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Digital Communications: A Discrete-Time Approach

数字通信：离散时间方法

Michael Rice 著

（英文影印版）

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内 容 简 介

本书运用离散信号处理的原则来介绍和分析数字通信，连接了实时和离散方式，注重理论与实践相结合。涵盖了离散信号处理、离散滤波器设计、多速率处理及估计理论，并提出了基于离散信号的空间分析、数值算法。

本书可作为电子信息工程、通信工程专业本科生教材，也可作为相关领域工程技术人员的参考书。

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Foreword

Another book on Digital Communications! What possessed Michael Rice to write this book? It didn't dawn upon me to ask him because I already knew the answer based on the title: **Digital Communications,** ***A Discrete-Time Approach.*** The qualifier was the clue! Simply, it was time for this book to be written. It is clear to consumers and practitioners alike that Digital Communications has changed the way we communicate at a distance. The many text books and courses dealing with digital communications as well as the many communication sector companies that populate the Fortune-500 list bare witness to the importance of this technology in our society. Many of us have both witnessed and have guided the evolutionary transformation that has become the fabric of the physical layer in a communication system. This evolution has changed the way we manipulate and process waveforms in communication systems. We have seen Digital Signal Processing quickly replace most of the signal conditioning functions and tasks we perform in modulators and demodulators. Thus the title qualifier: ***A Discrete-Time Approach.***

This book is a response to the fact that Digital Signal Processing hardware and DSP algorithms have all but replaced analog hardware as the means to perform the various baseband and intermediate frequency signal processing functions required to perform modulation and demodulation. The change is dramatic. Cable TV modems, Satellite Modems, Local Area Wireless Network Modems, the ubiquitous Cell phone, Global Positioning System (GPS) receivers, and a multitude of consumer entertainment systems such as the audio MP3 player, the High Definition TV player, and the video TiVo player and others have embraced DSP as the core technology that enables the communication process.

An interesting note is that DSP based receivers are replacing analog receivers in non-digital modulation formats such as AM, FM and traditional analog TV. This happens in multi-mode receivers designed to demodulate digital modulation formats such as the OFDM based High Definition Radio signal but for marketing purpose must also demodulate the traditional analog AM and FM signal bands. Similar multimode receivers are required to demodulate digital modulated high definition TV signals as well as the traditional analog modulated National Television Standards Committee (NTSC) TV signals.

Traditionally the resources allocated to the communications task have been signal bandwidth (W) and signal energy (S) to noise (N) ratio as presented in Shannon's Capacity theorem: $C = W\log_2(S/N + 1)$. In the latter half of the 20-th century a third resource was added to the three legged structure we call Modern Communications. This third leg is computational complexity! We have learned that the modem needs a computer! Initially the computer performed transformations on data; *ones and zeros*. These transformations, owing much of their structure to Shannon, include channel coding and decoding, source coding and decoding, and encryption and decryption. In the past two decades the computer was assigned the additional heavy lifting task of applying transformations to sampled wave shapes. The collection of wave shape transformations is generally called Digital Signal Processing. The core of both sets of transformations is the digital computer. This core rides the bow-wave of Moore's law, which paraphrased states that "*for a fixed cost, the complexity, or computational power, of an integrated circuit doubles approximately every two years*". I regularly remind my students that there are only two justifications for the insertion of DSP in a communication system: these are improved performance and reduced cost. Moore's law offers both options! The second essential core that enables DSP insertion in communication systems is the availability of high performance analog-to-digital and digital-to-analog converters (ADC and DAC respectively). These devices perform the transduction and source coding tasks required to move waveform signals across the boundary separating the continuous analog domain and the discrete sample data domain.

Modern DSP based modulators present sampled data signals at their output ports while DSP demodulators expect sampled data signals at their input ports. Many systems contain mixed signal components in which the converters reside on the same chip performing the DSP functions. Since the industry and the economics of the process favors DSP implementations of the modulation and demodulation process it makes sense to teach communication systems from this implementation perspective.

