

<<数字信号处理实践方法>>

图书基本信息

书名：<<数字信号处理实践方法>>

13位ISBN编号：9787505389281

10位ISBN编号：7505389289

出版时间：2003-8

出版时间：电子工业

作者：EmmanuelCIfeachor

页数：933

版权说明：本站所提供下载的PDF图书仅提供预览和简介，请支持正版图书。

更多资源请访问：<http://www.tushu007.com>

## &lt;&lt;数字信号处理实践方法&gt;&gt;

## 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 and 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。

教材和学术专著不同,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。

## <<数字信号处理实践方法>>

### 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。

<<数字信号处理实践方法>>

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、网络和信息系統研究中心主任。

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

## 书籍目录

1 introduction  
 1.1 digital signal processing and its benefits  
 1.2 application areas  
 1.3 key dsp operations  
 1.3.1 convolution  
 1.3.2 correlation  
 1.3.3 digital filtering  
 1.3.4 discrete transformation  
 1.3.5 modulation  
 1.4 digital signal processors  
 1.5 overview of real-world applications of dsp  
 1.6 audio applications of dsp  
 1.6.1 digital audio mixing  
 1.6.2 speech synthesis and recognition  
 1.6.3 the compact disc digital audio system  
 1.7 telecommunication applications of dsp  
 1.7.1 digital cellular mobile telephony  
 1.7.2 set-top box for digital television reception  
 1.7.3 adaptive telephone echo cancellation  
 1.8 biomedical applications of dsp  
 1.8.1 fetal ecg monitoring  
 1.8.2 dsp-based closed loop controlled anaesthesia  
 1.9 summary  
 problems  
 references  
 bibliography  
 2 analog i/o interface for real-time dsp systems  
 2.1 typical real-time dsp systems  
 2.2 analog-to-digital conversion process  
 2.3 sampling- lowpass and bandpass signals  
 2.3.1 sampling lowpass signals  
 2.3.2 sampling bandpass signals  
 2.4 uniform and non-uniform quantization and encoding  
 2.4.1 uniform quantization and encoding (linear pulse code modulation (pcm))  
 2.4.2 non-uniform quantization and encoding (nonlinear pcm)  
 2.5 oversampling in aid conversion  
 2.5.1 introduction  
 2.5.2 oversampling and anti-aliasing filtering  
 2.5.3 oversampling and adc resolution  
 2.5.4 an application of oversampling - single-bit (oversampling) adc  
 2.6 digital-to-analog conversion process: signal recovery  
 2.7 the dac  
 2.8 anti-imaging filtering  
 2.9 oversampling in d/a conversion  
 2.9.1 oversampling d/a conversion in the cd player  
 2.10 constraints of real-time signal processing with analog input/output signals  
 2.11 application examples  
 2.12 summary  
 problems  
 references  
 bibliography  
 3 discrete transforms  
 3.1 introduction  
 3.1.1 fourier series  
 3.1.2 the fourier transform  
 3.2 dft and its inverse  
 3.3 properties of the dft  
 3.4 computational complexity of the dft  
 3.5 the decimation-in-time fast fourier transform algorithm  
 3.5.1 the butterfly  
 3.5.2 algorithmic development  
 3.5.3 computational advantages of the fft  
 3.6 inverse fast fourier transform  
 3.7 implementation of the fft  
 3.7.1 the decimation-in-frequency fft  
 3.7.2 comparison of dit and dif algorithms  
