

# Index

## A

- Adaptive filters
  - adaptive FIR filter structure, 696
  - adaptive IIR filter structure, 695
- Adaptive algorithms
  - LMS algorithm, 696, 700, 714
  - RLS algorithm, 698, 700, 701
- All-Pass and Minimum-Phase Decomposition, 152
- Analog lowpass filter design
  - analog filter design, 291
    - bandpass, 261, 263, 264, 271
    - bandstop, 261, 265, 266, 271
    - highpass, 261, 262, 264, 271
  - analog filter types comparison, 258
  - Bessel filter, 258, 260
  - Butterworth analog lowpass filter, 242, 246
  - Chebyshev analog lowpass filter, 247, 251
    - type 1 Chebyshev lowpass filter, 247, 250, 251, 297
    - type 2 Chebyshev filter, 252, 255, 263, 264, 266, 267
  - elliptic analog lowpass filter, 255, 256
  - filter specifications, 242, 261, 267
  - transformations, 261, 267
    - lowpass to bandpass, 261, 263, 285, 286, 293
    - lowpass to bandstop, 261, 265, 289, 290
    - lowpass to highpass, 261, 262
    - lowpass to lowpass, 261
- Application areas of DSP, 5
- Application examples
  - analog voice privacy system, 614
  - artificial reverberations, 312, 323
  - audio processing, 310

- compact disc player, 570
- denoising of a speech signal, 232
- detection of signals, 231
- digital audio system, 569
- DTMF tone detection, 233, 236, 237, 316
- hum suppression, 310
- peaking equalizers, 314, 315
- subband coding, 614
- transmultiplexers, 613
- Application examples of DSP
  - audio processing, 8
  - ECG signal processing, 8
  - GPS signal processing, 11
    - GPS positioning, 11
    - location-based mobile emergency services, 12
  - genomic signal processing, 6, 13
  - image processing, 6, 9, 17
    - image compression, 6, 10, 17
    - image restoration and enhancement, 10
    - medical imaging, 6, 9
  - noise reduction in speech signals, 7
  - telecommunications, 6
    - compression, 5, 6, 8, 10, 17
    - echo control, 6
- Applications, 702
- Arithmetic format: fixed or floating point, 757

## B

- Basic sequences
  - arbitrary, 25, 35, 40, 59
  - exponential and sinusoidal, 24
  - unit sample, 23, 25, 39, 40
  - unit step, 23, 24, 40
- BIBO stability theorem, 142