While discussing perspectives, we have to keep in mind that when we implement a DSP based modulator or demodulator our task is to perform specific functions as opposed to copy a legacy implementation of these functions. Often a conventional receiver already contains an existing analog circuit implementation and we may be tempted to emulate that implementation. We have to keep in mind that our task is not to emulate the analog solution but rather to perform the function currently performed by the analog circuitry. In doing so it is wise to return to first principles with the design tools at our disposal which is likely a richer set of tools than those to which the analog designer had access when he or she derived the legacy analog solution. If we use the DSP based tools to replicate the analog circuit implementation we may also inherit legacy compromises made by the designer which were appropriate for the set of design tools available when the compromises were made. Approaching the processing tasks from first principles may offer solution unique to the DSP approach for which there is not an analog prototype solution. This approach may occasionally lead to the same solution to a problem as that obtained from the analog perspective but often it will lead to a unique DSP based solution. This text differs from earlier texts in that it not only presents the science of what to do in a communication system it also presents some of the modern art of how to do it.

Here we cite a few simple examples in support of the first principle design implementation approach. It is common for an analog prototype receiver to use a pair of recursive Butterworth filters as part of a down conversion signal path. We are not obligated to use a DSP based Butterworth filter to perform the same function in a digital implementation of the same process. We would likely use a linear phase non-recursive filter to avoid the group delay distortion associated with the recursive filter. Here we improve system performance by not emulating the analog design.

In another example, a common analog receiver design translates an intermediate frequency (IF) narrow band signal to baseband using a pair of mixers and low-pass filters. The process can certainly be emulated by DSP techniques but may be implemented more efficiently using a uniquely DSP based option. Such an example is based on multirate signal processing in which the signal can be aliased to baseband by simply reducing the sample rate of the sampled data time series representing the waveform. Aliasing is a viable tool available to the DSP designer that is not available to the analog designer!

One more example is the phase detector in a phase locked loop used for carrier synchronization. In an analog system the phase detector is formed by first hard limiting the input signal (with a clipping circuit) and then translating the, now constant amplitude, input signal to baseband with a pair of mixers. Our interest in the product signal is the phase angle of the resulting ordered pair. In the analog world it is difficult to access the angle directly so we usually use the small angle approximation that the sine of an angle is approximately the angle and use the output of the sine mixer as our approximate angle. In a DSP based solution we would never hard limit the sampled input signal. The resulting harmonics would alias back into the measurement process as undesired artifacts. We recognize that if we want the angle of the ordered pair we can simply compute the angle with an ATAN or by an equivalent but computationally simpler task of applying a CoRDiC rotate to the output of the digital mixers. The ATAN provides a superior phase detection S-curve compared to that offered by the imaginary part of the complex product.

Our final example related to first principles DSP based solution to communication system tasks is the timing recovery process. In timing recovery, samples at the output of the matched filter must be time aligned with the peaks of the periodic correlation function. This corresponds to aligning the output sample points with the maximum eye-opening of the eye-diagram. In legacy analog receivers alignment is accomplished by advancing or retarding the sampling clock by varying the control signal to the voltage controlled oscillator (VCO) supplying the clock signal. Modern receivers do not return to the analog domain via a DAC to affect the same control but rather acquire the time alignment in the DSP domain. We can do so because we trust Nyquist who assured us that values of the signal waveform collected at the Nyquist rate contain all the information residing in the original analog wave shape. In modern receivers the alignment is performed in one of two ways. In one method, an interpolator is used to compute the samples at the desired sample location from the nearby offset sample values. The PLL of the timing recovery system controls the interpolation process. In the second method, the receiver contains a bank of matched filters spanning a range of offsets between the time sample locations and the peak location of the matched filter. Here the PLL of the timing recovery system controls the process of identifying the correct time aligned filter.

