

INDEX

- A/D conversion. *See* Analog-to-digital (A/D) conversion
- Absolute summability, 50–52, 65
defined, 50
for suddenly-applied exponential, 50–51
- Accumulator, 19–20, 23, 33, 35
and the backward difference system, 35
difference equation representation of, 36
impulse response of, 33
in cascade with backward difference, 35
inverse system, 35
system, 19–20
as time-invariant system, 21
- Additivity property, 19
- Alias cancellation condition, 203
- Aliasing, 159
antialiasing filter, 206–207
and bilinear transformation, 506
distortion, 159
downsampling with, 181–184
prefiltering to avoid, 206–209
in sampling a sinusoidal signal, 162
- All-pass systems, 305–310
first- and second-order, 307–309
- All-pole lattice structure, 412–414
- All-pole model lattice network, 923–924
- All-pole modeling, 891–895
autocorrelation matching property, 898
determination of the gain parameter G , 899–900
of finite-energy deterministic signals, 896–897
least-squares approximation, 892
least-squares inverse model, 892–894
linear prediction, 892
linear prediction formulation of, 895
linear predictive analysis, 892
minimum mean-squared error, 898
random signal modeling, 897
- All-pole spectrum analysis, 907–915
pole locations, 911–913
sinusoidal signals, 913–915
speech signals, 908–911
- Alternation theorem, 558–565
defined, 558
and polynomials, 558
- Analog signals, 9
digital filtering of, 205–224
A/D conversion, 209–214
D/A conversion, 220–224
ideal continuous-to-discrete (C/D) converter, 205
ideal discrete-to-continuous (D/C) converter, 205
- Analog-to-digital (A/D) conversion, 2, 209–214
measurements of quantization noise, 217–218
offset binary coding scheme, 212
oversampling and noise shaping in, 224–236
physical configuration for, 209
quantization errors:
analysis of, 214–215, 214–220
for a sinusoidal signal, 215–217
quantizer, 210–213
- Analog-to-digital (A/D) converters, 205
- Analytic signals, 956, 969
and bandpass sampling, 966–969
as a complex time function, 943
defined, 943
and narrowband communication, 963
- Analyzer-synthesizer filter banks, 266
- Antialiasing filter, 206–207, 793–795
frequency response of, 793
- Aperiodic discrete-time sinusoids, 15
- Asymptotically unbiased estimators, 837

- Autocorrelation:
 circular, 853
 deterministic autocorrelation sequence, 67
 invariance, 607
 method, 900–903
 and parametric signal modeling, 900–903
- Autocorrelation matching property, 898
 all-pole modeling, 898
- Autocorrelation normal equations, 896
 Levinson–Durbin algorithm, derivation of, 917–920
 Levinson–Durbin recursion, 916–917
 solutions of, 915–919
- Autocorrelation sequence of $h[n]$, 67–68
- Autoregressive (AR) linear random process, 887
- Autoregressive moving-average (ARMA) linear random process, 887
- Backward difference, 13, 22, 33
- Backward difference system, 22, 33, 93, 96
 and the accumulator, 35
 impulse response of, 34
- Backward prediction error, 922–923, 925–926, 941
- Bandlimited interpolation, 264
- Bandlimited signal, reconstruction from its samples, 163–166
- Bandpass filter, and FIR equiripple approximation, 576–577
- Bartlett (triangular) windows, 536–539, 823–824, 862–863
- Bartlett's procedure, 844
- Basic sequences/sequence operations:
 complex exponential sequences, 14–15
 exponential sequences, 13–14
 sinusoidal sequences, 14
 unit sample sequence, 12
 unit step sequence, 12–13
- Basis sequences, 673
- Bilateral z -transform, 100, 135
- Bilinear transformation, 504–508
 and aliasing, 506
 of a Butterworth filter, 509–513
 frequency warping, 506–507
- Bit-reversed order, 732
- Blackman–Tukey estimates, 862
- Blackman windows, 536–539, 824
- Blackman–Tukey method, 850, 862
- Block convolution, 668
- Block floating point, 762
- Block processing, 792
- Bounded-input, bounded-output (BIBO), 22
- Butterfly computation, 730
- Butterworth filter, 581
 bilinear transformation of, 509–513
 impulse invariance, 500–504
- Canonic direct form implementation, 381
- Canonic form implementation, 380–381
- Cascade-form structures, 390–393
 illustration of, 392
- Cascade IIR structure, analysis of, 448–453
- Cascaded systems, 34–35
- Cauchy integral theorem, 943
- Cauchy principal value, 949
- Cauchy–Riemann conditions, 943
- Causal generalized linear-phase systems, 328–338
- FIR linear-phase systems:
 examples of, 330–331
 locations of zeros for, 335–338
 relation to minimum-phase systems, 338–340
 type I FIR linear-phase systems, 330
 example, 331–332
 type II FIR linear-phase systems, 330
 example, 332–333
 type III FIR linear-phase systems, 330
 example, 333–334
 type IV FIR linear-phase systems, 330
 example, 335
- Causal sequences, 32
 even and odd parts of, 945
 exponential sequence, 947
 finite-length sequence, 946
 real- and imaginary-part sufficiency of the Fourier transform for, 944–949
- Causal systems, 32
- Causality, 22
- Cepstrum:
 complex, defined, 982–984
 defined, 981–982
 real, 984fn
- Cepstrum analysis, 980
- Characteristic system for convolution, 984
- Characteristic, use of term, 419, 458
- Chebyshev criterion, 557
- Chebyshev I design, 581
- Chebyshev II design, 581
- Chirp signals, 751
- Chirp transform algorithm (CTA), 749–754
 parameters, 754
- Chirp transform, defined, 751