- Biorthogonal scaling functions and wavelets
  - generation, 653
  - wavelet filter coefficients, 656
    - Impact of Wavelet Properties, 660
    - 1-D DWT and Inverse 1-D DWT, 660, 665
    - computation, 660-663, 664
    - 2-D DWT and inverse 2-D DWT, 665, 667
    - computation using matlab, 651
- C**
- Causal and stable conditions, 141, 142
- Causality theorem, 142
- Choice of DSP processor, 757
- Circulant matrix method, 188
- Computation of circular convolution, 188
- Comparison of computational complexity, 220
- Computation of DFT and IDFT using MATLAB, 230
- Continuous signal, 19
- Conventional DSP processors, 753, 754
- Correlation of discrete-time signals, 95
- Cost function, 693, 695
- D**
- Data word size, 757
- Daubechies orthonormal scaling functions
  - generation, 642, 643, 645, 647, 649, 651-653
- Daubechies wavelets
  - decomposition filters ( $H_0, H_1$ ), 642
  - D4 filter, 647
  - D8 filter, 647-649
  - generation using Matlab, 651
  - generation, 659
  - reconstruction filters ( $G_0, G_1$ ) relation, 642
  - symmlets, 652
  - wavelet filter coefficients, 644, 656, 684
    - coefficient domain solutions, 644
    - frequency domain solutions, 645
- DFT approach, 189, 191
- Digital filter bank
  - analysis filter, 575
  - synthesis filter, 575
- Digital filter design
  - digital to digital transformations, 292
  - lowpass to highpass transformation, 393, 395, 302
  - lowpass to lowpass transformation, 292, 300, 301
- Digital filter design from analog filters
  - bilinear transformation, 271-275, 277, 283, 285, 287, 288, 290, 303, 304, 321-323
    - warping effect, 273
  - digital filter specifications, 267, 273
  - impulse-invariant method, 268, 270, 271, 275, 321, 322
- Digital image water marking
  - embedding, 680-682
  - extraction, 681, 682
- Digital signal processing
  - digital signals, 2
  - finite word length, 3
- Discrete Fourier transform
  - twiddle factors, 169, 202, 212, 218, 228
  - circular operations on finite length
    - sequence, 173
  - circular convolution, 174, 175
  - circular correlation, 175
  - circular shift, 173, 174
  - circular time-reversal, 174
  - properties of DFT, 165, 176, 181
  - circular correlation, 179, 181
  - circular frequency shifting, 178, 181
  - circular time shifting, 177, 181
  - linearity, 176, 181
  - multiplication, 180, 181
  - N-point circular convolution, 175, 181
  - Parseval's theorem, 180, 181
  - periodicity, 181
  - time reversal, 176, 181
- Discrete-time Fourier series
  - Fourier coefficients, 164, 167
  - multiplication, 165
  - periodic Convolution, 165
  - symmetry properties, 165
- Discrete time LTI systems in z-domain, 136
- Discrete-time random signals
  - LTI system output for white noise input, 93
  - power of white noise input, 92
  - statistical properties of discrete-time random signals, 91
- Discrete-time signal, 19, 20, 25, 58, 95
- Discrete time signals classification
  - energy and power signals, 27
  - examples, 80
  - finite and infinite length, 26
  - periodic and aperiodic, 27
  - right-sided and left-sided, 26
  - symmetric and antisymmetric, 25
- Discrete-time system characterisation
  - non-recursive difference equation, 50, 51
  - recursive difference equation, 50, 51, 78
- Discrete-time systems
  - classification, 25, 35
    - casual
    - examples, 80

- linear, 19, 24, 33, 35, 36, 38–40, 42, 43, 50–52, 78, 81, 82, 97, 100
- stable, 38, 39, 49, 50, 66, 67, 97
- time-invariant, 19, 24, 36, 37, 40, 51
- impulse and step responses, 39, 57
- DWT
  - wavelet coefficients computation, 629, 630, 634, 662, 676, 677
- E**
- Effect of FIR filtering on voice signals
  - FIR notch filter, 404–408, 414
  - FIR null filter, 403–406, 413
- Elementary operations on sequences
  - addition, 21, 50, 95
  - multiplication, 21, 30, 84
  - scalar multiplication, 21
  - shifting, 21, 22, 43, 59, 62, 67, 77, 84, 89, 100
  - time reversal, 21, 22, 59, 62
- Evolution of DSP processors
  - Harvard architecture, 753, 754, 755
  - Von Neuman architecture, 753, 754
- F**
- Fast Fourier transform
  - chirp transform, 221, 224, 225
  - decimation-in-frequency, 210
    - radix-2 FFT algorithm, 210
  - decimation-in time, 198, 210, 225
    - composite number, 225, 240
    - radix-2 FFT algorithm, 198, 225
  - Goertzel algorithm, 221, 223
  - in-place computation, 209
    - Bit-Reversal, 209, 210
  - radix-4 DIF FFT algorithm, 217
- Filter design
  - using frequency sampling method, 373
- Final value theorem, 156
- Finite word length effect
  - effect of quantization, 454
  - filter coefficients quantization, 458
    - pole sensitivity, 459
  - fixed-point quantization, 454
  - input-quantization errors, 470
    - output noise power, 473
  - number representation, 449
    - fixed-point, 450
    - floating-point, 453
  - optimum coefficient word length, 469
  - product-quantization effect, 475
    - direct Form I, 477
    - direct Form II, 477
  - cascade, 479
  - parallel, 480
- FIR differentiator design, 366
- FIR filter design using windowing method
  - comparison of the fixed windows, 343
  - design procedure, 355
    - bandpass, 356
    - bandstop, 357
    - highpass, 356
    - using Kaiser window, 355
  - FIR filter design using fixed windows, 348
- fixed window functions, 341
  - rectangular, 341, 344–347
  - triangular or Bartlett, 341, 344–347, 411
  - raised cosine, 342
  - Hanning, 342, 345–348, 410, 411
  - Hamming, 342–346, 348, 350–353
  - Blackman, 343, 345–348
- Gibb's oscillations, 340, 348
- Kaiser window, 354
- FIR filter ideal impulse response
  - bandpass filter, 327, 328, 350, 351
  - bandstop filter, 328, 329
  - highpass filter, 326–328
  - lowpass filter, 326, 328
- FIR filter structures
  - cascade, 442
  - direct-form, 440
  - FIR lattice, 445
  - linear phase, 443
  - realization using matlab, 448
- FIR transfer function, 138
- Fourier transform of discrete-time signals
  - convergence of the DTFT, 58
  - properties of DTFT, 64, 66, 68, 100
    - for a complex sequence, 20, 63, 64, 100
    - for a real sequence, 20, 64, 65, 66, 100
  - theorems on DTFT, 59
    - convolution theorem, 60, 62
    - correlation Theorem, 60, 62
    - differentiation in frequency, 60
    - frequency shifting, 59, 62, 84, 100
    - linearity, 35, 38, 39, 59, 62, 68, 97, 100
    - Parseval's theorem, 61, 62, 72
    - time reversal, 21, 22, 59, 62
    - time shifting, 21, 59, 62, 67, 77, 100
    - windowing theorem, 60, 62
- Frequency response from poles and zeros, 139
- Frequency response of discrete-time systems
  - frequency response computation using MATLAB, 80
  - phase and group delays, 75

