《数字信号处理》

图书基本信息

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作者：蔡坤宝

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前言

21世纪是国际化的知识经济时代，理工科教育已发生深刻的变化，各专业涉及的新理论与新技术发展日新月异，科技的创新在很大程度上已依赖于信息的及时获取、准确理解和有效利用。当今，数字信号处理理论和算法的研究、应用与实现技术的发展，以及其在现代信息与通信技术中的重要性和巨大潜力，已超越了初期人们所做的估计与预测。与此同时，社会对高素质信息技术人才的需求对高等学校专业基础课程的教学质量提出了越来越高的要求，然而“数字信号处理”及其相关技术的基础课程所能获得的学时数反而在减少。有知名专家与学者将当今的教学改革难题归结为：人类知识的无限积累与个人学习能力和时间的有限形成间日益尖锐的矛盾。　　高等教育的国际化是当今教育与教学改革的必然趋势，国际视野和国际交流能力已成为我国高等学校人才培养的一项基本要求。为适应教学改革的新要求，我总结了“数字信号处理”教学与科研工作20余年所积累的经验与创新成果，并力图继承国内老一代专家与学者编著的优秀教材的知识体系结构严谨、系统性强的特色与传统，参考了20余本国内外一流或知名高等学校的优秀教材，通过消化、吸收和创新，编写了本书。2007年8月本书第1版由电子工业出版社出版。此后，本书在作者本人执教的重庆大学通信工程学院本科数字信号处理课程双语教学班连续使用了4届。本书第1版于2010年已脱销，由于作者工作繁忙，直至今日才进行修订工作。　　我在教材内容的选择、知识体系的组织和编排方面，做了慎重考虑——本书内容要适应我国高等学校的教学和课程设置的实际情况。面对数字信号处理知识内容迅速扩展和学时数有限的实际情况，我在编写过程中始终贯彻的基本思想是：使读者系统地掌握离散时间信号与系统分析与设计的基本理论；在两种常用的数字信号处理技术方面（基于DFT的连续时间信号的频谱分析、IIR和FIR滤波器那样的数字信号处理系统的设计），力求使读者对分析与设计的原理和方法有较透彻的理解与掌握；在数字信号处理系统中的有限字长效应和多抽样率数字信号处理方面打下一定的基础；通过进一步自学或学习更加深入的后续课程，即可较容易地扩充数字信号处理的理论知识与实际技能。　　基于我使用第1版作为教材的实际经验与体会，第2版保留了第1版中的主要内容，以适应目前本科教学的基本需要；压缩了篇幅，以适应学时数减少的实际情况；修正了第1版中的文字与公式符号错误，润色了语句文字。具体修订情况如下：　　① 基于提高课堂教学效率和提高学生分析与解决问题能力的考虑，对第1版第2、3章中一些相对较简单的例题进行了精简。这些例题的题目被插入到相应章的习题中，可以作为学生课后作业。　　② 考虑到学时数有限的实际情况，而且第1版未介绍频率抽样滤波器设计的内容，删除原6.3.4节。　　③ 基于方便教师检验课堂教学效果的考虑，删除第1版附录F课后习题参考答案。为方便学生自学，课后习题参考答案可登录华信教育资源网注册下载。

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内容概要

《数字信号处理(第2版)(英文版)》系统地阐述了数字信号处理所涉及的信号与系统分析和系统设计的基本理论、基本分析与设计方法、基本算法和处理技术。《数字信号处理(第2版)(英文版)》共10章，主要内容包括：离散时间信号与系统的基本概念，离散时间信号与系统的变换域分析，包括z变换和离散时间傅里叶变换、连续时间信号的抽样与重建，离散傅里叶变换及其快速算法（FFT），数字滤波器实现的基本结构，IIR和FIR数字滤波器的设计原理与基本设计方法，数字信号处理中的有限字长效应，多抽样率数字信号处理。《数字信号处理(第2版)(英文版)》配有多媒体电子课件、英文版教学大纲、习题指导与实验手册。

《数字信号处理(第2版)(英文版)》可以作为电子与通信相关专业的本科数字信号处理课程中英文双语教学的教材，或中文授课的英文版教学参考书，也可供从事数字信号处理的工程技术人员学习参考。《数字信号处理(第2版)(英文版)》尤其适合初步开展数字信号处理课程中英文双语授课的师生选用。

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作者简介

蔡坤宝博士，重庆大学通信工程学院教授，信号与信息处理硕士学位点负责人。长期从事信号与信息处理的教学与科研工作。近十余年来，积极探索和实施中英文双语教学，现任重庆大学大类系列课程“信号与系统”建设项目负责人，重庆市精品课程“信号与线性系统”负责人、国家级双语教学示范课程“信号与系统”负责人，并承担重庆市精品课程“数字信号处理”的建设工作。

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