- Chirp z -transform (CZT) algorithm, 754
Chirps, defined, 812
Cholesky decomposition, 916
Circular autocorrelation, 853
Circular convolution, discrete Fourier transform (DFT), 654–659
Circular shift of a sequence, discrete Fourier transform (DFT), 648–650
Clipped samples, 214
Clipping, 416
“Closed-form” formulas, 29
Clutter, 835
Coefficient quantization, effects of, 421–436
 in an elliptical filter, 423–427
 in FIR systems, 429–431
 in IIR systems, 422–423
 maintaining linear phase, 434–436
 in an optimum FIR filter, 431–434
 poles of quantized second-order sections, 427–429
Commonly used windows, 535–536
Complex cepstrum, 955–956, 979
 alternative expressions for, 985–986
 computation of, 992–1000
 exponential weighting, 1000
 minimum-phase realizations for minimum-phase sequences, 998
 phase unwrapping, 993–997
 recursive computation for minimum- and maximum-phase sequences, 999–1000
 using polynomial roots, 1001–1002
 using the DFT, 1013–1016
 using the logarithmic derivative, 997–998
by z -transform analysis, 1009–1012
deconvolution using, 1002–1006
 minimum-phase/allpass homomorphic deconvolution, 1003–1004
 minimum-phase/maximum-phase homomorphic deconvolution, 1004–1005
defined, 982–984
for exponential sequences, 986–989
minimum phase and causality of, 956
for minimum-phase and maximum-phase sequences, 989–990
relationship between the real cepstrum and, 990–992
for a simple multipath model, 1006–1024
 generalizations, 1024
 homomorphic deconvolution, 1016–1017
 minimum-phase decomposition, 1017–1023
speech processing applications, 1024–1032
 applications, 1032
 formants, 1024
 fricative sounds, 1024
 homomorphic deconvolution of speech, example of, 1028–1030
 plosive sounds, 1024
speech model, 1024–1027
 speech model, estimating the parameters of, 1030–1032
 vocal tract, 1025
 voiced sounds, 1024
Complex exponential sequences, 14–15, 53–54
Complex logarithm, 982
 properties of, 984–985
Complex sequences, Hilbert transform relations for, 956–969
Complex time functions, analytic signals as, 959
Compressor, 180
Compressor, defined, 21
Conjugate-antisymmetric sequence, 54–55
Conjugate quadrature filters (CQF), 203
Conjugate-symmetric sequence, 54–55
Conjugation property, z -transform, 129
Consistent estimators, 837
Consistent resampling, 250
Constant, Fourier transform of, 52–53
Continuous-time filters, design of discrete-time IIR filters from, 496–508
 bilinear transformation, 504–508
 filter design by impulse invariance, 497–504
Continuous-time processing of discrete-time signals, 175–178
 noninteger delay, 176–177
 moving-average system with noninteger with, 177–179
Continuous-time signals, 9
 aliasing in sampling a sinusoidal signal, 162
 bandlimited signal, reconstruction from its samples, 163–166
 digital filtering of analog signals, 205–224
 discrete-time lowpass filter, ideal continuous-time lowpass filtering using, 169–171
 discrete-time processing of, 167–176
 LTI, 168–169
 discrete-time signals, continuous-time processing of, 175–179
 frequency-domain representation of sampling, 154, 156–163

- Continuous-time signals (*continued*)
 ideal continuous-time bandlimited differentiator,
 discrete-time implementation of, 171–172
 impulse invariance, 173–175
 applied to continuous-time systems with rational
 system functions, 174–175
 discrete-time lowpass filter obtained by, 174
 multirate signal processing, 194–205
 reconstruction of a bandlimited signal from
 samples, 163–166
 sampling and reconstruction of a sinusoidal
 signal, 161–162
 sampling of, 153–273
 sampling rate, changing using discrete-time
 processing, 179–193
- Convolution:
 characteristic system for, 984
 circular, 654–659
 commutative property of, 34
 linear, 660–672
 with aliasing, circular convolution as, 661
 of two finite-length sequences, 660–672
 operation, 30
- Convolution property, *z*-transform, 130–131
 convolution of finite-length sequences, 131
- Convolution sum, 24–26
 analytical evaluation of, 27–29
 computation of, 23–27
 defined, 24
- Convolution theorem:
 Fourier transform, 60–61
z-transform, 130–131
- Cooley-Tukey algorithms, 735, 746–747, 749
- Coupled form, for second-order systems, 429
- Coupled form oscillator, 471
- Critically damped system, 351
- Cross-correlation, 69–70, 881, 883, 925
- CTA, *See* Chirp transform algorithm (CTA)
- D/A conversion, *See* Digital-to-analog (D/A)
 conversion
- DCT, *See* Discrete cosine transform (DCT)
- DCT-1/DCT-2, *See also* Discrete cosine transform
 (DCT)
 defined, 675
 relationship between, 676–678
- Dead bands, defined, 461
- Decimation:
 defined, 184
 multistage, 195–197
- Decimation filters, polyphase implementation of,
 199–200
- Decimation-in-frequency FFT algorithms, 737–743
 alternative forms, 741–743
 in-place computations, 741
- Decimation-in-time FFT algorithms, 723–737
 alternative forms, 734–737
 defined, 723
 generalization and programming the FFT, 731
 in-place computations, 731–734
- Decimator, defined, 184
- Decomposition:
 Cholesky, 916
 linear time-invariant (LTI) systems, 311–313
 minimum-phase, 1017–1023
 of one-sided Fourier transform, 958
 polyphase, 197–9
- Deconvolution:
 using the complex cepstrum, 1002–1006
 minimum-phase/allpass homomorphic
 deconvolution, 1003–1004
 minimum-phase/maximum-phase homomorphic
 deconvolution, 1004–1005
- Delay register, 375
- Deterministic autocorrelation sequence, 67–68
- DFS, *See* Discrete Fourier series (DFS)
- DFT, *See* Discrete Fourier transform (DFT)
- Difference equation representation, of accumulator,
 36
- Difference equations:
 block diagram representation of, 376–377
 determining the impulse response from, 64
- Differentiation property, *z*-transform, 127–129
 inverse of non-rational *z*-transform, 128
 second-order pole, 128–129
- Digital filters, 494
- Digital signal processing, 10–17
- Digital signal processors, 1fn
- Digital filtering of analog signals, 205–224
 A/D conversion, 209–214
 D/A conversion, 220–224
 ideal continuous-to-discrete (C/D) converter, 205
 ideal discrete-to-continuous (D/C) converter, 205
- Digital signals, 9

