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VLSI 数字信号处理系统 设计与实现

(英文版)

VLSI DIGITAL SIGNAL PROCESSING SYSTEMS

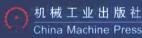
Design and Implementation

Keshab K. Parhi



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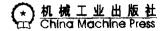
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(美) Keshab K. Parhi 著



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Preface

Digital signal processing (DSP) is used in numerous applications such as video compression, digital set-top box, cable modems, digital versatile disk, portable video systems/computers, digital audio, multimedia and wireless communications, digital radio, digital still and network cameras, speech processing, transmission systems, radar imaging, acoustic beamformers, global positioning systems, and biomedical signal processing. The field of DSP has always been driven by the advances in DSP applications and in scaled very-large-scale-integrated (VLSI) technologies. Therefore, at any given time, DSP applications impose several challenges on the implementations of the DSP systems. These implementations must satisfy the enforced sampling rate constraints of the real-time DSP applications and must require less space and power consumption.

This book addresses the methodologies needed to design custom or semicustom VLSI circuits for these applications. Many of the techniques presented in the book are also applicable for faster implementations using off-the-shelf programmable digital signal processors. This book is intended to be used as a textbook for first-year graduate or senior courses on VLSI DSP architectures, or DSP structures for VLSI or High-Performance VLSI system design. This book is also an excellent reference for those involved in algorithm or architecture or circuit design for DSP applications.

This book brings together the distinct fields of computer architecture theory and DSP. DSP computation is different from general-purpose computation in the sense that the DSP programs are nonterminating programs. In DSP

such as addition and multiplication and digital filters, high-performance VLSI system design issues such as pipelining styles and low-power design, and programmable digital signal processors. Design of adders and multipliers using various implementation styles, such as bit-parallel, bit-serial, and digit-serial, and various number systems such as two's complement, canonic signed digit, and carry-save are discussed in Chapter 13. This chapter also addresses distributed arithmetic. Chapter 14 addresses arithmetic architectures based on redundant or signed-digit implementations. The main advantage of redundant arithmetic lies in its carry-free property, which enables computation in both least significant bit and most significant bit first modes. Conversion from redundant to nonredundant and vice versa is also addressed. In these chapters, bit-serial multipliers are derived from bit-parallel designs by systolic design methodology. Residue arithmetic, which can be used for implementation of FIR digital filters and transforms, is not studied in this book. Chapter 15 presents strength reduction at numerical level to reduce the area and power consumption of two's complement and canonic signed digit number based digital filters. Chapter 16 discusses various pipelining styles, such as synchronous, wave, and asynchronous pipelining. Approaches to reduction of clock skew in synchronous systems and synthesis of interface circuits in asynchronous systems are also addressed. Chapter 17 on low-power design presents various approaches for reduction of power consumption at architectural and technology levels and for estimation of power consumption. Chapter 18 addresses various architectures used in programmable digital signal processors.

Seven appendixes in the book cover shortest path algorithms used for determining the iteration bound and for retiming, scheduling, and allocation techniques used for determining the folding sets for design of folded architectures; Euclid's GCD algorithm, which is used for Winograd's convolution; orthonormality of Schur polynomials used for design of lattice digital filters; fast bit-parallel addition and multiplication; scheduling techniques for bit-serial systems; and coefficient quantization in FIR filters.

The concepts in this book have been described in a technology-independent manner. The examples in this book are based on digital filters and transforms. Many real-time DSP systems make use of control flow constructs such as conditionals, interrupts, and jump. Design of control-dominated DSP systems is beyond the scope of this book. The exercises can be completed using any programming language such as MATLAB or C. Many application-driven problems have been included at the end of the chapters. For example, the problems at the end of the algorithmic strength reduction chapter address the use of fast filters in design of equalizers in communications systems, wavelets, two-dimensional FIR digital filters, and motion estimation. These problems introduce the reader to different applications where the concepts covered in the chapter can be applied.

This book is based on the material taught at the University of Minnesota in two current semester courses: EE 5329: VLSI Digital Signal Processing

on the iteration period of any recursive signal processing algorithm. Two algorithms are described for determining this bound. The next 5 chapters address various transformations for improving performance of digital signal processing implementations. In Chapter 3, the basic concepts of pipelining and parallel processing are reviewed and the use of these techniques in design of high-speed or low-power applications is demonstrated. Chapter 4 addresses the retiming transformation, which is a generalization of the pipelining approach. Chapter 5 addresses unfolding, which can be used to design parallel architectures. Chapters 6 and 7 address folding techniques used to design time-multiplexed architectures where area reduction is important. While Chapter 6 addresses folding of arbitrary data-flow graphs, Chapter 7 addresses folding of regular data-flow graphs based on systolic design methodology.