 3.7.3 modifications for increased speed  
 3.8 other discrete transforms  
 3.8.1 discrete cosine transform  
 3.8.2 walsh transform  
 3.8.3 hadamard transform  
 3.8.4 wavelet transform  
 3.8.5 multiresolution analysis by the wavelet method  
 3.8.6 signal representation by singularities: the wavelet transform method  
 3.9 an application of the dct: image compression  
 3.9.1 the discrete cosine transform  
 3.9.2 2d dct coefficient quantization  
 3.9.3 coding  
 3.10 worked examples  
 problems  
 references  
 appendices  
 3a c language program for direct dft computation  
 3b c program for radix-2 decimation-in-time fft  
 3c dft and fft with matlab references for appendices  
 4 the z-transform and its applications in signal processing  
 4.1 discrete-time signals and systems  
 4.2 the z-transform  
 4.3 the inverse z-transform  
 4.3.1 power series method  
 4.3.2 partial fraction expansion method  
 4.3.3 residue method  
 4.3.4 comparison of the inverse z-transform methods  
 4.4 properties of the z-transform  
 4.5 some applications of the z-transform in signal processing  
 4.5.1 pole-zero description of discrete-time systems  
 4.5.2 frequency response estimation  
 4.5.3 geometric evaluation of frequency response  
 4.5.4 direct computer evaluation of frequency response  
 4.5.5 frequency response estimation via fft  
 4.5.6 frequency units used in discrete-time systems  
 4.5.7 stability considerations  
 4.5.8 difference equations  
 4.5.9 impulse response estimation  
 4.5.10 applications in digital filter design  
 4.5.11 realization structures for digital filters  
 4.6 summary  
 problems  
 references  
 bibliography  
 appendices  
 4a recursive algorithm for the inverse z-transform  
 4b c program for evaluating the inverse z-transform and for cascade-to-parallel structure conversion  
 4c c program for estimating frequency response  
 4d z-transform operations with matlab references for appendices  
 5 correlation and convolution  
 5.1 introduction  
 5.2 correlation description  
 5.2.1 cross- and autocorrelation  
 5.2.2 applications of correlation  
 5.2.3 fast correlation  
 5.3 convolution description  
 5.3.1 properties of convolution  
 5.3.2 circular convolution  
 5.3.3 system identification  
 5.3.4 deconvolution  
 5.3.5 blind deconvolution  
 5.3.6 fast linear convolution  
 5.3.7 computational advantages of fast linear convolution  
 5.3.8 convolution and correlation by sectioning  
 5.3.9 overlap-add method  
 5.3.10 overlap-save method  
 5.3.11 computational advantages of fast convolution by sectioning  
 5.3.12 the relationship between convolution and correlation  
 5.4 implementation of correlation and convolution  
 5.5 application examples  
 5.5.1 correlation  
 5.5.2 convolution  
 5.6 summary  
 problems  
 references  
 appendix  
 5a c language program for computing cross- and autocorrelation  
 6 a framework for digital filter design  
 6.1 introduction to digital filters  
 6.2 types of digital filters: fir and iir filters  
 6.3