- Digital-to-analog (D/A) conversion, 2, 220–224
 block diagram, 221
 ideal D/C converter, 221
 oversampling and noise shaping in, 224–236
 zero-order hold, 222–223
- Digital-to-analog (D/A) converters, 205
- Dirac delta function, 12, 154
- Direct-form FIR systems, analysis of, 453–458
- Direct form I implementation, 381
- Direct form II implementation, 381
- Direct-form IIR structures, analysis of, 436–445
- Direct-form structures, 388–390
 illustrations of, 390
- Discrete cosine transform (DCT), 673–683
 applications of, 682–683
 DCT-1/DCT-2:
 defined, 675
 relationship between, 676–678
 definitions of, 673–674
 energy compaction property of the DCT-2, 679–682
 relationship between the DFT and the DCT-2, 678–679
- Discrete Fourier series (DFS), 624–628
 duality in, 627
 Fourier representation of finite-duration sequences, 642–646
- Fourier transform:
 of one period, relationship between Fourier series coefficients and, 637–638
 of a periodic discrete-time impulse train, 635
 of periodic signals, 633–638
 sampling, 638–641
 of a periodic rectangular pulse train, 627–628
 properties of, 628–633, 647–660
 circular convolution, 654–659
 circular shift of a sequence, 648–650
 duality, 629–630, 650–652
 linearity, 629, 647–648
 periodic convolution, 630–633
 shift of a sequence, 629
 summary, 634, 660–661
 symmetry, 630
 symmetry properties, 653–654
 of a rectangular pulse, 644–646
 representation of periodic sequences, 624–628
- Discrete Fourier transform (DFT), 3, 623–715
 computation of, 716–791
 coefficients, 745
 direct computation of, 718–723
 direct evaluation of the definition of, 718–719
 exploiting both symmetry and periodicity, 722–723
 Goertzel algorithm, 717, 719–722, 749
 indexing, 743–745
 practical considerations, 743–745
 computation of average periodograms using, 845
 computation of the complex cepstrum using, 1013–1016
 computing linear convolution using, 660–672
 decimation-in-frequency FFT algorithms, 737–743
 alternative forms, 741–743
 in-place computations, 741
 decimation-in-time FFT algorithms, 723–737
 alternative forms, 734–737
 defined, 723
 generalization and programming the FFT, 731
 in-place computations, 731–734
 defined, 623, 641
 DFT analysis of sinusoidal signals, 797–810
 effect of spectral sampling, 801–810
 effect of windowing, 797–800
 window properties, 800–801
 DFT analysis of sinusoidal signals using a Kaiser window, 806–808
 discrete cosine transform (DCT), 673–683
 discrete Fourier series, 624–628
 properties of, 628–633
 finite-length sequences, sufficiency theorems for, 950
 finite register length, effects of, 754–762
 Fourier analysis of nonstationary signals:
 examples of, 829–836
 radar signals, 834–836
 speech signals, 830–834
 Fourier analysis of signals using, 792–796
 Fourier analysis of stationary random signals, 836–849
 computation of average periodograms using the DFT, 845
 periodogram, 837–843
 periodogram analysis, 837, 845–849
 periodogram averaging, 843–845
 general FFT algorithms, 745–748
 chirp transform algorithm (CTA), 749–754
 Winograd Fourier transform algorithm (WFTA), 749

- Discrete Fourier transform (DFT) (*continued*)
 implementing linear time-invariant systems using, 667–672
 linear convolution, 660–672
 with aliasing, circular convolution as, 661–667
 of two finite-length sequences, 661
 properties of, 647–660
 circular convolution, 655–659
 circular shift of a sequence, 648–650
 duality, 650–652
 linearity, 647–648
 summary, 659–660
 symmetry properties, 653–654
 of a rectangular pulse, 644–646
 signal frequencies matching DFT frequencies exactly, 805–806
 spectrum analysis of random signals using
 autocorrelation sequence estimates, 849–862
 correlation and power spectrum estimates, 853–855
 power spectrum of quantization noise, 855–860
 power spectrum of speech, 860–862
 time-dependent Fourier transform, 811–829
 defined, 811
 effect of the window, 817–818
 filter bank interpretation of, 826–829
 filter bank interpretation of $X[n, \lambda]$, 816–817
 invertibility of $X[n, \lambda]$, 815–816
 of a linear chirp signal, 811–814
 overlap-add method of reconstruction, 822–825
 sampling in time and frequency, 819–822
 signal processing based on, 825–826
 spectrogram, 814–815
 Discrete Hilbert transform relationships, 949
 Discrete Hilbert transforms, 942–979
 Discrete sinc transform (DST), 674
 Discrete-time Butterworth filter, design of, 508–526
 Discrete-time convolution, implementing, 26–27
 Discrete-time differentiators, 507, 550
 and Kaiser window filter design method, 550–553
 Discrete-time filters:
 design of, 493–494
 determining specifications for, 495–496
 IIR filter design, from continuous-time filters, 496–508
 bilinear transformation, 504–508
 filter design by impulse invariance, 497–504
 Discrete-time Fourier transform (DTFT), 49fn, 623, 792
 Discrete-time linear time-invariant (LTI) filter, 4
 Discrete-time model of speech production, 1025, 1025–1026
 Discrete-time processing, of continuous-time signals, 167–176
 Discrete-time random signals, 64–70
 Discrete-time signal processing, 2, 17–23
 backward difference system, 22
 causality, 22
 defined, 17
 discrete-time random signals, 64–70
 forward difference system, 22
 Fourier transforms, representation of sequences by, 48–54
 ideal delay system, 17
 instability, testing for, 23
 linear systems, 19–20
 accumulator system, 19–20
 nonlinear system, 20
 memoryless systems, 18–19
 moving average, 18
 stability, 22–23
 testing for, 23
 techniques, future promise of, 8
 time-invariant systems, 20–21
 accumulator as, 21
 Discrete-time signals, 10
 basic sequences/sequence operations:
 complex exponential sequences, 14–15
 exponential sequences, 13–14
 sinusoidal sequences, 14
 unit sample sequence, 12
 unit step sequence, 12–13
 continuous-time processing of, 175–178
 defined, 9
 discrete-time systems, 17–23
 graphic depiction of, 11
 graphic representation of, 11
 sampling frequency, 10
 sampling period, 10
 as sequences of numbers, 10–17
 signal-processing systems, classification of, 10
 Discrete-time sinusoids, periodic/aperiodic, 15
 Discrete-time systems, 17–23
 coefficient quantization effects, 421–436
 in an elliptical filter, 423–427
 in FIR systems, 429–431
 in IIR systems, 422–423
 maintaining linear phase, 434–436
 in an optimum FIR filter, 431–434

- poles of quantized second-order sections, 427–429
discrete-time random signals:
autocorrelation/autocovariance sequence, 65–66
deterministic autocorrelation sequence, 67–68
power density spectrum, 68
random process, 65
white noise, 69–70
finite-precision numerical effects, 415–421
number representations, 415–419
quantization in implementing systems, 419–421
FIR systems:
basic network structures for, 401–405
cascade form structures, 402
direct-form structures, 401–402
floating-point realizations of, 458–459
Fourier transform theorems, 59–64
convolution theorem, 60–61
differentiation in frequency, 59
frequency shifting, 59
linearity of Fourier transform, 59
modulation or windowing theorem, 61–62
Parseval's theorem, 60
time reversal, 59
time shifting, 59
frequency-domain representation of, 40–48
eigenfunctions for linear time-invariant systems, 40–45
frequency response of the ideal delay system, 41
frequency response of the moving-average system, 45–46
ideal frequency-selective filters, 43–44
sinusoidal response of linear time-invariant systems, 42–43
suddenly-applied complex exponential inputs, 46–48
ideal delay system, 17
IIR systems:
basic structures for, 388–397
cascade form structures, 390–393
direct form structures, 388–390
feedback in, 395–397
parallel form structures, 393–395
lattice filters, 405–415
all-pole lattice structure, 412–414
FIR, 406–412
generalization of lattice systems, 415
lattice implementation of an IIR system, 414
linear constant-coefficient difference equations, 35–41
block diagram representation of, 375–382
signal flow graph representation of, 382–388
linear-phase FIR systems, structures for, 403–405
linear systems, 19
accumulator system, 19–20
nonlinear system, 20
linear time-invariant systems, 23–35
convolution sum, 23–29
eigenfunctions for, 40–46
properties of, 30–35
memoryless systems, 18–19
moving average, 18
representation of sequences by Fourier transforms, 48–54
absolute summability for suddenly-applied exponential, 50
Fourier transform of a constant, 52–53
Fourier transform of complex exponential sequences, 53
inverse Fourier transform, 48
square-summability for the ideal lowpass filter, 51–52
round-off noise in digital filters, effects of, 436–459
stability, 22–23
testing for, 23
structures for, 374–492
symmetric properties of Fourier transform, 55–57
conjugate-antisymmetric sequence, 55–56
conjugate-symmetric sequence, 55–56
even function, 55
even sequence, 54
illustration of, 56–57
odd function, 55
odd sequence, 54
time-invariant systems, 20–21
accumulator as, 21
compressor system, 21
transposed forms, 397–401
Doppler frequency, 834–835
Doppler radar:
defined, 793
signals, 835–836
Downsampling, 180–182
with aliasing, 181–184
defined, 180
frequency-domain illustration of, 181–182
with prefiltering to avoid aliasing, 183

