<<数字信号处理实践方法>>

图书基本信息

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作者：EmmanuelCIfeachor

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前言

2001年7月间，电子工业出版社的领导同志邀请各高校十几位通信领域方面的老师，商量引进国外教材问题。
与会同志对出版社提出的计划十分赞同，大家认为，这对我国通信事业、特别是对高等院校通信学科的教学工作会很有好处。
教材建设是高校教学建设的主要内容之一。
编写、出版一本好的教材，意味着开设了一门好的课程，甚至可能预示着一个崭新学科的诞生。
20世纪40年代MIT林肯实验室出版的一套28本雷达丛书，对近代电子学科、特别是对雷达技术的推动作用，就是一个很好的例子。
我国领导部门对教材建设一直非常重视。
20世纪80年代，在原教委教材编审委员会的领导下，汇集了高等院校几百位富有教学经验的专家，编写、出版了一大批教材；很多院校还根据学校的特点和需要，陆续编写了大量的讲义和参考书。
这些教材对高校的教学工作发挥了极好的作用。
近年来，随着教学改革不断深入和科学技术的飞速进步，有的教材内容已比较陈旧、落后，难以适应教学的要求，特别是在电子学和通信技术发展神速、可以讲是日新月异的今天，如何适应这种情况，更是一个必须认真考虑的问题。
解决这个问题，除了依靠高校的老师和专家撰写新的符合要求的教科书外，引进和出版一些国外优秀电子与通信教材，尤其是有选择地引进一批英文原版教材，是会有好处的。
一年多来，电子工业出版社为此做了很多工作。
他们成立了一个“国外电子与通信教材系列”项目组，选派了富有经验的业务骨干负责有关工作，收集了230余种通信教材和参考书的详细资料，调来了100余种原版教材样书，依靠由20余位专家组成的出版委员会，从中精选了40多种，内容丰富，覆盖了电路理论与应用、信号与系统、数字信号处理、微电子、通信系统、电磁场与微波等方面，既可作为通信专业本科生和研究生的教学用书，也可作为有关专业人员的参考材料。
此外，这批教材，有的翻译为中文，还有部分教材直接影印出版，以供教师用英语直接授课。
希望这些教材的引进和出版对高校通信教学和教材改革能起一定作用。
在这里，我还要感谢参加工作的各位教授、专家、老师与参加翻译、编辑和出版的同志们。
各位专家认真负责、严谨细致、不辞辛劳、不怕琐碎和精益求精的态度，充分体现了中国教育工作者和出版工作者的良好美德。
随着我国经济建设的发展和科学技术的不断进步，对高校教学工作会不断提出新的要求和希望。
我想，无论如何，要做好引进国外教材的工作，一定要联系我国的实际。
教材和学术专着不同，既要注意科学性、学术性，也要重视可读性，要深入浅出，便于读者自学；引进的教材要适应高校教学改革的需要，针对目前一些教材内容较为陈旧的问题，有目的地引进一些先进的和正在发展中的交叉学科的参考书；要与国内出版的教材相配套，安排好出版英文原版教材和翻译教材的比例。
我们努力使这套教材能尽量满足上述要求，希望它们能放在学生们的课桌上，发挥一定的作用。
最后，预祝“国外电子与通信教材系列”项目取得成功，为我国电子与通信教学和通信产业的发展培土施肥。
也恳切希望读者能对这些书籍的不足之处、特别是翻译中存在的问题，提出意见和建议，以便再版时更正。

内容概要

《数字信号处理实践方法（第二版）》根据实际工程应用和具体实例，详细介绍了数字信号处理（DSP）领域内的基本概念和相关技术。
全书共分为14章，首先讲解了DSP的基本概念及其应用，并从实际的例子出发，阐述了DSP的一些基本内容，如信号的抽样、量化及其在实时DSP上的内涵。
然后，作者介绍了离散变换（DFT和FFT），离散时间信号与系统分析的工具（z变换），以及DSP的基本运算（相关和卷积），并分析了数字滤波器设计的实际问题。
《数字信号处理实践方法（第二版）》还介绍了多抽样率数字信号处理、自适应数字滤波器、谱估计及其分析等现代数字信号处理理论，最后讨论了通用和专用数字信号处理器、定点DSP系统有限字长效应分析及DSP的应用和设计实例。
另外，书中还提供了有关范例和实验的MATLAB实现方法。
《数字信号处理实践方法（第二版）》可作为通信与电子信息类专业高年级本科生和研究生的教材或教学参考书，而且对于相关学科的工程技术人员也具有很好的参考价值。

作者简介

Emmanuel C.Ifeachor：智能电子系统方向的教授，英国普利茅斯大学通信、网络和信息系统研究中心主任。

Barriv W.Jervis：英国Sheffield Hallam大学电子工程系教授。

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编辑推荐

《数字信号处理实践方法（第二版）》是《数字信号处理实践方法》一书的第二版，除了修正原有内容之外，还增加了许多对工程应用日显重要的新内容。
作者将理论与工程的应用紧密结合，根据实际工程应用和具体实例来讲解数字信号处理领域内的基本概念。
这本实用的、介绍性的教材涵盖了电学、电子工程和通信工程等专业的专业课程中与数字信号处理相关的绝大部分内容。
此外，《数字信号处理实践方法（第二版）》还介绍了许多DSP技术，例如自适应滤波、多速率信号处理等，这些技术与工业应用及正在进行的科学研究紧密相关。