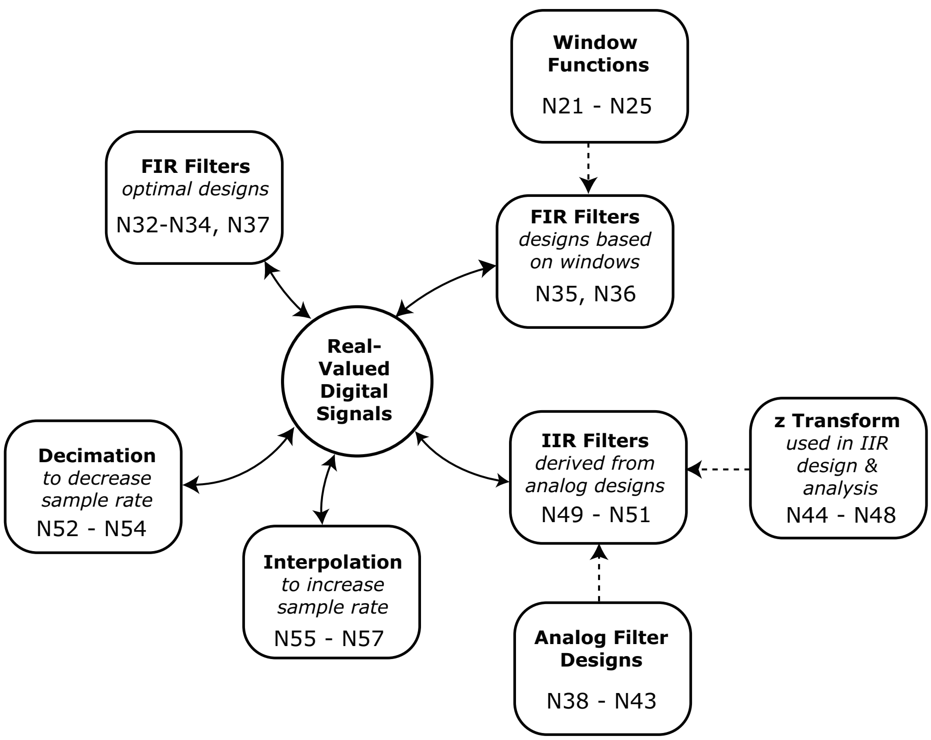


Figure 1.1 Processing techniques that modify the properties of real-valued digital signals. The numbers "Nnn" indicate the Notes in which each technique is discussed. Solid paths indicate "run-time" data connections. Dashed paths indicate "design-time" connections.

for organizational purposes. Figure 1.1 shows those techniques that are concerned primarily with operating on real-valued digital signals to modify in some way the properties of those signals while leaving their fundamental real-valued and digital natures intact. Given that complex-valued signals are really just quadrature pairs of real-valued signals, most of these techniques are easily extended to corresponding complex cases. Figure 1.2 shows those techniques that are concerned primarily with conversion between real-valued digital signals and other entities such as analog signals, complex-valued signals, and spectrum estimates.