- Duality:
 discrete Fourier series, 627, 629–630, 650–652
 discrete Fourier transform (DFT), 650–652
- Echo detection, and cepstrum, 982
- Eigenfunctions, 40–45
 for linear time-invariant (LTI) systems, 40–46, 61
- Eigenvalues, 40–41, *See also* Frequency response of Elliptic filter design, 508–526, 581
- Energy compaction property of the DCT-2, 679–682
- Energy density spectrum, 60
- Equiripple approximations, 560
- Even function, 55–56
- Even sequence, 54–55
- Even symmetry, 673
- Expander (sampling rate), 184
- Exponential multiplication property, *z*-transform, 126–127
- Exponential sequences, 13–14, 947
- External points, 558
- Extraripple casc, 561
- Extremals, 558
- Fast Fourier transform (FFT) algorithms, 3–4, 6–7, 660, 671, 716
 decimation-in-frequency FFT algorithms, 737–743
 decimation-in-time FFT algorithms, 723–737
 finite register length, effects of, 754–762
 general FFT algorithms, 745–748
- FFT, *See* Fast Fourier transform (FFT) algorithms
- FFTW (“Fastest Fourier Transform in the West”) algorithm, 748
- Filter bank interpretation, of time-dependent Fourier transform, 826–829
- Filter banks:
 analyzer-synthesizer, 266
 multirate, 201–205
 alias cancellation condition, 203
 quadrature mirror filters, 203
- Filter design:
 bilinear transformation, 504–508
 Butterworth filter, 508–526
 bilinear transformation of, 509–513
 Chebyshev filter, 508–526
 elliptic filters, 508–526
 FIR filters, 578
 design by windowing, 533–545
 FIR equiripple approximation examples, 570–577
 optimal type I lowpass filters, 559–564
 optimal type II lowpass filters, 565–566
 optimum approximations of, 554–559
 optimum lowpass FIR filter characteristics, 568–570
 Parks-McClellan algorithm, 566–568
- IIR filters, 578
 design comparisons, 513–519
 design example for comparison with FIR designs, 519–526
 design examples, 509–526
 by impulse invariance, 497–504
 lowpass IIR filters, frequency transformations of, 526–532
 specifications, 494–496
 stages of, 494
 techniques, 493–622
 upsampling filter, 579–582
- Filters, 493
- Financial engineering, defined, 3
- Finite-energy deterministic signals, all-pole modeling of, 896–897
- Finite impulse response (FIR) systems, 493
- Finite-length sequences, 946
 convolution of, 131
 sufficiency theorems for, 949–954
- Finite-length truncated exponential sequence, 109
- Finite-precision numerical effects, 415–421
 number representations, 415–419
 quantization in implementing systems, 419–421
- Finite register length, effects of, 754–762
- FIR equiripple approximation, examples of, 570–577
 bandpass filter, 576–577
 compensation for zero-order hold, 571–575
 lowpass filter, 570–571
- FIR filters:
 design by windowing, 533–545
 incorporation of generalized linear phase, 538–541
 Kaiser window filter design method, 541–553
 properties of commonly used windows, 535–538
 optimum approximations of, 554–559
- FIR lattice filters, 406–412
- FIR linear-phase systems:
 examples of, 330–331
 locations of zeros for, 335–338
 relation to minimum-phase systems, 338–340
 type I FIR linear-phase systems, 329
 example, 331–332
 type II FIR linear-phase systems, 330
 example, 332–333

- type III FIR linear-phase systems, 330
 - example, 333–334
 - type IV FIR linear-phase systems, 330
 - example, 335
 - FIR systems:
 - basic network structures for, 401–405
 - cascade form structures, 403
 - direct-form structures, 401–402
 - coefficient quantization, effects of, 429–431
 - First backward difference, 93
 - Fixed-point realizations of IIR digital filters,
 - zero-input limit cycles in, 459–463
 - Floating-point arithmetic, 458
 - Floating-point operations (FLOPS), 747–748
 - Floating-point realizations of discrete-time systems, 458–459
 - Floating-point representations, 419
 - Flow graph:
 - reversal, 397
 - transposition of, 397–401
 - Formants, 830, 1024
 - Forward difference, 33
 - Forward difference systems, 22
 - noncausal, 34–35
 - Forward prediction error, 922
 - Fourier analysis of nonstationary signals:
 - examples of, 829–836
 - radar signals, 834–836
 - speech signals, 830–834
 - Fourier analysis of signals using DFT, 792–796
 - basic steps, 793
 - DFT analysis of sinusoidal signals, 797–810
 - effect of spectral sampling, 801–810
 - effect of windowing, 797–800
 - window properties, 800–801
 - relationship between DFT values, 796
 - Fourier analysis of sinusoidal signals:
 - effect of windowing on, 798–800
 - Fourier analysis of stationary random signals, 836–849
 - computation of average periodograms using the DFT, 845
 - periodogram, 837–843
 - periodogram analysis, example of, 845–849
 - periodogram averaging, 843–845
 - Fourier transform:
 - of complex exponential sequences, 53–54
 - of a constant, 52–53
 - convolution theorem, 60–61
 - differentiation in frequency theorem, 59
 - linearity of, 59
 - magnitude of, 49
 - modulation or windowing theorem, 61–62
 - of one period, relationship between Fourier series coefficients and, 637–638
 - pairs, 62
 - Parseval's theorem for, 60
 - of a periodic discrete-time impulse train, 635
 - of periodic signals, 633–638
 - phase of, 49
 - sampling, 638–641
 - symmetry properties of, 54–58
 - theorems, 58–64
 - time reversal theorem, 59
 - time shifting and frequency shifting theorem, 59
 - of a typical window sequence, 795
- Fourier transforms:
- defined, 49
 - discrete-time (DTFT), 49fn
 - inverse, 48–49
 - representation of sequences by, 48–54
- Frequency, 14
- Frequency-division multiplexing (FDM), 266
- Frequency-domain representation:
 - of discrete-time signals/systems, 40–48
 - of sampling, 154–157
- Frequency estimation:
 - oversampling and linear interpolation for, 809–810
- Frequency response, 40–43
 - of antialiasing filter, 793
 - defined, 40–41
 - determining the impulse response from, 63
 - of the ideal delay system, 41
 - of linear time-invariant (LTI) systems, 275–283
 - effects of group delay and attenuation, 278–283
 - frequency response phase and group delay, 275–278
 - of the moving-average system, 45–46
- for rational system functions, 290–301
 - examples with multiple poles and zeros, 296–301
 - first-order systems, 292–296
 - second-order FIR system, 298
 - second-order IIR system, 296–298
 - third-order IIR system, 299–301
- Frequency-response compensation of non-minimum-phase systems, 313–318

