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—— 信息技术学科与电气工程学科系列

15

Real-Time Digital Signal Processing

Implementations, Applications and Experiments With
the TMS320C55X

实时数字信号处理

—— 基于TMS320C55X的实现、应用和实验

Sen M. Kuo

Bob H. Lee

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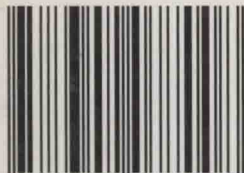


内容简介

“实时”功能是高性能数字信号处理器和DSP应用中面临的最新挑战。它的成功应用不仅需要彻底理解DSP理论，还要全面掌握实时DSP设计和应用技术。本书为读者提供了一本非常实用的教材和参考书。本书具有如下特点：

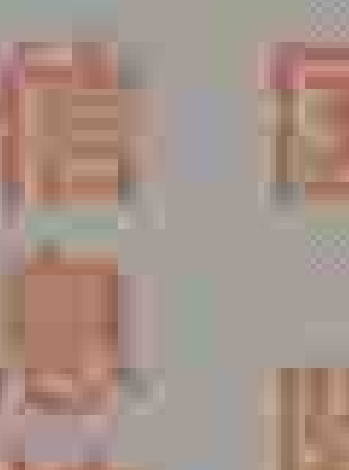
- ◆ 具有使用MATLAB, C和TMS320C55X汇编语言的丰富的习题和实验，涵盖了从基本概念到通信应用的所有内容。
- ◆ 对数字信号处理的理论问题的讨论，没有复杂的理论推导，而且都是围绕如何实现来展开的。
- ◆ 指导如何选择DSP芯片使之满足各种不同应用的要求，如实时性约束、不同硬件选择、定点或浮点器件等。
- ◆ 配套网址上(<http://www.ceet.niu.edu/faculty/kuo/books/rtdsp.html>)有全书各章实验的软件，方便读者使用。

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出版说明

郑大钟

清华大学信息科学与技术学院

当前,在我国的高等学校中,教学内容和课程体系的改革已经成为教学改革中的一个非常突出的问题,而为数不少的课程教材中普遍存在的“课程体系老化,内容落伍时代,本研层次不清”的现象又是其中的急需改变的一个重要方面。同时,随着科教兴国方针的贯彻落实,要求我们进一步转变观念扩大视野,使教学过程适应以信息技术为先导的技术革命和我国社会主义市场经济体制的需要,加快教学过程的国际化进程。在这方面,系统地研究和借鉴国外知名大学的相关教材,将会对推进我们的课程改革和推进我国大学教学的国际化进程,乃至对我们一些重点大学建设国际一流大学的努力,都将具有重要的借鉴推动作用。正是基于这种背景,我们决定在国内推出信息技术学科和电气工程学科国外知名大学原版系列教材。

本系列教材的组编将遵循如下的几点基本原则。(1)书目的范围限于信息技术学科和电气工程学科所属专业的技术基础课和主要的专业课。(2)教材的范围选自于具有较大影响且为国外知名大学所采用的教材。(3)教材属于在近5年内所出版的新书或新版书。(4)教材适合于作为我国大学相应课程的教材或主要教学参考书。(5)每本列选的教材都须经过国内相应领域的资深专家审看和推荐。(6)教材的形式直接以英文原版形式印刷出版。

本系列教材将按分期分批的方式组织出版。为了便于使用本系列教材的相关教师和学生从学科和教学的角度对其在体系和内容上的特点和特色有所了解,在每本教材中都附有我们所约请的相关领域资深教授撰写的影印版序言。此外,出于多样化的考虑,对于某些基本类型的课程,我们还同时列选了多于一本的不同体系、不同风格和不同层次的教材,以供不同要求和不同学时的同类课程的选用。

本系列教材的读者对象为信息技术学科和电气工程学科所属各专业的本科生,同时兼顾其他工程学科专业的本科生或研究生。本系列教材,既可采用作为相应课程的教材或教学参考书,也可提供作为工作于各个技术领域的工程师和技术人员的自学读物。

组编这套国外知名大学原版系列教材是一个尝试。不管是书目确定的合理性,教材选择的恰当性,还是评论看法的确切性,都有待于通过使用和实践来检验。感谢使用本系列教材的广大教师和学生的支持。期望广大读者提出意见和建议。

Real-Time Digital Signal Processing

Implementations, Applications and Experiments with the TMS320C55X

影 印 版 序

本书的中文名字可以译为《实时数字信号处理——基于 TMS320C55X 的实现、应用和实验》。所谓“实时 (Real-Time) 实现”，是指一个实际的系统在人们听觉、视觉或按任务要求所允许的时间范围内能及时地完成对输入信号的处理并将其输出。例如，我们每天使用的手机、将要普及的数字电视等，都是实时的数字信号处理系统。要想在极短的时间内完成对信号的处理，一方面需要快速的算法、高效的编程，另一方面，则需要高性能的硬件支持。数字信号处理器 (DSP) 即是为实时实现数字信号处理任务而特殊设计的高性能的一类 CPU。

随着信息科学和微电子技术的飞速发展，数字信号处理的理论及数字信号处理器已广泛应用于通信、家电、航空航天、工业测量和控制、生物医学工程及军事等许许多多的领域。由于设计和实现一个实时的数字信号处理系统不仅需要系统地掌握信号处理的理论，而且要熟练地掌握 DSP 硬件的知识，因此，对设计者的要求是非常高、也是相当全面的。美国伊利诺斯大学电机工程系的 Sen M. Kuo 教授和美国德州仪器公司的 Bob H. Lee 博士合著的这本书为培养这一类人才提供了一本值得推荐的好教材。