Bear in mind that DSP performs other functions beside the replacement of the analog circuitry in a modem. Besides the traditional tasks of AGC, spectral translation, spectral shaping, matched filtering and channelization, timing and carrier synchronization, upsampling and down sampling, and signal synthesis DSP is also used to enhance the performance of the digital and analog hardware. DSP is used to equalize the group delay and amplitude distortion of the receiver analog filters, to balance the gain and phase mismatches between the quadrature signal paths of the modulator and demodulator, and to pre-compensate for the spectral distortion caused by the DAC spectral response. DSP techniques also cancel spectral intrusion components caused by parasitic coupling of sampling clock lines, including the DC generated by self mixing in the analog mixers and DC offsets caused by the ADC. DSP techniques also support a variety of optimal processing options that are difficult to implement in analog realizations. These include non-linear (such as TANH) signal-to-noise ratio dependent gains and signal-to-noise ratio estimators.

A final note of why the discrete-time signal processing has become so entrenched in modulator and demodulator design is the flexibility afforded by the ability to reconfigure the hardware by software upgrades or to modify system parameters by software control. Not the least of the DSP implementation benefits is product manufacturability. DSP parameters do not change with time and temperature and do not have values spanning a tolerance range as do analog components.

Now that I have had a chance to voice my biases on the importance of the *Discrete-Time Approach* to the implementation of *Digital Communication* Systems I would like to share my assessment of Michael Rice's book. I hope you are reading your own copy of this book! Are you?

First I will share with you the fact that I have known Michael Rice for some 10 years. He spent a year at SDSU where we shared common interests related to implementing digital receivers and Multirate Signal Processing. We tried to entice him to join our faculty but he resisted our offer and returned to BYU. Our loss, their gain! He is great instructor and has an interesting sense of humor. When you meet him ask him how Scooby-Doo was named.

I thoroughly enjoyed examining the book. In particular I enjoyed the footnotes distributed throughout the book. I read every one of them! The text starts with Chapter 1, the **Introduction**: a fun section to read. It continues with an initial review of continuous and discrete **Signal and Systems I** in Chapter 2 followed by a review of **Discrete-Time Techniques Useful for Digital Communications** in Chapter 3. Here the emphasis is processes and filters that have application in modem implementation. You can't write a text on communication without a section on probability so Chapter 4 presents a **Review of Probability Theory**. Chapter 5, **Linear Modulation 1: Modulation, Demodulation, and Detection**, presents digital communications using linear modulation techniques. This is followed by Chapter 6 **Linear Modulation 2: Performance** which presents measures of performance of in additive white Gaussian noise.

At this point, the text book swings off the beaten path by presenting the following chapters with an emphasis on discrete-time implementation of the offered material. Chapter 7 does a very nice job presenting the basics and implementation techniques of **Carrier Phase Synchronization**. The nice style continues in Chapter 7 with an introduction and presentation of techniques to obtain **Symbol Timing Synchronization**. Aligned with the interest in Discrete-Time signal processing techniques in communication systems, Chapter 9, titled **System Components**, presents detailed descriptions of important DSP building blocks. These include the ADC and DAC, Discrete-time Oscillators, Resampling Filters, the CoRDiC, and AGC systems. With all the pieces in place, Chapter 10 presents System Design showing modulator and demodulator architectures and channelizers. A number of important appendices follow the chapter text. These include a section on **Pulse Shapes** in Appendix A, an exposition on **Complex-Valued Representation for QAM** signals in Appendix B and a section on **Phase Locked Loops** in Appendix C. The book closes at the end of an extensive **Bibliography**. Of particular interest to the academic community and to the motivated student is the thorough set of problems following each of the nine chapters following the introduction as well as the three appendices. In a style I enjoy, similar to mine, appropriate well annotated MATLAB examples are distributed throughout the text.

