Digital Signal Processing

Digital Signal Processing

Concepts and Applications

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Preface

Digital signal processing (DSP) provides a rapidly advancing portfolio of filtering and estimating techniques or algorithms which are used in signal analysis and processing. Significant current applications are in the development of mobile communications equipment, particularly for personal use and design of sophisticated radar systems. The aim of this book is to provide an introduction to the fundamental DSP operations of filtering, estimation and signal analysis as used in signal processing.

Most of the chapters include substantive numerical examples to illustrate the material developed, as well as self assessment questions which have been designed to help readers aid their comprehension of this material. All the chapters conclude with further problem questions for the student.

Chapters 1 to 3 cover basic analogue signal theory as a prerequisite to this DSP text. Chapter 4 extends these concepts to sampled-data systems and here discrete convolution is introduced. Chapters 5 and 6 explore digital filters, both infinite and finite impulse response, which implement the convolution operation and include both analytical design and software optimisation techniques. The two chapters conclude with a brief discussion on the problems of finite precision arithmetic. Chapters 7 and 8 introduce, at a more mathematical level, the concept of random signals, correlation and spectral density. Chapter 8 covers adaptive or self learning filters which alter their characteristics dependent on the signal scenario which is present. They find widespread application in communications systems as equalisers and echo cancellers. The final three chapters deal with spectral analysis techniques. Chapter 9 covers the discrete Fourier transform (DFT), its derivation and the design of DFT processors. This chapter then investigates the application of DFT processors in classical spectrum analysis equipment before introducing the modern analysers which are based on the adaptive filter technique of Chapter 8. Chapter 10 deals with the fast Fourier transform which is the most widely applied implementation of the DFT processing function. Chapter 11 introduces multirate techniques to extend the capabilities of these analysers to speech and image processing applications.

With this balance between signal theory, processor design and systems applications we hope that this text will be useful both in academia and in the rapidly growing commercial signal processing community. Advanced DSP is of fundamental importance in implementing many high performance systems such as personal communications and radar.

To aid the class instructor, the authors can provide a printed set of outline solutions to the end-of-chapter problems and these are available via the WWW on password access. MATLABTM source code is also provided on an open access basis to assist the

instructor with presentation of the material and the student in understanding of the material. In general the source code provides a computer animation of some of the figures in the book. For example the m-file "fig1_4.m" contains MATLAB code which produces an animation of the complex phasor of Figure 1.4. These are identified within the text by the □ symbol. These software and solutions to problems are available via the menu at the Edinburgh WWW server address: <http://www.ee.ed.ac.uk/~pmg/SIGPRO/index.html>. Subsequent corrections to this text will also be available at the same WWW address.

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Abbreviations

AC	Alternating current (implying sinusoidal signal)
ACF	Autocorrelation function
A/D	Analogue to digital (converter)
ADPCM	Adaptive DPCM
AGC	Automatic gain control
AM	Amplitude modulation
AR	Autoregressive
BLMS	Block least mean squares
BP	Bandpass
BPF	Bandpass filter
BS	Bandstop
CD	Compact disc
CDMA	Code division multiple access
CELP	Codebook of excited linear prediction
CMOS	Complementary metal oxide silicon (transistor)
COFDM	Coded orthogonal frequency-division multiplex
D/A	Digital to analogue (converter)
DC	Direct current (implying a 0 Hz component)
DCT	Discrete cosine transform
DFS	Discrete Fourier series
DFT	Discrete Fourier transform
DIF	Decimation in frequency
DIT	Decimation in time
DM	Delta modulation
DPCM	Differential pulse code modulation
DPSK	Differential phase shift keying
DSB	Double sideband
DSP	Digital signal processing
DTFT	Discrete-time Fourier transform
ESD	Energy spectral density

EVR	Eigenvalue ratio
FFT	Fast Fourier transform
FIR	Finite impulse response
FM	Frequency modulation
FS	Fourier series; Federal Standard
FSK	Frequency shift keying
FT	Fourier transform
HP	High pass
I	Imaginary (quadrature signal) component
IDFT	Inverse DFT
IF	Intermediate frequency
IIR	Infinite impulse response
ISI	Inter-symbol interference
KCL	Kirchhoff's current laws
LMS	Least mean squares
LO	Local oscillator
LOS	Line of sight
LP	Low pass
LPC	Linear predictive coding
LS	Least squares
LTI	Linear time invariant
MA	Moving average
MAC	Multiply and accumulate
MATLAB	MATrix LABoratory commercial DSP software product
MFSK	Multiple frequency shift keying
MMSE	Minimum mean square error
MODEM	Modulator/demodulator
MOS	Mean opinion score (for speech quality assessment)
	Metal oxide silicon (transistor)
MPE	Multipulse excitation
MSE	Mean square error
NATO	North Atlantic Treaty Organisation
NPSD	Noise power spectral density
PAM	Pulse amplitude modulation
PCM	Pulse code modulation
pdf	Probability density function
PFE	Partial fraction expansion