## &lt;&lt;数字信号处理实践方法&gt;&gt;

choosing between fir and iir filters6.4 filter design steps6.4.1 specification of the filter requirements6.4.2 coefficient calculation6.4.3 representation of a filter by a suitable structure (realization)6.4.4 analysis of finite wordlength effects6.4.5 implementation of a filter6.5 illustrative examples6.6 summaryproblemsreferencebibliography7 finite impulse response (fir) filter design7.1 introduction7.1.1 summary of key characteristic features of fir filters7.1.2 linear phase response and its implications7.1.3 types of linear phase fir filters7.2 fir filter design7.3 fir filter specifications7.4 fir coefficient calculation methods7.5 window method7.5.1 some common window functions7.5.2 summary of the window method of calculating fir filter coefficients7.5.3 advantages and disadvantages of the window method7.6 the optimal method7.6.1 basic concepts7.6.2 parameters required to use the optimal program7.6.3 relationships for estimating filter length,  $n$ 7.6.4 summary of procedure for calculating filter coefficients by the optimal method7.6.5 illustrative examples7.7 frequency sampling method7.7.1 nonrecursive frequency sampling filters7.7.2 recursive frequency sampling filters7.7.3 frequency sampling filters with simple coefficients7.7.4 summary of the frequency sampling method7.8 comparison of the window, optimum and frequency sampling methods7.9 special fir filter design topics7.9.1 half-band fir filters7.9.2 frequency transformation7.9.3 computationally efficient fir filters7.10 realization structures for fir filters7.10.1 transversal structure7.10.2 linear phase structure7.10.3 other structures7.10.4 choosing between structures7.11 finite wordlength effects in fir digital filters7.11.1 coefficient quantization errors7.11.2 roundoff errors7.11.3 overfbw errors7.12 fir implementation techniques7.13 design example7.14 summary7.15 application examples of fir filtersproblemsreferencesbibliographyappendices7a c programs for fir filter design7b fir filter design with matlab8 design of infinite impulse response (iir) digital filters8.1 introduction: summary of the basic features of iir filters8.2 design stages for digital iir filters8.3 performance specification8.4 coefficient calculation methods for iir filters8.5 pole-zero placement method of coefficient calculation8.5.1 basic concepts and illustrative design examples8.6 impulse invariant method of coefficient calculation8.6.1 basic concepts and illustrative design examples8.6.2 summary of the impulse invariant method8.6.3 remarks on the impulse invariant method8.7 matched z-transform (mzt) method of coefficient calculation8.7.1 basic concepts and illustrative design examples8.7.2 summary of the matched z-transform method8.7.3 remarks on the matched z-transform method8.8 bilinear z-transform (bzt) method of coefficient calculation8.8.1 basic concepts and illustrative design examples8.8.2 summary of the bzt method of coefficient calculation8.8.3 comments on the bilinear transformation method8.9 use of bzt and classical analog filters to design iir filters8.9.1 characteristic features of classical analog filters8.9.2 the bzt methodology using classical analog filters8.9.3 illustrative design examples (lowpass, highpass, bandpass and bandstop filters)8.10 calculating iir filter coefficients by mapping s-plane poles and zeros8.10.1 basic concepts8.10.2 illustrative examples8.11 using iir filter design programs8.12 choice of coefficient calculation methods for iir filters8.12.1 nyquist effect8.13 realization structures for iir digital filters8.13.1 practical building blocks for iir filters8.13.2 cascade and parallel realization structures for higher-order iir filters8.14 finite wordlength effects in iir filters8.14.1 coefficient quantization errors8.15 implementation of iir filters8.16 a detailed design example of an iir digital filter8.17 summary8.18 application examples in digital audio and instrumentation8.18.1 digital audio8.18.2 digital control8.18.3 digital frequency oscillators8.19 application examples in telecommunication8.19.1 touch-tone generation and reception for digital telephones8.19.2 digital telephony: dual tone multifrequency (dtmf) detection using the goertzel algorithm8.19.3 clock recovery for data communicationproblemsreferencesbibliographyappendices8a c programs for iir digital filter design8b iir filter design with matlab8c evaluation of complex square roots using real arithmetic9 multirate digital signal processing9.1 introduction9.1.1 some current uses of multirate processing in industry9.2 concepts of multirate signal processing9.2.1 sampling rate reduction: decimation by integer factors9.2.2 sampling rate increase: interpolation by integer factors9.2.3 sampling rate conversion by non-integer factors9.2.4 multistage approach to sampling rate conversion9.3 design of practical sampling rate converters9.3.1 filter specification9.3.2 filter requirements for individual stages9.3.3 determining the number of stages and decimation factors9.3.4 illustrative design examples9.4 software implementation of sampling rate converters-decimators9.4.1 program for multistage decimation9.4.2 test example for the decimation program9.5 software implementation of interpolators9.5.1 program for multistage