**G**

Graphical method, 188

**H**

Hilbert transformer, 369, 385, 394, 395, 412

IIR digital filter design using MATLAB, 295

IIR filter, 241, 268, 277, 278, 280, 303, 310, 312, 313, 323, 324

IIR filter design

using MATLAB GUI filter designer  
SPTOOL, 241, 303

IIR filter structures

cascade, 422

direct form I, 420

direct form II, 421

Gray-Markel's lattice structure, 431

lattice structure of all -pole system, 426

parallel form, 424

**I**

Image compression, 660, 673, 683

Image wavelet packets

admissible quad tree, 674

image denoising, 679

signal denoising, 662, 675, 678

Impulse and step responses using MATLAB, 39, 57

Initial value theorem, 119

Inverse discrete Fourier transform, 168, 227

Inverse STFT, 619, 623

Inverse z-transform

Cauchy's residue theorem, 124

modulation theorem, 121

Parseval's relation, 122, 123, 150

partial fraction expansion, 124–126, 128–130, 160

partial fraction expansion using MATLAB, 129, 134, 138

power series expansion, 124, 130, 131, 134, 161

power series expansion Using MATLAB, 134

**K**

Kaiser window-based filter design

using MATLAB, 370

**L**

Limit cycles in IIR digital filters

due to round-off and truncation of products, 499

overflow limit cycles, 503

Linear convolution using DFT

two finite length sequences, 190

finite length sequence with a long duration sequence, 192

overlap-add, 192, 239

overlap-save, 192, 193, 239

Linear phase FIR filters

FIR transfer functions, 331

type 1, 331, 332, 337, 391

type 2, 332, 333, 337

type 3, 333, 334, 337, 338, 369

type 4, 334, 335, 337, 338

zero locations, 335, 337–339

Linear phase FIR PR QMF banks design methods

Johnston method, 599

example, 594

lattice structures, 602, 604

two-channel FIR filter bank, 582

design using Matlab, 605

perfect reconstruction

LTI discrete time systems

computation of linear convolution, 42, 43

graphical method, 43

matrix method, 42

computation of convolution sum using matlab, 44

convolution sum, 19, 41, 43–46, 74

examples, 80

input-output relationship, 35, 40

properties of convolution sum, 45

**M**

MATLAB exercises, 412

Memory organization, 758

Multiprocessor support, 758

Multi-rate bandpass and bandstop filter design

notch frequency, 562

quadrature modulation structure, 560, 561, 563, 565

Multirate signal processing

advantages, 513

Multirate signal processing concepts

changing the sampling rate, 530–532  
down-sampling, 514, 515, 517–520, 522, 523, 526, 570