PLL	Phase locked loop
PM	Phase modulation
PN	Pseudo-noise
PO	Percentage overshoot
PPM	Pulse position modulation
PR	Perfect reconstruction
PSD	Power spectral density
PSK	Phase shift keying
PWM	Pulse width modulation
Q	Quantiser
R	Real (in-phase signal) component
RAM	Random access memory
ROM	Read only memory
RLS	Recursive least squares
RMS	Root mean square
SAQ	Self assessment question
SBC	Sub-band coder
SG	Stochastic gradient
S/H	Sample and hold
SNR	Signal-to-noise ratio
TDM	Time division multiplex
THD	Total harmonic distortion
VLSI	Very large scale integrated (circuit)
WWW	World Wide Web
ZOH	Zero order hold

Principal symbols

$\mathbf{a}$	tap weight vector
a_i	digital filter weight coefficient value for tap i
A	A-law PCM compander constant
A_n	n th trigonometric Fourier component
$A(k)$	k th real Fourier coefficient
b_i	digital filter recursive weight coefficient value for tap i
B	signal bandwidth in Hz
B_n	n th trigonometric Fourier component
$B(k)$	k th imaginary (real valued) Fourier coefficient
c_n	FIR filter coefficient value
c'_n	windowed coefficient value
C	constant, capacitance (Farads)
$C_n(\omega)$	Chebyshev polynomial of order n
d	lag
$\mathbf{e}$	error vector
$e(n)$	scalar error signal at data sample n
E	energy of signal $x(t)$
f_0	centre frequency
f_1	passband cut-off frequency
f_2	stopband edge frequency
f_{3dB}	half power bandwidth
f_b	bit rate
f_c	centre frequency
f_H	highest frequency component
f_L	lowest frequency component
f_{LO}	local oscillator frequency
f_s	sample frequency (in Hz)
$F_m(z)$	z -plane reconstruction filter m response

F_N	speech formant N frequency component
$F(\omega)$	frequency response
$g(t)$	baseband signal
$\tilde{g}(kT_s)$	estimate of $g(t)$
$\hat{g}(kT_s)$	prediction of $g(t)$
G	amplifier gain
G_p	processing gain
$\mathbf{h}(n)$	filter impulse response vector
$h(n)$	filter impulse response for sample value n
$h(t)$	filter impulse response
$H_n/H(s)$	system transfer function
$H_A(\omega)$	real frequency response
$H_D(\omega)$	required or desired frequency response
$H(s)$	Laplace transfer function
$H(z)$	z-transfer function
$H(\omega)$	angular frequency response
$I_0[\]$	modified Bessel function of first kind and order zero
k	Boltzmann's constant
K	constant
L	upsampling ratio
$L(\omega)$	weighting function
M	number of feedback taps in an IIR filter; downsampling ratio
n	type of semiconductor material
N	order of DFT
N	number of feedforward taps in an IIR/FIR filter
N_0	noise power spectral density
p	type of semiconductor material
p_1	filter pole number 1
$p(v)$	probability density function of variable v
P	power
q	quantisation step size
Q	quality factor; f_H/B
$\mathbf{r}_{yx}$	cross-correlation vector
$\mathbf{R}_{yy}$	correlation matrix
R	resistance in ohms

R_D	dynamic range
RC	resistor–capacitor time constant τ_c
s	Laplace variable
$S_{xx}(\omega)$	power spectral density
t	time
Δt	sample period
$\mathbf{v}$	eigenvector of autocorrelation matrix
$v(t)$	information signal
$v_i(t)$	voltage waveform of symbol i
w_n	FIR filter/DFT window or weight coefficient n
W_N^k	k th value of the N roots of unity
$W_T(\omega)$	DFT of window function
$\mathbf{x}$	vector x
$x(n)$	sampled input signal
$\hat{x}(n)$	estimate of input signal
X_n	complex Fourier coefficient
$X(f/\omega)$	Fourier (voltage) spectrum of $x(n)$
$ X(f/\omega) $	Fourier amplitude spectrum
$X(k)$	DFT output value for bin or sample number k
$X(s)$	Laplace transform of $x(t)$
$y(n)$	processed output signal
$y_D(n)$	decimated signal
$y_I(n)$	interpolated signal
z_n	filter zero number n
α	attenuation; forgetting factor or window taper
β	constant
δ	Dirac delta
$\delta(t)$	impulse function
$\delta_T(t)$	impulse train
ΔV	voltage difference
∇	gradient vector
ε	error voltage; Chebyshev design parameter

