Digital Signal Processing
Digital Signal Processing

An Introduction with MATLAB and Applications
This book is dedicated to our loving families.
Signal Processing (SP) is a subject of central importance in engineering and the applied sciences. Signals are information-bearing functions, and SP deals with the analysis and processing of signals (by dedicated systems) to extract or modify information. Signal processing is necessary because signals normally contain information that is not readily usable or understandable, or which might be disturbed by unwanted sources such as noise. Although many signals are non-electrical, it is common to convert them into electrical signals for processing. Most natural signals (such as acoustic and biomedical signals) are continuous functions of time, with these signals being referred to as analog signals. Prior to the onset of digital computers, Analog Signal Processing (ASP) and analog systems were the only tools to deal with analog signals. Although ASP and analog systems are still widely used, Digital Signal Processing (DSP) and digital systems are attracting more attention, due in large part to the significant advantages of digital systems over their analog counterparts. These advantages include superiority in performance, speed, reliability, efficiency of storage, size and cost. In addition, DSP can solve problems that cannot be solved using ASP, like the spectral analysis of multicomponent signals, adaptive filtering, and operations at very low frequencies.

Following the recent developments in engineering which occurred in the 1980s and 1990s, DSP became one of the world’s fastest growing industries. Since that time DSP has not only impacted on traditional areas of electrical engineering, but has had far reaching effects on other domains that deal with information such as economics, meteorology, seismology, bioengineering, oceanology, communications, astronomy, radar engineering, control engineering and various other applications.

This book is based on the Lecture Notes of Associate Professor Zahir M. Hussain at RMIT University (Melbourne, 2001–2009), the research of Dr. Amin Z. Sadik (at QUT & RMIT, 2005–2008), and the Notes of Professor Peter O'Shea at Queensland University of Technology.

Part I of the book addresses the representation of analog and digital signals and systems in the time domain and in the frequency domain. The core topics covered are convolution, transforms (Fourier, Laplace, Z, Discrete-time Fourier, and
Discrete Fourier), filters, and random signal analysis. There is also a treatment of some important applications of DSP, including signal detection in noise, radar range estimation for airborne targets, binary communication systems, channel estimation, banking and financial applications, and audio effects production. Design and implementation of digital systems (such as integrators, differentiators, resonators and oscillators are also considered, along with the design of conventional digital filters. Part I is suitable for an elementary course in DSP.

Part II (which is suitable for an advanced signal processing course), considers selected signal processing systems and techniques. Core topics covered are the Hilbert transformer, binary signal transmission, phase-locked loops, sigma–delta modulation, noise shaping, quantization, adaptive filters, and non-stationary signal analysis.

Part III presents some selected advanced DSP topics.

We hope that this book will contribute to the advancement of engineering education and that it will serve as a general reference book on digital signal processing.

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Prerequisites:
Basic knowledge in calculus, programming, and circuit theory is recommended.

Objectives:
The book aims to facilitate the development of expertise in analyzing and synthesizing signals, both natural and synthetic. It provides various tools which can reveal the critical information contained in the time and frequency structure of signals of interest. The book also provides advanced applications and topics in signal processing, with MATLAB experiments to give practical experience in implementing analog and digital signal processing systems.

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## Authors’ Biographies
Acronyms, Symbols and Abbreviations

\( \forall n \) For all \( n \)

\( C_n \) \textit{nth-Order Chebychev polynomial}

\( \delta(t) \) Dirac delta function

\( \mathcal{E} \) Expectation functional

\( \mathcal{F} \) Fourier transform (FT)

\( G_m \) Maximum filter gain

\( G_c \) Filter gain at cutoff

\( G_o \) Filter gain at DC

\( \mathcal{H} \) Hilbert transform

\( \mathcal{L} \) Laplace transform (LT)

\( \mathcal{L}_d \) Double Laplace Transform (DLT)

\( \rho \) Time–frequency distribution (TFD)

\( h(n) \) Vector of an FIR impulse response coefficients at sample time \( n \)

\( w(n) \) Vector of an adaptive FIR filter coefficients at sample time \( n \)

\( \omega_c \) Filter cutoff (radian) frequency

\( z^* \) Complex conjugate of \( z \)