## &lt;&lt;数字信号处理实践方法&gt;&gt;

interpolation9.5.2 test example9.6 sample rate conversion using polyphase filter structure9.6.1 polyphase implementation of interpolators9.7 application examples9.7.1 high quality analog-to-digital conversion for digital audio9.7.2 efficient digital-to-analog conversion in compact hi-fi systems9.7.3 application in the acquisition of high quality data9.7.4 multirate narrowband digital filtering9.7.5 high resolution narrowband spectral analysis9.8 summaryproblemsreferencesbibliographyappendices9a c programs for multirate processing and systems design9b multirate digital signal processing with matlab10 adaptive digital filters10.1 when to use adaptive filters and where they have been used10.2 concepts of adaptive filtering10.2.1 adaptive filters as a noise canceller10.2.2 other configurations of the adaptive filter10.2.3 main components of the adaptive filter10.2.4 adaptive algorithms10.3 basic wiener filter theory10.4 the basic lms adaptive algorithm10.4.1 implementation of the basic lms algorithm10.4.2 practical limitations of the basic lms algorithm10.4.3 other lms-based algorithms10.5 recursive least squares algorithm10.5.1 recursive least squares algorithm10.5.2 limitations of the recursive least squares algorithm10.5.3 factorization algorithms10.6 application example 1 - adaptive filtering of ocular artefacts from the human eeg10.6.1 the physiological problem10.6.2 artefact processing algorithm10.6.3 real-time implementation10.7 application example 2 - adaptive telephone echo cancellation10.8 other applications10.8.1 loudspeaking telephones10.8.2 multipath compensation10.8.3 adaptive jammer suppression10.8.4 radar signal processing10.8.5 separation of speech signals from background noise10.8.6 fetal monitoring - cancelling of maternal eeg during labourproblemsreferencesbibliographyappendices10a c language programs for adaptive filtering10b matlab programs for adaptive filtering11 spectrum estimation and analysis11.1 introduction11.2 principles of spectrum estimation11.3 traditional methods11.3.1 pitfalls11.3.2 windowing11.3.3 the periodogram method and periodogram properties11.3.4 modified periodogram methods11.3.5 the blackman-tukey method11.3.6 the fast correlation method11.3.7 comparison of the power spectral density estimation methods11.4 modern parametric estimation methods11.5 autoregressive spectrum estimation11.5.1 autoregressive model and filter11.5.2 power spectrum density of ar series11.5.3 computation of model parameters - yule-walker equations11.5.4 solution of the yule-walker equations11.5.5 model order11.6 comparison of estimation methods11.7 application examples11.7.1 use of spectral analysis by a dft for differentiating between brain diseases11.7.2 spectral analysis of eegs using autoregressive modelling11.8 summary11.9 worked exampleproblemsreferencesappendix11a matlab programs for spectrum estimation and analysis12 general- and special-purpose digital signal processors12.1 introduction12.2 computer architectures for signal processing12.2.1 harvard architecture12.2.2 pipelining12.2.3 hardware multiplier-accumulator12.2.4 special instructions12.2.5 replication12.2.6 on-chip memory/cache12.2.7 extended parallelism - simd, vliw and static superscalar processing12.3 general-purpose digital signal processors12.3.1 fixed-point digital signal processors12.3.2 floating-point digital signal processors12.4 selecting digital signal processors12.5 implementation of dsp algorithms on general-purpose digital signal processors12.5.1 fir digital filtering12.5.2 iir digital filtering12.5.3 fft processing12.5.4 multirate processing12.5.5 adaptive filtering12.6 special-purpose dsp hardware12.6.1 hardware digital filters12.6.2 hardware fft processors12.7 summaryproblemsreferencesbibliographyappendix12a tms320 assembly language programs for real-time signal processing and a c language program for constant geometry radix-2 fft13 analysis of finite wordlength effects in fixed-point dsp systems13.1 introduction13.2 dsp arithmetic13.2.1 fixed-point arithmetic13.2.2 floating-point arithmetic13.3 adc quantization noise and signal quality13.4 finite wordlength effects in iir digital filters13.4.1 influence of filter structure on finite wordlength effects13.4.2 coefficient quantization errors in iir digital filters13.4.3 coefficient wordlength requirements for stability and desired frequency response13.4.4 addition overflow errors and their effects13.4.5 principles of scaling13.4.6 scaling in cascade realization13.4.7 scaling in parallel realization13.4.8 output overflow detection and prevention13.4.9 product roundoff errors in iir digital filters13.4.10 effects of roundoff errors on filter performance13.4.11 roundoff noise in cascade and parallel realizations13.4.12 effects of product roundoff noise in modern dsp systems13.4.13 roundoff noise reduction schemes13.4.14 determining practical values for error feedback coefficients13.4.15 limit cycles due to product roundoff errors13.4.16 other nonlinear phenomena13.5 finite wordlength effects in fft algorithms13.5.1 roundoff errors in fft13.5.2 overflow errors and scaling in fft13.5.3 coefficient quantization in fft13.6

<<数字信号处理实践方法>>

summaryproblemsreferencesbibliographyappendices13a finite wordlength analysis program for iir filters13b l2 scaling factor equations14 applications and design studies14.1 evaluation boards for real-time signal processing14.1.1 background14.1.2 tms320c 10 target board14.1.3 dsp56002 evaluation module for real-time dsp14.1.4 tms320c54 and dsp56300 evaluation boards14.2 dsp applications14.2.1 detection of fetal heartbeats during labour14.2.2 adaptive removal of ocular artefacts from human ergs14.2.3 equalization of digital audio signals14.3 design studies14.4 computer-based multiple choice dsp questions14.5 summaryproblemsreferencesbibliographyappendix14a the modified ud factorization algorithmindex



## <<数字信号处理实践方法>>

### 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。

<<数字信号处理实践方法>>

版权说明

本站所提供下载的PDF图书仅提供预览和简介，请支持正版图书。

更多资源请访问:<http://www.tushu007.com>