Chapters 8 to 12 address design of algorithm structures for various DSP algorithms based on algorithm transformations such as strength reduction, look-ahead and relaxed look-ahead, and scaling and roundoff noise in digital filters. Chapter 8 addresses fast convolution based on Cook-Toom and Winograd convolution algorithms. In Chapter 9, algorithmic strength reduction is exploited to reduce the number of multiplication operations in parallel FIR filters, discrete cosine transforms, and parallel rank-order filters. Design of fast Fourier transform (FFT) structures is also based on strength reduction transformations but is not covered in this book since it is covered in many introductory DSP textbooks. While it is easy to achieve pipelining and parallel processing in nonrecursive computations, recursive and adaptive digital filters cannot be easily pipelined or processed in parallel due to the presence of feedback loops. In Chapter 10, the look-ahead technique is discussed and is used to pipeline first-order infinite impulse response (IIR) digital filters. For higher order filters, two types of look-ahead techniques, clustered and scattered look-ahead, are discussed. It is shown that the scattered look-ahead technique guarantees stability in pipelined IIR filters. The parallel implementations of IIR digital filters and how to combine pipelining and parallel processing in these digital filters are also addressed. Adaptive digital filters are pipelined based on relaxed look-ahead, which are based on certain approximations or relaxations of look-ahead. Chapter 11 addresses scaling and roundoff noise, which are important for VLSI implementations of DSP systems using fixed-point arithmetic. Roundoff noise computation techniques cannot be applied to many digital filters. These filters are preprocessed using slowdown, pipelining and/or retiming so that every roundoff noise node can be expressed as a state variable. The direct-form IIR digital filters cannot meet the filter requirements in certain applications. Lattice digital filters may be better suited for these applications due to their excellent roundoff noise property. Chapter 12 presents Schur polynomials, orthonormality of Schur polynomials, and use of these polynomials to design basic (two multiplier and one multiplier), normalized, and scaled-normalized lattice digital filters. Pipelined implementation of these lattice digital filters is also discussed.

Chapters 13 to 18 address VLSI implementations of arithmetic operations

such as addition and multiplication and digital filters, high-performance VLSI system design issues such as pipelining styles and low-power design, and programmable digital signal processors. Design of adders and multipliers using various implementation styles, such as bit-parallel, bit-serial, and digit-serial, and various number systems such as two's complement, canonic signed digit, and carry-save are discussed in Chapter 13. This chapter also addresses distributed arithmetic. Chapter 14 addresses arithmetic architectures based on redundant or signed-digit implementations. The main advantage of redundant arithmetic lies in its carry-free property, which enables computation in both least significant bit and most significant bit first modes. Conversion from redundant to nonredundant and vice versa is also addressed. In these chapters, bit-serial multipliers are derived from bit-parallel designs by systolic design methodology. Residue arithmetic, which can be used for implementation of FIR digital filters and transforms, is not studied in this book. Chapter 15 presents strength reduction at numerical level to reduce the area and power consumption of two's complement and canonic signed digit number based digital filters. Chapter 16 discusses various pipelining styles, such as synchronous, wave, and asynchronous pipelining. Approaches to reduction of clock skew in synchronous systems and synthesis of interface circuits in asynchronous systems are also addressed. Chapter 17 on low-power design presents various approaches for reduction of power consumption at architectural and technology levels and for estimation of power consumption. Chapter 18 addresses various architectures used in programmable digital signal processors.

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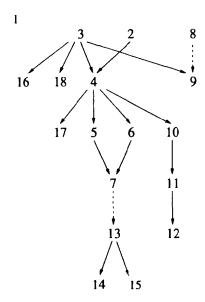


Fig. 0.1 Precedence constraints among different chapters.

Systems and EE 5549: Digital Signal Processing Structures for VLSI. EE 5329 (with a basic course on VLSI Design as prerequisite) covers chapters 2 through 7 and parts of chapters 13 through 18 (in that order). EE 5549 (with a basic course on digital signal processing as prerequisite) covers parts of chapters 2, 3, and 4, chapters 8 through 12, and some architectures for video compression based on journal and conference papers. These two semester courses were taught as three-quarter courses in the past. For a single semester course on VLSI Digital Signal Processing, chapters 2 through 7, parts of chapters 9, 10, 13 and 15, and an overview of topics in chapters 17 and 18 are recommended. However, the instructors can select the chapters that suit their needs.

The chapters need not be followed in the order they are presented. Many chapters can be taught independently. The precedence graph in Fig. 0.1 shows the dependencies among chapters. The dashed lines represent weak dependencies where a section of the current chapter is dependent on the preceding chapter.

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1

$\begin{array}{c} Introduction\ to\ Digital\\ Signal\ Processing\\ Systems \end{array}$

1.1 INTRODUCTION

Digital signal processing (DSP) has many advantages over analog signal processing. Digital signals are more robust than analog signals with respect to temperature and process variations. The accuracy in digital representations can be controlled better by changing the wordlength of the signal. Furthermore, DSP techniques can cancel the noise and interference while amplifying the signal. In contrast, both signal and noise are amplified in analog signal processing. Digital signals can be stored and recovered, transmitted and received, processed and manipulated, all virtually without error. While analog signal processing is indispensable for systems that require extremely high frequencies such as the radio frequency transceiver in wireless communications, or extremely low area and low power such as micro machine sensors used to detect cracks and other stress-related material defects, many complex systems are realized digitally with high precision, high signal to noise ratio (SNR), repeatability, and flexibility.

DSP systems can be realized using programmable processors or custom designed hardware circuits fabricated using very-large-scale-integrated (VLSI) circuit technology. The goal of digital design is to maximize the performance while keeping the cost down. In the context of general digital design, performance is measured in terms of the amount of hardware circuitry and resources required (i.e., space or area); the speed of execution, which depends on both throughput and clock rate; and the amount of power dissipation or total energy required to perform a given task. For fixed-point DSP systems, the finite wordlength performance (i.e., quantization and roundoff noise) is the fourth