

<<数字信号处理>>

图书基本信息

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内容概要

《数字信号处理（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。

全书共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（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（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语双语教学的教材，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。本书尤其适合初步开展数字信号处理课程中英语双语授课的教师与学生选用。

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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。多年来致力于随机信号的产生与处理、生物组织粘弹性波动的有限元分析、现代信号处理及其应用和人工神经网络等方面的研究工作。十余年来，积极探索和实施中英文双语教学，现任重庆市级精品课程“信号与线性系统”的负责人，并参加重庆大学精品课程“数字信号处理”的建设工作。

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