ε_q	quantisation error
ε_q^2	mean-square quantisation error
ξ	filter damping factor
$\xi(n)$	MSE cost function
η	efficiency
$\eta(n)$	additive noise signal
λ_i	eigenvalue i
μ	step-size scaling constant
ρ	normalised correlation coefficient; sum of squared error cost function
$\rho(n)$	vector norm or deviation
σ	real part of Laplacian; standard deviation
σ_x^2	variance of signal x
τ	time constant
$\phi_{xx}(\omega)$	autocorrelation sequence $x(n)$
$\Phi_{xx}(\omega)$	autocorrelation matrix for $x(n)$
$\Phi_{xy}(\omega)$	cross-correlation vector
ω_0	filter centre frequency
ω_a	analogue filter cut-off frequency
ω_{cl}	lower bandpass/bandstop cut-off frequency
ω_{cu}	upper bandpass/bandstop cut-off frequency
ω_d	digital filter cut-off frequency
ω_s	(angular) sample frequency (in rad/s)
$\Delta\omega$	filter transition bandwidth, DFT bin spacing

Special functions

$E[\]$	statistical expectation operator
$[\ , \]$	scalar product
$\langle \ \rangle$	time average
$*$	convolution operation
a^*	complex conjugate of a
$F[\ , \]$	Fourier transform operator
$\Im$	operator corresponding to imaginary part of ...
$\int (.)$	integer part of ...
$L[\ , \]$	Laplace transform operator
$\Re$	operator corresponding to real part of ...
$\text{sa} (x)$	sampling function
$\text{sgn} (x)$	signum (sign) function
$\text{sinc} (x)$	sinc function
$u(t)$	unit step function
Z^{-1}	inverse z transform
Z	z transform

Introduction

Real life signals are generally continuous in time and analogue, i.e. they exist at all time instances and can assume any value, within a predefined range, at these time instances. There are many kinds of analogue signals appearing in nature:

- Electrical signals: voltages, currents, electric and magnetic fields.
- Acoustic signals: mechanical vibrations, sound waves.
- Mechanical signals: displacements, angular motion, velocities, forces, moments, pressures.

Acoustic signals such as sound waves are converted to electrical voltages or currents by sensors or transducers, (i.e. a microphone) in order for them to be processed in an electronic system. Analogue processing involves linear operations such as amplification, filtering, integration and differentiation, as well as various forms of nonlinear processing such as squaring or rectification. This text does not cover the field of nonlinear signal processing, but, in practice, saturation in amplifiers or mixing of signals often introduces nonlinearities. Limitations of practical analogue processing operations are:

- Restricted accuracy.
- Sensitivity to noise.
- Restricted dynamic range.
- Poor repeatability due to component variations with time, temperature, etc.
- Inflexibility to alter or adjust the processing functions.
- Problem in implementing accurate nonlinear and time-varying operations.
- Limited speed of operation.
- High cost of storage for analogue waveforms.

However, in the 1970s, analogue signal processing in the form of surface acoustic wave and charge-coupled sampled-data devices delivered very sophisticated matched filter and correlator parts which were widely used in military equipment. At this time it was not possible to match the analogue speed/performance capability with digital devices so the accuracy of analogue processors was severely limited. The initial implementations of Dolby noise reduction systems also employed analogue filter techniques and the early V.21–V.29 data modems, for transmission of digital data over telephone lines, used exclusively analogue filters and modulators.

Digital signal processing (DSP) is achieved by sampling the analogue signal at regular intervals and representing each of these sample values with a binary number. These are subsequently passed to a specialised digital processor to perform numerical or computational operations on these signals. Operations in digital signal processing systems encompass additions, multiplications, data transfers, logical operations and can be extended to implementation of complex matrix manipulations. The essential operations of many DSP systems include:

- Converting analogue signals into a sequence of digital binary numbers, which requires both sampling and analogue-to-digital (A/D) conversion.
- Performing numerical manipulations, predominantly multiplications, on the digital information data stream.
- Converting the digital information back to analogue signal, by digital-to-analogue (D/A) conversion and filtering.