本书共分 9 章。第 1 章简要介绍了实时数字信号处理系统的基本概念；第 2 章介绍了美国德州仪器公司最新推出的 DSP 芯片——TMS320C55X 的结构及编程；第 3 章讨论了 DSP 系统实时实现时的结构、量化及溢出等问题；第 4~8 章分别讨论数字信号处理中的时域分析、频域分析、FIR 和 IIR 滤波器设计、快速傅里叶变换及自适应滤波等基本问题；第 9 章较为详细地给出了 DSP 实时系统应用的例子，如信号产生 (正弦信号、Chirp 信号、噪声等)，自适应噪声抵消及语音增强等。

本书的特点是：

1. 全书以 TMS320C55X DSP 芯片为主线，系统地介绍了实时数字信号处理系统的设计和实现问题。

2. 本书虽然也以较大的篇幅讨论了数字信号处理的理论问题，但目的都是围绕着如何实现来展开的。

3. 本书每一章都安排了实验举例，或用 C55 的汇编语言，或用 C 语言来说明该章内容在 DSP 上的实现问题。通过这些实验，读者可以很快地掌握实时 DSP 系统的设计和实现问题。

4. 除第 9 章，本书每一章都附有丰富的习题。这些习题一部分是涉及信号处理的理论

问题，但大部分是有关实时 DSP 系统实现的上机练习。

5. 本书没有涉及复杂的理论推导，因此通俗易懂。

本书可作为本科生、研究生的教材，也可作为工程技术人员的参考书。

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清华大学生物医学工程系
2003 年 10 月

Preface

Real-time digital signal processing (DSP) using general-purpose DSP processors is very challenging work in today's engineering fields. It promises an effective way to design, experiment, and implement a variety of signal processing algorithms for real-world applications. With DSP penetrating into various applications, the demand for high-performance digital signal processors has expanded rapidly in recent years. Many industrial companies are currently engaged in real-time DSP research and development. It becomes increasingly important for today's students and practicing engineers to master not only the theory of DSP, but equally important, the skill of real-time DSP system design and implementation techniques.

This book offers readers a hands-on approach to understanding real-time DSP principles, system design and implementation considerations, real-world applications, as well as many DSP experiments using MATLAB, C/C++, and the TMS320C55x. This is a practical book about DSP and using digital signal processors for DSP applications. This book is intended as a text for senior/graduate level college students with emphasis on real-time DSP implementations and applications. This book can also serve as a desktop reference for practicing engineer and embedded system programmer to learn DSP concepts and to develop real-time DSP applications at work. We use a practical approach that avoids a lot of theoretical derivations. Many useful DSP textbooks with solid mathematical proofs are listed at the end of each chapter. To efficiently develop a DSP system, the reader must understand DSP algorithms as well as basic DSP chip architecture and programming. It is helpful to have several manuals and application notes on the TMS320C55x from Texas Instruments at <http://www.ti.com>.

The DSP processor we will use as an example in this book is the TMS320C55x, the newest 16-bit fixed-point DSP processor from Texas Instruments. To effectively illustrate real-time DSP concepts and applications, MATLAB will be introduced for analysis and filter design, C will be used for implementing DSP algorithms, and Code Composer Studio (CCS) of the TMS320C55x are integrated into lab experiments, projects, and applications. To efficiently utilize the advanced DSP architecture for fast software development and maintenance, the mixing of C and assembly programs are emphasized.

Chapter 1 reviews the fundamentals of real-time DSP functional blocks, DSP hardware options, fixed- and floating-point DSP devices, real-time constraints, algorithm development, selection of DSP chips, and software development. In Chapter 2, we introduce the architecture and assembly programming of the TMS320C55x. Chapter 3 presents some fundamental DSP concepts in time domain and practical considerations for the implementation of digital filters and algorithms on DSP hardware. Readers who are familiar with these DSP fundamentals should be able to skip through some of these sections. However, most notations used throughout the book will be defined in this chapter. In Chapter 4, the Fourier series, the Fourier transform, the z -transform, and the discrete Fourier transforms are introduced. Frequency analysis is extremely helpful

in understanding the characteristics of both signals and systems. Chapter 5 is focused on the design, implementation, and application of FIR filters; digital IIR filters are covered in Chapter 6, and adaptive filters are presented in Chapter 8. The development, implementation, and application of FFT algorithms are introduced in Chapter 7. In Chapter 9, we introduce some selected DSP applications in communications that have played an important role in the realization of the systems.

As with any book attempting to capture the state of the art at a given time, there will necessarily be omissions that are necessitated by the rapidly evolving developments in this dynamic field of exciting practical interest. We hope, at least, that this book will serve as a guide for what has already come and as an inspiration for what will follow. To aid teaching of the course a Solution Manual that presents detailed solutions to most of the problems in the book is available from the publisher.

Availability of Software

The MATLAB, C, and assembly programs that implement many DSP examples and applications are listed in the book. These programs along with many other programs for DSP implementations and lab experiments are available in the software package at <http://www.ceet.niu.edu/faculty/kuo/books/rtdsp.html> and <http://pages.prodigy.net/sunheel/web/dspweb.htm>. Several real-world data files for some applications introduced in the book also are included in the software package. The list of files in the software package is given in Appendix D. It is not critical you have this software as you read the book, but it will help you to gain insight into the implementation of DSP algorithms, and it will be required for doing experiments at the last section of each chapter. Some of these experiments involve minor modification of the example code. By examining, studying and modifying the example code, the software can also be used as a prototype for other practical applications. Every attempt has been made to ensure the correctness of the code. We would appreciate readers bringing to our attention (kuo@ceet.niu.edu) any coding errors so that we can correct and update the codes available in the software package on the web.

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Sen M. Kuo and Bob H. Lee

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