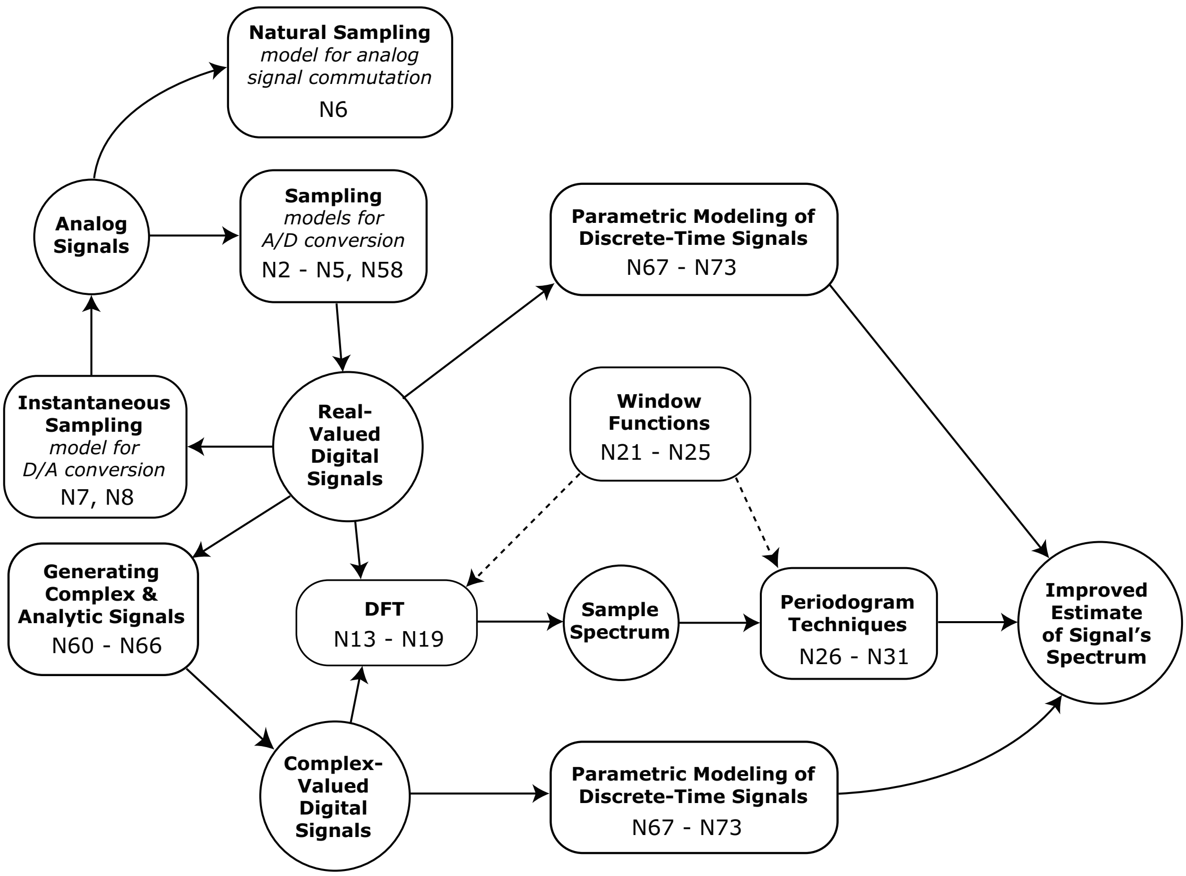


Figure 1.2 Processing techniques that convert real-valued digital signals to or from other things such as analog signals, complex-valued digital signals, or spectral estimates. The numbers "Nnn" indicate the Notes in which each technique is discussed. Solid paths indicate "run-time" data connections. Dashed paths indicate "design-time" connections.

Overview of Sampling Techniques

This note discusses the difference between *implicit* and *explicit* sampling. It introduces three different mathematical models of explicit sampling techniques—*ideal* sampling, *natural* sampling, and *instantaneous* sampling.

Digitization of an analog signal is the one process above all others that makes DSP such a useful technology. If it were limited to working with only those signals that orginate in digital form, DSP would be just an academic curiosity. Digitization actually comprises two distinct operations: sampling and quantization, which are usually analyzed separately.

2.1 Implicit Sampling Techniques

In *implicit sampling*, a sample measurement is triggered by the signal attaining some specified value or crossing some specified threshold. Recording the times at which zero-crossings occur in a bipolar signal is an example of implict sampling.

2.2 Explicit Sampling Techniques

Unlike implict sampling, in which samples are triggered by some aspect of signal behavior, in *explicit sampling*, signal values are measured at specified times without regard to the signal's behavior. Consider the continuous-time sinusoidal signal and its two-sided magnitude spectrum depicted in Figure 2.1. There are three explicit sampling techniques—*natural sampling*, *instantaneous sampling*, and *ideal sampling*—that can be used to sample such a signal. The results produced by these techniques, and the corresponding impacts on the signal's spectrum are compared in Key Concept 2.1.