- Frequency-sampling filters, 480, 487
 Frequency-sampling systems, 396
 Frequency-selective filters, 493, *See also Filter design*
 obtaining from a lowpass discrete-time filter, 527
 Frequency shifting, 59
 Frequency warping, and bilinear transformation of,
 506–507
 Fricative sounds, 830, 1024
 Gain, 275
 Generalized linear phase, linear systems with,
 326–328
 examples of, 327–328
 Gibbs phenomenon, 52, 534
 Goertzel algorithm, 717, 719–722
 Hamming windows, 280fn, 536–539, 823–824,
 862–863
 Hann windows, 536–539, 823–824
 Highpass filter, transformation of a lowpass filter to,
 530–532
 Hilbert transform relations, 942–979
 for complex sequences, 956–969
 defined, 943
 finite-length sequences, 946
 sufficiency theorems for, 949–954
 between magnitude and phase, 955
 Poisson's formulas, 943
 real- and imaginary-part sufficiency of the Fourier
 transform for causal sequences, 944–949
 relationships between magnitude and phase,
 955–956
 Hilbert transform relationships, 942, 969
 discrete, 948–949
 Hilbert transformer, 958
 bandpass sampling, 966–969
 bandpass signals, representation of, 963–966
 design of, 960–963
 impulse response of, 960
 Kaiser window design of, 960–963
 Homogeneity property, 19
 Homogeneous difference equation, 38
 Homogeneous solution, 38
 Homomorphic deconvolution, 980, 1026
 of speech, example of, 1028–1030
 Homomorphic systems, 980
 Ideal 90-degree phase shifter, 602, 958–959
 Ideal continuous-to-discrete-time (C/D) converter,
 154, 205
 Ideal D/C converter, 221
 Ideal delay impulse response, 32
 Ideal delay system, 17
 frequency response of, 41–43
 Ideal discrete-to-continuous (D/C) converter, 205
 Ideal lowpass filter, square-summability for, 51–52
 IIR digital filters, zero-input limit cycles in
 fixed-point realizations of, 459–463
 IIR systems:
 basic structures for, 388–397
 cascade form structures, 390–393
 coefficient quantization, effects of, 422–423
 direct forms structures, 388–390
 feedback in, 395–397
 lattice implementation of, 414
 parallel form structures, 393–395
 scaling in fixed-point implementations of, 445–448
 Impulse invariance:
 basis for, 504
 with a Butterworth filter, 500–504
 design procedure, 504
 filter design by, 497–504
 Impulse response:
 of accumulator, 33
 determining for a difference equation, 64
 determining from the frequency response, 63
 for rational system functions, 288–290
 Impulse sequence, 13
 In-place computations:
 decimation-in-frequency FFT algorithms, 741
 decimation-in-time FFT algorithms, 731–734
 defined, 732
 Indexing, discrete Fourier transform (DFT), 743–745
 Infinite-duration impulse response (IIR) systems, 33
 Infinite impulse response (IIR) systems, 493
 Initial-rest conditions, 39
 Initial value theorem, 151
 for right-sided sequences, 1037
 Inspection method, inverse z -transform, 116
 Instability, testing for, 23
 Instantaneous frequency, 812
 Integer factor:
 increasing sampling rate by, 184–187
 reducing sampling rate by, 180–184
 Integrator, 589
 Interpolated FIR filter, defined, 196
 Interpolation, 187–190
 defined, 185

- Interpolation filters, 187–190
 polyphase implementation of, 200–201
- Interpolator, 187–190
 defined, 185
 multistage, 195–197
- Inverse Fourier transforms, 48–49, 63
- Inverse systems, 35, 286–288
 accumulator, 35
 defined, 33
 for first-order system, 287
 inverse for system with a zero in the ROC, 288
- Inverse z -transform, 115–124
 inspection method, 116
 inverse by partial fractions, 120–121
 partial fraction expansion, 116–117
 power series expansion, 122–124
 second-order z -transform, 118–120
- JPEG (Joint Photographic Expert Group), 1fn
- k -parameters, 406, 920
 direct computation of, 925–926
- Kaiser window, 800–801, 823
 design, 581
 DFT analysis of sinusoidal signals using, 806–808
 and zero-padding, DFT analysis with, 808–809
- Kaiser window design, of Hilbert transformers, 960–962
- Kaiser window filter design method, 541–553
 examples of FIR filter design by, 545–553
 discrete-time differentiators, 550–553
 highpass filter, 547–550
 lowpass filter, 545–547
 relationship of the Kaiser window to other windows, 544–545
- LabView, 3, 509
- Lattice filters, 405–415, 920–926
 all-pole lattice structure, 412–414
 all-pole model lattice network, 923–924
 direct computation of k -parameters, 925–926
 FIR, 406–412
 generalization of lattice systems, 415
 lattice implementation of an IIR system, 414
 prediction error lattice network, 921–923
- Lattice k -to- α algorithm, 921
- Lattice systems:
 generalization of, 415
- Laurent series, 102
- Leakage, 800
- Least-squares approximation, 892
 all-pole modeling, 892
- Least-squares inverse model, 892–894
 all-pole modeling, 892–894
- Left-sided exponential sequence, 104–105
- Levinson–Durbin algorithm, derivation of, 917–921, 926
- Levinson–Durbin recursion, 916–917, 923
- Limit cycles:
 avoiding, 463
 defined, 667
 owing to overflow, 462–463
 owing to round-off and truncation, 460–462
- Limit cycles, defined, 461
- Linear chirp signal, of a time-dependent Fourier transform, 811–814
- Linear constant-coefficient difference equations, 35–41
 difference equation representation of, 36
 of the moving-average system, 37
 homogeneous difference equation, 38
 systems characterized by, 283–290
 impulse response for rational system functions, 288–290
 inverse systems, 286–288
 second-order system, 284–285
 stability and causality, 285–286
- Linear convolution, 660–672
 with aliasing, circular convolution as, 661–667
 of two finite-length sequences, 661
- Linear interpolation, 187–190
- Linear phase:
 causal generalized linear-phase systems, 328–338
 generalized, 326–338
 ideal lowpass with, 324–326
 systems with, 322–326
- Linear-phase filters, 345
- Linear-phase FIR systems, structures for, 403–405
- Linear-phase lowpass filters, and windowing, 540–541
- Linear prediction formulation, of all-pole modeling, 895
- Linear predictive analysis, 892
- Linear predictive coding (LPC), 4, 892fn
- Linear predictor, 895
- Linear quantizers, 211fn
- Linear systems, 19–20, 326
 accumulator system, 19–20
 nonlinear system, 20