I think this is fine book. I also think that the reader will be richer for his or her understanding of the Digital Communications through the modern DSP centric implementation perspectives it presents. Nice piece of work Michael Rice!

fred harris
San Diego State University

Preface

It was the summer of 1998. The Mathworks had introduced the Communications Blockset as part of Simulink the previous summer and I had been trying to use this as a basis for a series of extended design exercises for my senior-level elective course in digital communications at Brigham Young University. I was inspired to develop these exercises by my colleague Brad Hutchings, who had developed a series of laboratory exercises for his computer organization course. In these exercises, the students designed different parts of a simple RISC processor. Each subsystem was tested using a "test vector." The culminating laboratory exercise required the students to connect the subsystems together to form a complete processor and use it to run a simple program. My thinking was, if this could be done with RISC processors, it could be done with a QAM demodulator. Just as a processor has an instruction fetch/decode function, an arithmetic logic unit (ALU), registers, and control, a QAM demodulator has matched filters, a decision device, a carrier phase synchronizer, and a symbol timing synchronizer. In place of test vectors, I envisioned a sequence of known bits used to create a modulated waveform. I struggled to find just the right software environment for the students. Simulink with the Communications Blockset seemed to fit the bill.

I dabbled in the development of QAM demodulators using Simulink throughout the 1997–1998 academic year. My resource was the remarkably large number of text books devoted (in part or in whole) to digital communications. Although these texts explained pulse shapes, matched filtering, and synchronization, most were content to develop the theoretical underpinnings of these concepts in the continuous-time domain. This is not a bad thing. But the material offered little guidance on the proper way to perform these tasks in discrete time: What is the proper sample rate for the matched filter? How is this chosen? What are the trade-offs? How is carrier phase synchronization done in a discrete-time system? How is timing synchronization performed in a discrete-time system (do I really have to form a clock whose edges are aligned with the proper time)? And what about PLLs in discrete time?

Unable to make the proper connections between the continuous-time concepts developed in these texts and my discrete-time environment, I made many false assumptions and errors. Over and over, I ended up with a discrete-time system that was a kluged replica of a continuous-time system described in one of these texts. Something just wasn't right. Like any academic, I searched for conference and journal papers on the subject. Time and time again, I encountered an author who always (when the editors allowed it) spelled his name in lower case letters.

Now back to the summer of 1998. I received a notice that the Institute for Telecommunications Research[1] (ITS) was sponsoring its first International Symposium on Advanced Radio Technologies (ISART) in Boulder Colorado. As I parsed the Advance Program, I was delighted to see that now-somewhat-familiar-name in lower case letters: fred harris. Even more intriguing was the title of his presentation: "A Trap to Avoid: A DSP Radio is Not a Digitized Analog Radio." In an instant, I knew what my problem was and I knew where to find the solution. I attended the ISART in 1998 and listened to fred's talk like I had listened to no other. I really was a magic moment. fred and I became friends and I wound up visiting him on a sabbatical during the 1999–2000 academic year.

I learned a lot that year and I was eager to share it with students here at BYU. I revamped those Simulink exercises and retooled my course. Teaching my new-found knowledge was fun. But having the students retain enough of it to complete Simulink exercises was an entirely different matter. At first, I copied notes from a multitude of sources for use as resource material. (There were a few books devoted to discrete-time processing for digital communications. But these texts presumed a-priori knowledge of digital communications.) Inevitably, these sources used different terminology and different notation that made it more difficult than it needed to be for the students who were seeing for the first time. I produced notes for the class to augment these sources. Over time, we used the notes more and more and the available texts less and less. Well, you see what happened....

There are many challenges associated with writing a text book such as this one. First, the text must teach digital communication theory. Next, the text must present both continuous-time and discrete-time realizations in such a way that the differences between the two can be appreciated. This sounds much easier than it really is. Sufficient background material must also be included to produce a work as self-contained as possible. I also experimented with the development. Continuous-time random processes are an efficient and elegant way to describe most of the concepts fundamental to digital communications. Unfortunately, for seniors and first year graduate students, the newness of continuous-time random processes often gets in the way. In this text, I tried to leverage the experience most modern electrical engineering majors have with random variables. Much of the more theoretical development involves random variables.