Ideal Sampling

As depicted in Key Concept 2.1, zero-width samples take on instantaneous values of the analog signal. Neglecting quantization and timing errors, the sequence of values produced by an analog-to-digital converter can be modeled as the output of

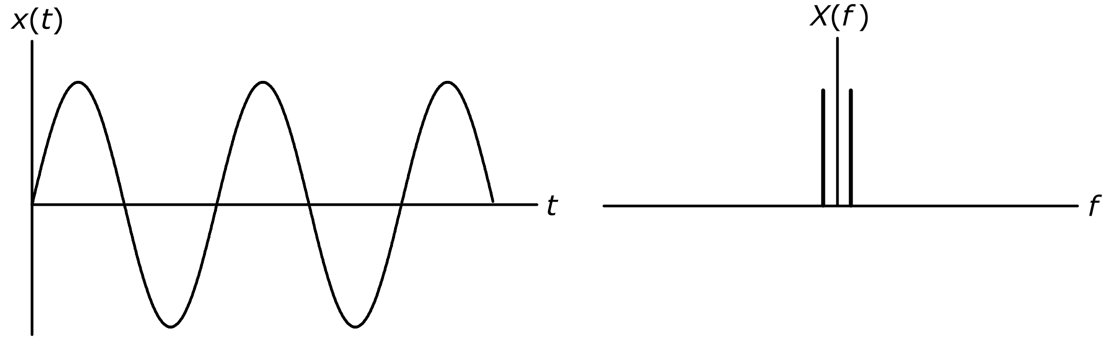


Figure 2.1 Continuous-time sinusoid and its two-sided magnitude spectrum

an ideal sampling process. Ideal sampling is discussed further in Note 3.

Natural Sampling

Nonzero-width samples have time-varying amplitudes that follow the contours of the analog signal, as shown in Key Concept 2.1. Commutator systems for time-division multiplexing of telegraph signals, first proposed in 1848, used an approximation to natural sampling. The sample pulses were created by gating a signal with rotating mechanical contacts. This multiplexing technique was subsequently applied to telephone signals in 1891. It was the application of natural sampling to telephony that first led to consideration of just how rapidly a continuous-time signal needed to be sampled in order to preserve fidelity and ensure the ability to reconstruct exactly the original, unsampled signal. Natural sampling is explored further in Note 6.

Instantaneous Sampling

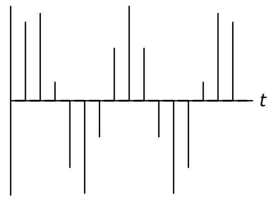
In instantaneous sampling, nonzero-width samples each have a constant amplitude that corresponds to the instantaneous value of the analog signal at the beginning of the sample. The sample values are held constant long enough to create flat-topped sample pulses. The output of a digital-to-analog converter (DAC) can be modeled as the output of an instantaneous sampling process, often as the limiting case in which the sample width equals the sampling interval. As discussed in Note 7, the results of the instantaneous sampling model play a key role in the specification of the analog filter used to smooth the DAC output.

Key Concept 2.1

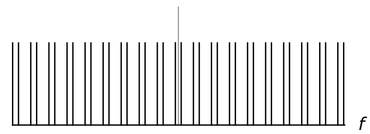
Explicit Sampling Techniques

Ideal Sampling

Models the output of an *A/D converter*



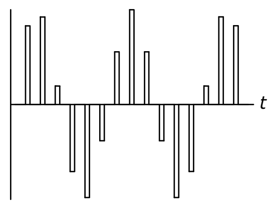
Zero-width samples take on instantaneous values of the analog signal.



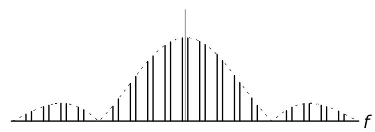
Spectral images all scaled equally

Instantaneous Sampling

Models the output of a *sample-and-hold amplifier*, *zero-order data hold*, or *D/A converter*



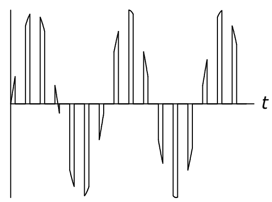
Non-zero-width samples, each with amplitude held constant over the width of the pulse



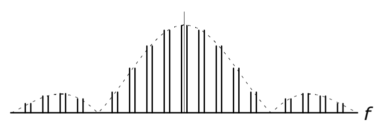
Spectral images, including baseband, are distorted by (sin x)/x envelope.

Natural Sampling

Models the output of an *analog demultiplexer*



Non-zero-width samples with time-varying amplitudes that follow the contours of the analog signal



Spectral images, with each image scaled by a factor that is constant over the image but varies image to image

Reference

1. H. D. Lüke, "The Origins of the Sampling Theorem," *IEEE Communications,* April 1999, pp. 106–108.

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