Linear time-invariant (LTI) systems, 4, 23–35
 all-pass systems, 305–310
 first- and second-order, 307–309
 cascade combination of, 31
 convolution operation, 30
 convolution sum, 24–26
 analytical evaluation of, 27–29
 eigenfunctions for, 40–46
 FIR linear-phase systems, relation to, 338–340
 frequency response for rational system functions, 290–301
 frequency response of, 275–283
 linear constant-coefficient difference equations, 35–40
 systems characterized by, 283–290
 linear systems with generalized linear phase, 322–340
 causal generalized linear-phase systems, 328–338
 generalized linear phase, 326–328
 systems with linear phase, 322–326
 minimum-phase systems, 311–322
 decomposition, 311–313
 relation of FIR linear-phase systems to, 338–340
 non-minimum-phase systems, frequency response compensation of, 313–318
 parallel combination of, 31
 properties of, 30–35
 relationship between magnitude and phase, 301–305
 sinusoidal response of, 42
 suddenly applied complex exponential inputs, 46–48
 system function, 115, 132
 transform analysis of, 274–373
 z -transform and, 131–134

Linearity:
 discrete Fourier series (DFS), 647–648
 discrete Fourier transform (DFT), 647–648
 of Fourier transform, 59
 z -transform, 124–125

Linearity property, z -transform, 124–125

Long division, power series expansion by, 119

Lowpass filter:
 and FIR equiripple approximation, 570–571
 tolerance scheme, 495
 transformation to a highpass filter, 530–532

Lowpass IIR filters, frequency transformations, 526–532

Magnitude:
 of Fourier transform, 49
 of frequency response, 275, 281
 relationship between phase and, 301–305

Magnitude response, 275

Magnitude spectrum, 915

Magnitude-squared function, 291, 305, 311, 501, 510–511, 513, 1020

Mantissa, 419, 458

Mason's gain formula of signal flow graph theory, 397fn

Matched filter, 79

Mathematica, 3

MATLAB, 3, 283, 509, 579

Maximally decimated filter banks, 202fn

Maximum energy-delay systems, 320

Maximum-phase systems, 320

Memoryless systems, 18–19

Microelectronic mechanical systems (MEMS), 8

Microelectronics engineers, and digital signal processing, 8

Minimax criterion, 557

Minimum energy-delay systems, 320

Minimum group-delay property, 319

Minimum mean-squared error, 898

Minimum-phase/allpass homomorphic deconvolution, 1003–1004

Minimum-phase echo system, complex cepstrum of, 989–990

Minimum phase-lag system, 318

Minimum-phase LTI filter, 349

Minimum-phase/maximum-phase homomorphic deconvolution, 1004–1005

Minimum-phase systems, 311–322
 decomposition, 311–313
 defined, 311
 frequency-response compensation of non-minimum-phase systems, 313–318
 properties of, 318–322
 minimum energy-delay property, 319–322
 minimum group-delay property, 319
 minimum phase-lag property, 318–319

Model order, 905–907
 selection, 906–907

Modified periodogram, 838

Modulation theorem, 61–62

Moore's Law, 2

Moving average, 18, 31–32

Moving-average (MA) linear random process, 887

- Moving-average system, 40
difference equation representation of, 37
frequency response of, 45–46
with noninteger delay, 177–179
- MP3 audio coding, 825
- MPEG-II audio coding standard, 828
- MPEG (Moving Picture Expert Group), 1fn
- Multidimensional signal processing, 4
- Multirate filter banks, 201–205
alias cancellation condition, 203
quadrature mirror filters, 203
- Multirate signal processing, 194–205
compressor/expander, interchange of filtering with, 194–195
defined, 194
interpolation filters, polyphase implementation of, 200–201
multirate filter banks, 201–205
multistage decimation and interpolation, 195–197
polyphase decompositions, 197–199
polyphase implementation of decimation filters, 199–200
- Multistage noise shaping (MASH), 233, 272
- N-point DFT, 723fn
- Narrowband communication, and analytic signals, 963
- Narrowband time-dependent Fourier analysis, 832
- Networking, 480
- Networks, use of term, 375fn
- 90-degree phase shifter, ideal, 958–959
- 90-degree phase splitters, 589
- Noise shaping:
in A/D conversion, 224–234
in analog-to-digital (A/D) conversion, 224–236
in D/A conversion, 234–236
multistage noise shaping (MASH), 233, 272
- Noise-shaping quantizer, 220–221
- Non-minimum-phase systems, frequency-response compensation of, 313–318
- Non-overlapping regions of convergence (ROC), 113
- Nonanticipative system, 22
- Noncausal window, 816
- Noncomputable network, 396–397
- Noninteger factor, changing sampling rate by, 190–193
- Nonlinear systems, 20
- n th-order Chebyshev polynomial, 556fn
- Number-theoretic transforms (NTTs), 789
- Nyquist frequency, 160
- Nyquist rate, 160
- Nyquist-Shannon sampling theorem, 160, 236
- Odd function, 55–56
- Odd sequence, 54–55
- Offset binary coding scheme, 212
- One-sided Fourier transform, decomposition of, 958
- One-sided z -transform, 100, 135
- One's complement, 415
- Optimum lowpass FIR filter, characteristics of, 568–570
- Overflow, 416
- Overflow oscillations:
defined, 462
in second-order system, 462–463
- Overlap-add method, 670, 777, 825, 826, 828
- Overlap-save method, 671, 777
- Overloaded quantizers, 214
- Oversampling:
in A/D conversion, 224–234
in analog-to-digital (A/D) conversion, 224–236
in D/A conversion, 234–236
and linear interpolation for frequency estimation, 809–810
oversampled A/D conversion with direct quantization, 225–229
- Oversampling ratio, 225
- Parallel-form structures, 393–395
illustration of, 394–395
- Parametric signal modeling, 890–941
all-pole modeling of signals, 891–895
autocorrelation matching property, 898
determination of the gain parameter G , 899–900
of finite-energy deterministic signals, 896–897
least-squares approximation, 892
least-squares inverse model, 892–894
linear prediction, 892
linear prediction formulation of, 895
linear predictive analysis, 892
minimum mean-squared error, 898
random signal modeling, 897
- all-pole spectrum analysis, 907–915
pole locations, 911–913
sinusoidal signals, 913–915
speech signals, 908–911
- applications, 891