The basic components of the digital processor, which is equivalent to the analogue processing function, is considered later in Chapter 4. The DSP function is generally described as an algorithm or program which defines the in-built arithmetic operations. DSP is in fact an extension of the conventional microprocessor function except that a fast multiplier is added as a hardware accelerator element.

The main attraction of digital processors is that their accuracy, which is controlled by the quantisation step size or word length employed in the A/D converter, is extendable only at the cost of greater complexity for the ensuing processing operations, i.e. additions and multiplications. However, floating-point number representations permit the accuracy to be maintained over a wide dynamic range. Further, digital processors are generally repeatable and much less sensitive to noise than analogue processors as they are dealing with binary symbols and the processed outputs must always possess binary values. The microelectronics industry has been reducing continuously silicon VLSI circuit geometrics and hence improving the speed, complexity and storage capacity. The feature size in the 1970s was $5\mu\text{m}$ while in the year 2000 we will be designing VLSI circuits with $0.18\mu\text{m}$ feature sizes and single chip complexities of 40 M individual transistors, and by the year 2010 $0.7\mu\text{m}$ feature size circuits are predicted to have 3 GHz clock rates.

VLSI permits the design of application specific integrated circuits with exceptional cost/performance capabilities to execute sophisticated DSP processing functions. Current developments in microelectronics are delivering an order of magnitude increase in processor operating speed, coupled with a 30-fold reduction in power consumption, every eight years.

Some of the key benefits which derive from digital signal processing are:

- Efficient implementation of linear phase filters for communication and radar receiver design.
- Easy realisation of Fourier transform processors for signal analysis.
- Possibility of implementing complicated processing functions involving matrix manipulations.

These advances in DSP, complexity and functionality over analogue signal processing are not obtained without some penalties. High accuracy, high speed A/D and D/A conversion hardware is expensive and it can lead to noise and distortion problems. We always require bandlimiting filters before the sampling function and this introduces some loss of information. For some applications, e.g. RF signals, digital processors cannot achieve the necessary high speed sampling requirements. Even with these drawbacks the use of DSP is becoming ubiquitous, partly because of the 40% per year cumulative growth of the DSP since 1988, with predicted market sales of 6,000,000,000 dollars in 2000. The generic application areas are:

Speech and audio: noise reduction (Dolby), coding, compression (MPEG), recognition, speech synthesis.

Music: recording, playback and mixing, synthesis of digital music, CD players.

Telephony: speech, data and video transmission by wire, radio or optical fibre.

Radio: digital modulators and modems for cellular telephony.

Signal analysis: spectrum estimation, parameter estimation, signal modelling and classification.

Instrumentation: signal generation, filtering, signal parameter measurement.

Image processing: 2-D filtering, enhancement, coding, compression, pattern recognition.

Multimedia: generation, storage and transmission of sound, motion pictures, digital TV, HDTV, DVD, MPEG, video conferencing, satellite TV.

Radar: filtering, target detection, position and velocity estimation, tracking, imaging, direction finding, identification.

Sonar: as for radar but also for use in acoustic media such as the sea.

Control: servomechanisms, automatic pilots, chemical plant control.

Biomedical: analysis, diagnosis, patient monitoring, preventive health care, telemedicine.

Transport: vehicle control (braking, engine management) and vehicle speed measurement.

Navigation: Accurate position determination, global positioning, map display.

The key attraction of DSP devices is that their in-built programmability allows one standard part to be applied to most of these processing functions by changing the stored program or instruction set.

In terms of specific uses, the modern car has many tens of processors attached to sensors for fuel injection, engine management, passenger compartment climate control, braking system protection, detection of component failure, etc. In fact a major part of the modern automobile comprises electronics rather than mechanical engineering. Modern mobile and cellular telephone systems rely on advanced signal processing to detect and decode the received signals, to minimise received errors, to control the mobile transmissions, to enhance battery life, etc. The TV set and new set-top boxes contain a large number of DSP chips to detect the low power signals received and to decompress the video traffic efficiently.

In the late 1990s there are a tremendous number of applications for DSP and, in the future, we confidently expect these to increase as the continuous growth in processor power and novel algorithm development unlocks even more sophisticated processing algorithms which cannot be contemplated today. DSP is thus now firmly entrenched in the consumer marketplace.