sampling rate conversion

via multi-stage approach, 533

up sampling, 523, 524, 550

Multiresolution analysis, 619, 629, 636, 669

**N**

Non stationary signals, 747, 748

STFT, 619, 621–624, 685, 720, 747

**O**

- On-chip memory, [755](#), [771](#)
- On-chip peripherals, [757](#), [758](#), [759](#), [764](#), [770](#)
- One-sided  $z$ -transform, [154](#), [157](#), [158](#), [160](#)
- Optimal FIR filter design
  - alternation theorem, [379](#)
  - Lagrange interpolation, [380](#)
  - Remez exchange algorithm, [380](#), [387](#)
  - tolerance ratio, [381](#), [383](#), [412](#)
  - type 1, [375](#)
  - type 2, [376](#)
  - type 3, [76](#), [379](#), [385](#)
  - type 4, [377](#), [385](#)
  - using matlab, [385](#)
- Oversampling ADC
  - resolution analysis, [553](#)
  - quantization noise reduction, [553](#), [554](#), [557](#), [558](#), [571](#)
  - sigma delta modulation ADC, [555](#), [557](#)

**P**

- Parallelism, [756](#), [771](#)
- Pipelining, [755](#)
- Polyphase decomposition
  - decimator and interpolator structures, [533](#), [534](#)
- Polyphase implementations
  - alias-free  $M$ -channel QMF bank, perfect reconstruction
  - alias-free  $M$ -channel QMF bank, polyphase representation
    - conditions for existence
    - equivalent analysis and synthesis filters
    - FIR analysis/synthesis filters
    - perfect reconstruction
    - polyphase representation
      - type 1 analysis filter
      - type 2 synthesis filter
  - DFT analysis filter bank implementation
  - two-channel Quadrature mirror filter (QMF) bank
    - analysis structure
- Power consumption and management, [758](#)
- Practical sampling rate converter design
  - filter requirements for individual stages, [534](#), [535](#)
  - illustrative design examples, [535](#)
  - overall filter specifications, [534](#)
- Properties of ROC
  - ROC for a finite duration causal sequence, [104](#)
  - ROC for a finite duration two-sided sequence, [105](#), [107](#), [108](#)

- ROC for an infinite duration left-sided sequence, [106](#)
- ROC for an infinite duration right-sided sequence, [105](#)
- ROC for a non-causal finite duration sequence, [104](#)
- ROC of an infinite duration two sided sequence, [107](#), [108](#)

**Q**

- Quantization effect in FFT computation
  - direct computation of the DFT, [505](#)
  - FFT computation, [506](#)

**R**

- Rational transfer function
  - poles and zeros, [108](#), [138](#), [139](#), [146](#), [150–152](#)
- Rational  $z$ -transform, [102](#), [124](#)
- Realization using matlab, [434](#)
- Reconstruction of a bandlimited signal from its samples, [16](#), [19](#), [88](#)

**S**

- Sampling in frequency-domain
  - aliasing, [32](#), [87](#)
  - over-sampling, [87](#)
  - sampling theorem, [32](#), [86](#), [88](#)
  - under sampling, [87](#)
- Sampling process of analog signals
  - impulse-train sampling, [30](#)
  - quantization and coding, [33](#)
  - quantization error, [34](#)
  - sampling frequency, [30](#), [32–34](#), [87](#)
  - sampling period, [30](#), [32](#)
  - sampling theorem, [32](#), [86](#), [88](#)
  - sampling with a zero-order hold, [31](#)
- Scaling functions and wavelets
  - dilation equation, [628](#), [629](#)
  - examples, [625](#), [627](#)
  - Haar, [625](#), [626](#), [627](#), [629–631](#), [644](#), [651](#)
- Scaling of IIR digital filters
  - cascade form, [492](#)
  - pole-zero pairing, [496](#)
  - ordering, [496](#)
  - for a second-order filter, [489](#)
  - in fixed-point realization, [486](#)
  - in parallel structure, [491](#)
- Signal flow graphs
  - transposition, [417](#)
- Single instruction, multiple data (SIMD), [754](#), [756](#)
- Special addressing modes, [756](#)
- Specialized execution control, [757](#)