- Parametric signal modeling, (*continued*)
 autocorrelation normal equations:
 Levinson-Durbin algorithm, derivation of, 917-920
 Levinson-Durbin recursion, 916-917
 solutions of, 915-919
 defined, 890
 deterministic and random signal models, 896-900
 estimation of correlation functions, 900-905
 autocorrelation method, 900-903
 comparison of methods, 904-905
 covariance method, 903-904
 equations for predictor coefficients, 905
 prediction error, 904
 stability of the model system, 905
 lattice filters, 920-926
 all-pole model lattice network, 923-924
 direct computation of k -parameters, 925-926
 prediction error lattice network, 921-923
 model order, 905-907
 selection, 906-907
 PARCOR coefficient, 916fn, 925
 Parks-McClellan algorithm, 494, 566-568, 568, 571, 579, 963
 Parseval's theorem, 58, 60, 96, 441, 446, 679, 707, 908
 Partial energy of an impulse response, 319
 Partial fraction expansion, inverse z -transform, 116-117, 120-121
 Periodic conjugate-antisymmetric components, 654
 Periodic conjugate-symmetric components, 654
 Periodic convolution, 61
 discrete Fourier series, 630-633
 Periodic discrete-time sinusoids, 15
 Periodic even components, 654
 Periodic impulse train, 1026
 Periodic odd components, 654
 Periodic sampling, 153-156, 236
 Periodic sequence, 14-15, 950, 954
 Periodogram, 837-843, 864
 computation of average periodograms using the DFT, 845
 defined, 838
 modified, 838
 periodogram analysis, example of, 837, 845-849
 periodogram averaging, 843-845
 properties of, 839-843
 smoothed, 851
 Periodogram analysis, 837
 Phase:
 defined, 14
 relationship between phase magnitude and, 301-305
 Phase distortions, 275
 Phase-lag function, 318
 Phase response, 275
 Plosive sounds, 830, 1024
 Poisson's formulas, 943
 Pole locations, all-pole spectrum analysis, 911-913
 Poles of quantized second-order sections, 427-429
 Polynomials, and alternation theorem, 558
 Polyphase components of $h[n]$, 197
 Polyphase implementation of decimation filters, 199-200
 Power density spectrum, 68
 Power series expansion, 122-124
 finite-length sequence, 122
 inverse transform by, 123
 by long division, 123
 Power spectral density, 225-228, 231-232, 236
 Power spectrum of quantization noise estimates, 855-860
 Power spectrum of speech estimates, 860-862
 Predictable random sinusoidal signals, 936fn
 Prediction coefficients, 895
 Prediction error residual, 895
 Prime factor algorithms, 747, 749
 Quadrature mirror filters, 203
 Quantization errors, 416
 analog-to-digital (A/D) conversion:
 analysis of, 214-220
 for a sinusoidal signal, 215-217
 analysis of, 214-215
 defined, 214
 for a sinusoidal signal, 215-217
 Quantization errors, analysis, 214-220
 Quantization noise, measurements of, 217-218
 Quantizers, 210-213
 linear, 211fn
 overloaded, 214
 Quasiperiodic voiced segments, 832
 Radar signals, time-dependent Fourier analysis of, 834-836
 Radix- m FFT algorithm, 783
 Random noise, 1026

- Random process, 65
autoregressive (AR) linear random process, 887
autoregressive moving-average (ARMA) linear random process, 887
moving-average (MA) linear random process, 887
- Random signal modeling, 897
- Random signals, 65–70
- Random sinusoidal signals, 936
- Range, 834
- Real cepstrum, 984fn
- Rectangular pulse, discrete Fourier series of, 644–646
- Rectangular windows, 535–537, 539
- Recursive computation, 38
- Reflection coefficients, 916fn
- Region of convergence (ROC), 101–102, 285–286
determining, 285–286
non-overlapping, 113
properties of for z -transform, 110–115
stability, causality and, 115
- Resampling, 179–180
consistent, 250
defined, 179
- Residual, 895, 905
- Right-sided exponential sequence, 103–104
- Rotation of a sequence, 650
- Rotations, 782
- Round-off noise in digital filters:
analysis of the direct-form IIR structures, 436–445
cascade IIR structure, analysis of, 448–453
direct-form FIR systems, analysis of, 453–458
effects of, 436–459
first-order system, 441–442
interaction between scaling and round-off noise, 448
scaling in fixed-point implementations of IIR systems, 445–448
second-order system, 442
- Sample mean, 837
- Sample variance, 837
- Sampled-data Delta-Sigma modulator, 220–221
- Sampling:
frequency-domain representation of, 156–163
in time and frequency, 819–822
- Sampling frequency/Nyquist frequency, 154
- Sampling period, 154
- Sampling rate:
changing by a noninteger factor, 190–193
increasing by an integer factor, 184–187
reduction by an integer factor, 180–184
- Sampling rate compressor, 180
- Sampling rate expander, 184
- Saturation overflow, 416
- Scaling, interaction between round-off noise and, 448
- Scaling property, 19
- Second-order z -transform, 118–120
- Seismic data analysis, 4
and multidimensional signal processing techniques, 4
- Sequence value, 723fn
- Sequences:
autocorrelation, 67–68
basic, 12–15
causal, 32
complex exponential, 14–15, 53–54
conjugate-antisymmetric, 54–55
conjugate-symmetric, 54–55
deterministic autocorrelation, 67
even, 54–55
exponential, 13–14, 947
finite-length, 946, 949–954
convolution of, 131
impulse, 13
left-sided exponential, 104–105
odd, 54–55
periodic, 14–15
right-sided exponential, 103–104
shifted exponential, 126
sinusoidal, 14
time-reversed exponential, 129
unit sample, 12–13
unit step, 12–13
- Sharpening, 609
- Shift-invariant system. *See* Time-invariant systems
- Shifted exponential sequence, 126
- Short-time Fourier transform. *See* Time-dependent Fourier transform
- Sifting property of the impulse function, 154
- Sign and magnitude, 415
- Sign bit, 415
- Signal flow graph representation of linear constant-coefficient difference equations, 382–388
- Signal interpretation, 3
- Signal modeling, 4–5