This text assumes, as prerequisites, courses in linear systems, transform theory, and probability theory (at least through random variables). Chapters 2–4 present the necessary background material to embark on the journey through this text. Chapters 2 and 3 describe signals and systems while Chapter 4 is devoted to random variables. While most seniors and first year graduate students are familiar with linear systems and transforms in both continuous time and discrete time, the *connection* between the two is usually quite nebulous. For this reason, Chapter 2 devotes substantial attention to this very important topic. In addition, some of the peculiarities of discrete-time processing of band-pass signals are also included. This receives almost no attention in prerequisite courses. Chapter 3 introduces some material that may not be familiar: multi-rate signal processing and discrete-time filter design.

[1] ITS is the research and engineering branch of the National Telecommunications and Information Administration (NTIA), a part of the U.S. Department of Commerce.

The principle body of an introductory course in digital communications is embodied in Chapters 5–8. These chapters focus on modulation, detection, performance, and synchronization. I have tried to approach these topics from simultaneous points of view. Modulation, detection, and synchronization are described in both continuous time and discrete time. Practical descriptions as well as theoretical developments from basic principles are also included. Organizing these different points of view in each chapter was a tremendous challenge: I did not have any good examples to follow. The continuous-time systems are those presented in the traditional texts. The discrete-time counterparts start with discrete-time versions of the continuous-time system where appropriate (symbol timing synchronization is the big exception). The full power of discrete-time processing is not exploited until Chapter 10—and yes, it was hard to wait to describe the good stuff. The practical descriptions use pulse amplitude modulation (PAM) and quaternary phase shift keying (QPSK) as examples. Chapters 5, 7, and 8 end with a derivation of the corresponding maximum likelihood estimator. I realize this is the reverse order from the traditional approach. I have selected the reverse order based on my experience that most students tend to learn specific-to-general rather than the other way around. When the principles are taught using a few special cases, the generalizations tend to be more meaningful.

The description of the digital modulation formats discussed in Chapter 5 depends on pulse shapes, which are described (in both continuous time and discrete time) in Appendix A. The synchronizers described in Chapters 7 and 8 are based on the phase locked loop (PLL) described in Appendix C (in both continuous time and discrete time).

A more extensive treatment of discrete-time processing for digital communications is presented in Chapters 9 and 10. Chapter 9 introduces analog-to-digital and digital-to-analog converters, discrete-time oscillators, resampling filters, the CoRDiC and CoRDiC-like algorithms, and discrete-time automatic gain control. The emphasis is on the performance of these components from a systems level point of view. These components are applied to discrete-time techniques for modulators, demodulators, and channelizers in Chapter 10.

I have tried to write an honest text book. Each chapter ends with a "Notes and References" section that provides some historical notes, references, and a description of related topics that were not discussed in the text. No text presents a completely exhaustive treatment. And this one is no exception. My intent is to describe the basic details that are common to most digital communication systems. Hopefully, this provides a context in which the information provided in the references is more meaningful.

I am indebted to a large number of people who helped make this text possible. I really do feel as though I am standing on the shoulders of giants. First is fred harris, my friend, colleague, and entertainer, who taught me a lot of this stuff. Second is my wife who had to suffer the tribulations of living in paradise (San Diego) during my sabbatical year with fred harris. And my children, who were fatherless on far too many evenings as I toiled away on this project. I am also grateful to my department chair, Prof. Rick Frost, who provided summer support and reduced teaching assignments from time to time as I completed this manuscript. I have many friends in industry and academia took time to explain to me things that I otherwise would not know. You know who you. Of special mention are those who reviewed this text: Xiaohua Li, State University of New York at Binghamton; Mohammad Saquib, University of Texas at Dallas; Jacob Gunther, Utah State University; Tongtong Li,

Michigan State University; John J. Shynk, University of California, Santa Barbara. Their comments and collective wisdom have made this a much better book than I could have produced on my own. Finally, I thank my two editors at Pearson Prentice-Hall: Tom Robbins, who got me into this mess, and Alice Dworkin, who helped me get out.

Michael Rice
Provo, Utah

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