- Specialized instruction sets, 757
- Specialized digital filter design
  - comb filter, 308, 309
  - notch filter, 304, 307–311, 323
  - pole-zero placement, 304, 307, 309
- Speed, 758, 765
- Solution of difference equations
  - characteristic equation, 53
  - complementary solution, 52–56
  - particular solution, 53–56
- Spectral analysis, 721, 747
- Stability and causality, 141, 142, 148
- Stability and causality of LTI Systems in terms of the impulse response
  - examples, 80
- Stationary signals
  - non-parametric methods, 721, 723, 727, 729
    - Bartlett method, 726
    - Blackman-Tukey method, 730
    - periodogram, 724
    - Welch method, 728
  - parametric methods, 721, 733
    - AR model, 735, 742
      - backward linear prediction, 738
      - Burg method, 736, 740
      - forward linear prediction, 738
      - MA model, 741, 742
      - model order selection
      - Yule-Walker method, 735, 736
  - power spectrum estimation
  - subspace methods, 721, 742
    - Eigen vector method, 746
    - MUSIC method, 745
    - Pisarenko harmonic decomposition, 743
- Streamlined I/O, 757
- Symmetry relations of DFT
  - DFTs of two real sequences from a single N-point DFT, 187
  - of complex-valued sequences, 181
  - of real-valued sequences, 184, 186
- Systems
  - all-pass, 160
  - inverse, 101, 120–122, 124–128, 147, 149, 157
  - maximum-phase, 146, 147
  - minimum-phase, 146–148, 152
  - mixed-phase, 146–148, 152
- T**
- TigerSHARC TS201 DSP Processor
  - architectural features, 758, 770
  - addressing modes, 759
  - creating a visualDSP++ project, 775
  - digital narrow bandstop filter implementation, 775
- Time-frequency representation of signals, 620, 621
- TMS320C54xx digital signal processor
  - architectural features, 758
  - data addressing modes, 759
  - code composer studio, 761
  - TMS320C54xx example programs, 762
- TMS320C67xx digital signal processor
  - FIR LPF implementations, 767, 768
- Tree structured filter banks
  - equal passband width, 606, 610
  - maximally decimated, 606, 607
  - unequal passband width, 613
- Two-band digital crossover design, 408–410
- Typical signal processing operations
  - correlation, 4, 11, 12
  - digital filtering, 4
  - discrete transformation, 4
  - elementary time-domain operations, 3
  - modulation and demodulation, 4
  - multiplexing and demultiplexing, 5
  - quadrature amplitude modulation, 5
- U**
- Uniform DFT filter banks
  - L*th- band filter, 576, 577, 578
  - linear phase *L*th-band filter design, 577
  - design example, 578
- V**
- Very Long Instruction Word* or *VLIW*, 754, 764, 765
- W**
- Wavelet packets
  - admissible tree, 670
  - decomposition, 636, 637, 642, 650, 653, 659, 665
  - wavelet reconstruction, 636, 637
- wavelets properties
  - orthogonality, 637, 638, 640, 645, 660
  - regularity condition, 640, 642, 645
- Z**
- z*-transform
  - definition, 101, 103, 109, 111, 115, 116, 132, 154, 155
  - Region of convergence (ROC), 104, 108, 111, 133
- z*-transform properties
  - conjugate of a complex sequence, 113
  - convolution of two sequences, 112

- correlation of two sequences, [112](#)
- differentiation in the  $z$ -domain, [110](#)
- imaginary part of a sequence, [114](#)
- linearity, [109](#), [114](#), [115](#), [137](#)
- real part of a sequence, [113](#)
- scaling in the  $z$ -domain, [110](#)
- time reversal, [109](#), [113](#), [115](#), [116](#)
- time shifting, [109](#), [115](#), [116](#), [137](#)
- $z$ -transforms of some commonly-used sequences
  - unit step sequence, [116](#)