- Signal predictability, 3
 Signal processing, 2
 based on time-dependent Fourier transform, 825–826
 historical perspective, 5–8
 multidimensional, 4
 problems/solutions, 4
 Signal-processing systems, classification of, 10
 Signal-to-noise ratio (SNR), 3
 Signal-to-quantization-noise ratio, 219–220
 Signals:
 defined, 9
 mathematical representation of, 9
 Simulink, 3
 Sink nodes, 383
 Sinusoidal signals:
 all-pole spectrum analysis, 913–915
 defined, 218
 quantization error for, 215–217
 signal-to-quantization-noise ratio, 219–220
 for uniform quantizers, 220
 Smoothed periodogram, 851
 Source nodes, 383
 Specifications, filter design, 494–496
 Spectral analysis, 4
 Spectral sampling, effect of, 801–810
 illustration, 803–805
 Spectrograms, 814–815
 plotting, 832
 wideband, 832
 Spectrum analysis of random signals using
 autocorrelation sequence estimates, 849–862
 correlation and power spectrum estimates, 853–855
 power spectrum of quantization noise estimates, 855–860
 power spectrum of speech estimates, 860–862
 Speech model, 1024–1027
 speech model, estimating the parameters of, 1030–1032
 vocal tract, 1025
 voiced sounds, 1024
 Speech signals:
 all-pole spectrum analysis, 908–911
 time-dependent Fourier analysis of, 830–834
 Split-radix FFT (SRFFT), 781
 Square summability, 51, 65
 for the ideal lowpass filter, 51–52
 Stability, 22–23
 testing for, 23
 Stationary, use of term, 66fn
 Steady-state response, 46
 Suddenly-applied exponential:
 absolute summability for, 51–52
 inputs, 46–48
 Summability:
 absolute, 50–52, 65
 square-, 51, 65
 for the ideal lowpass filter, 51–52
 Superposition, principle of, 19, 20, 23, 980,
 1002–1003
 Surface acoustic wave (SAW), 753
 Switched-capacitor technologies, 2
 Symmetry properties:
 discrete Fourier series, 630, 653–654
 discrete Fourier transform (DFT), 653–654
 Fourier transform, 54–57
 illustration of, 56–57
 System function, 274–275
 determination of, from a flow graph, 386–387
 linear time-invariant (LTI) systems, 115, 132
 Tapped delay line structure, 401
 Telecommunications, and discrete-time signal
 processing, 8
 Time and frequency, sampling in, 819–822
 Time-dependent Fourier analysis of radar signals,
 834–836
 clutter, 835
 Doppler radar signals, 835–836
 Time-dependent Fourier synthesis, 822, 825
 Time-dependent Fourier transform, 792, 811–829
 defined, 811
 effect of the window, 817–818
 filter bank interpretation, 826–829
 of $X[n, \lambda]$, 816–817
 invertibility of $X[n, \lambda]$, 815–816
 of a linear chirp signal, 811–814
 overlap-add method of reconstruction, 822–825
 sampling in time and frequency, 819–822
 signal processing based on, 825–826
 spectrogram, 814–815
 Time-dependent Fourier transform of speech,
 spectral display of, 832–834
 Time-division multiplexing (TDM), 266
 Time invariance, 24
 Time-invariant systems, 20–21
 accumulator as, 21
 accumulators as, 21
 compressor system, 21

- Time-reversal property, z -transform, 129
time-reversed exponential sequence, 129
- Time-shifting property, z -transform, 125–126
shifted exponential sequence, 126
- Tolerance schemes, 494
- Transform analysis of linear time-invariant (LTI) systems, 274–373
- Transposed form, 397–401
for a basic second-order section, 398–399
for a first-order system with no zeros, 397–398
flow graph reversal, 397
transposition, 397–401
- Transposition, 397–401
- Transversal filter structure, 401
- Trapezoidal approximation, 606
- Twicing, 609
“Twiddle factors,” 731
- Two-port flow graph, 405
- Two-sided exponential sequence, 107–108, 135
- Two-sided z -transform, 100
- Two's complement, 415
- Type-I periodic symmetry, 675
- Type-II periodic symmetry, 675
- Type I FIR linear-phase systems, 329
example, 331–332
- Type II FIR linear-phase systems, 330
example, 332–333
- Type III FIR linear-phase systems, 330
example, 333–334
- Type IV FIR linear-phase systems, 330
example, 335
- Unbiased estimators, 837
- Unilateral z -transform, 100, 135–137
of an impulse, 135
nonzero initial conditions, effect of, 136–137
- Unit circle, 100–101
- Unit impulse function, 154
- Unit sample sequence, 12–13
- Unit step sequence, 12–13
- Unitary transforms, 676
- Unquantized filter, 423
- Unvoiced segments, 832
- Unwrapped phase, 349
- Upsampling filter design, 579–582
- Vocal tract, 830, 1025–1026
- Voiced segments, 832
- Voiced sounds, 830, 1024
- Welch estimate, 862
- White noise, 69, 443
- Whitening filter, 897
- Whitening procedure, 366
- Wideband spectrogram, 832
- Window:
method, 530
noncausal, 816
- Windowing, 792
Bartlett (triangular) windows, 536–539
Blackman windows, 536–539
commonly used windows, 535–536
design of FIR filters by, 533–545
and FIR filters, 533–545
incorporation of generalized linear phase, 538–541
Kaiser window filter design method, 541–553
properties of commonly used windows, 535–538
Hamming windows, 536–539
Hann windows, 536–539
rectangular windows, 535–537, 539
theorem, 61–62
- Winograd Fourier transform algorithm (WFTA), 749
- Yule–Walker equations, 896, 898, 916
- z -transform, 99–152, *See also Inverse z -transform*
bilateral, 100, 135
common pairs, 110
defined, 99–100
finite-length truncated exponential sequence, 109
infinite sum, 103
inverse, 115–124
inspection method, 116
inverse by partial fractions, 120–121
partial fraction expansion, 116–117
power series expansion, 122–124
second-order z -transform, 118–120
left-sided exponential sequence, 104–105
LTI systems and, 131–134
one-sided, 100, 135
properties, 124–131
conjugation property, 129
convolution property, 130–131
differentiation property, 127–129
exponential multiplication property, 126–127
linearity property, 124–125
summary of, 131
time-reversal property, 129
time-shifting property, 125–126

z-transform (continued)

- region of convergence (ROC), 101–102
- properties of, 110–115
- right-sided exponential sequence, 103–104
- sum of two exponential sequences, 105–107
- two-sided, 100
- two-sided exponential sequence, 107–108
- uniform convergence of, 102
- unilateral, 100, 135–137
 - of an impulse, 135
 - nonzero initial conditions, effect of, 136–137
- unit circle, 100–101

*z-transform operator $Z[\cdot]$, 100**Zero-input limit cycles:*

- avoiding limit cycles, 463
 - in fixed-point realizations of IIR digital filters, 459–463
 - limit cycles owing to overflow, 462–463
 - limit cycles owing to round-off and truncation, 460–462
- Zero-order hold, 222–223**
- compensation for, 571–575
- Zero-padding, 667**