\( Z \) \textit{z-Transform}

AC Alternating current

ADC (or A/D) Analog-to-digital converter

a.k.a. Also known as

AM Amplitude modulation

ASP Analog signal processing

AWGN Additive White Gaussian noise

BIBO Bounded-input bounded-output

B-BPF Butterworth band-pass filter

bps Bits Per second

B-HPF Butterworth high-pass filter
<table>
<thead>
<tr>
<th>Acronym</th>
<th>Description</th>
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<tbody>
<tr>
<td>B-LPF</td>
<td>Butterworth low-pass filter</td>
</tr>
<tr>
<td>BPF</td>
<td>Band-pass filter</td>
</tr>
<tr>
<td>BP</td>
<td>Band-pass</td>
</tr>
<tr>
<td>BS</td>
<td>Band-stop</td>
</tr>
<tr>
<td>BSF</td>
<td>Band-stop filter</td>
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<tr>
<td>BW</td>
<td>Bandwidth</td>
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<tr>
<td>CFS</td>
<td>Complex Fourier series</td>
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<tr>
<td>C-BPF</td>
<td>Chebychev band-pass filter</td>
</tr>
<tr>
<td>C-HPF</td>
<td>Chebychev high-pass filter</td>
</tr>
<tr>
<td>C-LPF</td>
<td>Chebychev low-pass filter</td>
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<tr>
<td>D/A (or DAC)</td>
<td>Digital to analog converter</td>
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<tr>
<td>DC (or DC)</td>
<td>Direct current</td>
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<td>DCO</td>
<td>Digital controlled oscillator</td>
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<tr>
<td>DCT</td>
<td>Discrete cosine transform</td>
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<tr>
<td>DFS</td>
<td>Discrete Fourier series</td>
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<tr>
<td>DFT</td>
<td>Discrete FT (finite length N)</td>
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<tr>
<td>DLT</td>
<td>Double-sided Laplace transform</td>
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<tr>
<td>DM</td>
<td>Delta modulator</td>
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<tr>
<td>DSB/DSBTC</td>
<td>Double side-band transmitted carrier</td>
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<tr>
<td>DSP</td>
<td>Digital signal processing</td>
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<tr>
<td>DTFT</td>
<td>Discrete-time FT</td>
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<tr>
<td>ECG</td>
<td>Electrocardiogram</td>
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<td>EEG</td>
<td>Electroencephalogram</td>
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<td>EM</td>
<td>Electromagnetic</td>
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<td>EOG</td>
<td>Electrooculogram</td>
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<tr>
<td>ESD</td>
<td>Energy spectral density</td>
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<tr>
<td>FDM</td>
<td>Frequency division multiplexing</td>
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<tr>
<td>FFT</td>
<td>Fast FT (algorithm to compute DFT)</td>
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<tr>
<td>FIR</td>
<td>Finite impulse response</td>
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<tr>
<td>FM</td>
<td>Frequency modulation</td>
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<td>FT</td>
<td>Fourier transform</td>
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<td>Fourier series</td>
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<td>High-pass</td>
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<td>HPF</td>
<td>High-pass filter</td>
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<td>Hilbert transform</td>
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<td>Hertz</td>
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<tr>
<td>IDFT</td>
<td>Inverse discrete FT</td>
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<tr>
<td>IFFT</td>
<td>Inverse fast FT (algorithm to compute IDFT)</td>
</tr>
<tr>
<td>IIR</td>
<td>Infinite impulse response</td>
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<tr>
<td>I/O</td>
<td>Input/output</td>
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<tr>
<td>ILT</td>
<td>Inverse Laplace transform</td>
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<tr>
<td>ISI</td>
<td>Inter-symbol interference</td>
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<tr>
<td>IZT</td>
<td>Inverse z-transform</td>
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<td>LHS</td>
<td>Left-hand side</td>
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<tr>
<td>LMS</td>
<td>Least mean-square</td>
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LP Low-pass
LPF Low-pass filter
LSB Lower sideband
LT Laplace transform
LTI Linear time-invariant
MF Matched filter
mse Mean-square error
Mux Multiplexer
NBC Natural binary code
PAM Pulse–amplitude modulation
PCM Pulse code modulation
PDF Probability density function
PLL, DPLL Phase-locked loop, digital PLL
PM Phase modulation
P/S Parallel-to-serial converter
PSD Power spectral density
RF Radio frequency
RHS Right-hand side
ROC Region of convergence
Rx Receiver
SDM Sigma–delta modulator
SH (or S/H) Sample-and-hold
SLT Single-sided Laplace transform
SNR Signal-to-noise ratio
snr Signal to noise ratio = $E_b = N_o$ (digital comms)
SP Signal processing
S/P Serial-to-parallel converter
sps Symbol per second
SSBSC Single side-band suppressed carrier
TDM Time division multiplexing
TFD Time–frequency distribution
TFS Trigonometric Fourier series
Tx Transmitter
USB Upper sideband
Var Variance
VCO Voltage-controlled oscillator
WKT Wiener–Kinchin theorem
w.r.t With respect to
ZT z-Transform