

<<数字信号处理>>

图书基本信息

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内容概要

为了给读者在理论和实践应用之间进行合理的平衡，本书严谨地介绍了现代数字信号处理的基本概念和技术，并介绍了相关的算法和应用。

本书涵盖了线性离散时间系统分析的时域和频域方法，还涉及了诸如采样、数字滤波器设计、滤波器实现、去卷积、插值、状态矢量空间方法、频谱分析等相关主题的内容。

本书不仅要求对诸多示例、练习的理解，而且更强调对数字信号算法进行软件实现的实践环节。

本书特点：

- 覆盖离散傅立叶变换（DFT）和快速傅立叶变换（FFT）算法，并对其进行了更加合理清晰的重组——介绍DFT，并在阐明傅立叶分析后描述其快速计算
- 描述模拟信号模数转换中涉及的运算和技术
- 在时域研究线性时不变离散时间系统和离散时间信号的特性
- 考虑双边 z 变换和单边 z 变换，并描述了求 z 反变换的方法
- 在频域分析信号与系统，给出连续时间信号与离散时间信号的傅立叶级数与傅立叶变换
- 实现无限冲激响应（IIR）与有限冲激响应（FIR）系统的结构形式，包括直接型、级联型、并联型、格型和格梯型
- 采样频率转换基础与多采样率转换系统
- 功率谱估计的详细测试，并讨论了非参数方法、基于模型的方法和基于特征分解的方法，包括MUSIC算法和ESPRIT算法
- 全书囊括了许多实例，并提供大约500个可解决的问题

本书既适合作为本科生学习离散系统和数字信号处理课程的教材，又适合研究生一年级学习数字信号处理课程时作为教材使用。

作者简介

John

G.Proakis长期担任美国东北大学的电气工程教授，并担任该校电气与计算机工程系主任之职达14年之久。

他分别从麻省理工学院和哈佛大学获得了硕士和博士学位。

Proakis教授是众多成功教材的作者，其教材在世界上具有相当的影响力。

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