

Index

Note: Page Numbers with “f” denote figures; “t” tables

A

Adaptive filters

- adaptive echo cancellers, 479–480, 479f
- electrocardiography interference cancellation, 476–478, 477f, 478f
- least mean square adaptive FIR filters
 - corrupted signal and noise reference, 455–457, 456f, 456t
 - desired signal spectrum, 453, 454f
 - noise canceller, 454, 454f
 - one-tap FIR filter, 455, 457
- linear prediction
 - line enhancement, 473–475, 473f, 474f, 475f
 - periodic interference cancellation, 476, 476f
- long-distance telephone circuit, 479, 479f
- noise cancellation, *see* Noise cancellation
- system modeling, 468, 468f
 - MATLAB program, 470
 - spectrum for, 470, 472f
 - unknown system’s frequency responses, 469, 470f
 - waveforms for, 470, 471f
- TMS320C6713 DSK, *see* TMS320C6713 DSK
- Wiener filter theory
 - autocorrelation and cross-correlation, 459
 - LMS algorithm, 461–462
 - mean square error quadratic function, 457–458, 458f
 - noise cancellation, 457, 457f
 - statistical expectation, 457–458, 461–462
 - steepest descent algorithm, 459, 460f, 461–462

Adaptive differential pulse code modulation (ADPCM)

- decoder, 512–514, 513f
- discrete function, 514–515, 515t
- encoder, 512–514, 513f
- FIR filter, 516–517
- input and output characteristics, 514, 514t
- 16-level nonuniform adaptive quantizer, 514
- performance measurement, 518
- predictor z-transfer function, 516–517
- scale factor, 514–515
- speech samples, 517–518, 517f

ADC, *see* Analog-to-digital conversion (ADC)

Address generators

- circular buffering, 409, 410f
- equivalent FIFO, 410f, 411
- FIR filtering, 411

Aliasing level, 28

Amplitude modulation (AM), 603, 603f

Amplitude spectrum

- DFT, 87, 88f, 97–101
- Fourier series, 89, 89f

Analog-to-digital conversion (ADC)

- binary codes, 40, 41f
- 2-bit flash ADC unit, 36, 36f
- implementation, 35–36
- oversampling, 587, 587f
 - benefits of, 586
 - continuous vs. regular sampled vs. oversampled signal amplitudes, 588–589, 591f
 - frequency response, 588–589, 589f
 - in-band frequency range, 587
 - MATLAB program, 589
 - oversampling ratio, definition, 585–586
 - quantization noise power, 586–588
 - regular ADC system, 586–587, 586f
 - time vs. frequency domains, 588–589, 590f

quantization

- bipolar quantizer, 38–40, 39f, 40t
 - definition, 35, 36f
 - error, 37
 - notations and rules, 38
 - process, 37
 - SNR, 47
 - unipolar quantizer, 38, 38f, 39t
- SDM ADC, *see* Sigma-delta modulation analog-to-digital conversion (SDM ADC)

Analog filters

- lowpass prototype transformation, 305, 306t
 - bandpass filter, 305, 306f
 - bandstop filter, 305, 306f
 - cutoff frequency, 304–305
 - highpass filter, 305, 305f
 - lowpass filter, 304, 304f
 - magnitude response, 304–305
 - MATLAB function, 307
- steady-state frequency response, 178

Analog μ -law companding

- characteristics, 502, 502f
- compressor, 501, 501f
- expander, 501–502, 501f
- original speech data, 504, 505f
- quantization error, 501

Analog signal processing

convolution, 798–799

Fourier series

amplitude and phase spectrum, 779

amplitude-phase form, 776

complex exponential form, 776–782, 784t

Fourier transform, 786–791, 789t, 790t

rectangular waveform, 780, 780f

sine-cosine form, 775, 780, 783t

Laplace transform

differential equations, 793–794

and table, 791–793, 791t

transfer function, 794–795

poles, zeros and stability, 795–796

sinusoidal steady-state response, 799–804

Analog system program, 482

Analog video

“back porch”, 748

electrical signal demodulation, 748–750, 749f

frame via row-wise scanning, 747

frequency modulation, 751

interlaced raster scanning, 747, 748f

NTSC TV standard, 750, 751f

PAL system, 752

QAM, 751–752

SECAM system, 752, 752t

vertical synchronization, 749f, 750

video data, retrace and sync layout, 750, 750f

video-modulated waveform, 747, 748f

Analysis filter

channel 0, 622, 623f

channel 1, 622, 623f

channel 2, 622–624, 624f

channel 3, 624, 625f

4-channel filter bank, 621–622, 622f

Anti-aliasing filter

aliasing level percentage, 28

Butterworth magnitude frequency response,
25–26

Sallen-Key lowpass filter, 26–27, 26f

sampled analog signal, 25, 26f

Anti-image filter

DAC unit, 30–31, 30f

sample-and-hold effect

digital equalizer, 32, 33f

and distortion percentage, 31f, 32

lowpass filtering effect, 30–31, 31f

shaping effect, 32, 33f

transfer function, 30–31

Application-specific integrated circuit (ASIC), 412

Auxiliary register arithmetic units (ARAUs),
428–429**B**

Bandpass filters

amplitude spectra, 203, 205f

analog lowpass filters, 321, 322f

design specifications, 389

digital Chebyshev lowpass prototype functions, 321, 322f,
331–337

digital fourth-order bandpass Butterworth filter, 203

frequency responses, 203, 203f

lowpass prototype transformation, 305, 306f

MATLAB program, 204

normalized filter, 187, 187f

original and filtered speech plots, 203, 204f

second-order bandpass filter, 352–354, 353f

Bandpass signals, undersampling, 603–608, 603f, 604f,
605f, 606f

Bandstop filters

digital Butterworth lowpass prototype functions,
331–337digital Chebyshev lowpass prototype functions,
331–337

lowpass prototype transformation, 305, 306f

normalized filter, 188, 188f

Bartlett window, 230, 231f

Bilinear transformation (BLT) design method, 391

analog filters, lowpass prototype transformation,
305, 306t

bandpass filter, 305, 306f

bandstop filter, 305, 306f

cutoff frequency, 304–305

highpass filter, 305, 305f

lowpass filter, 304, 304f

magnitude response, 304–305

MATLAB function, 307

design procedure, 303–304, 303f, 314–318, 316t

frequency warping, 312, 313f

digital frequency, 312

digital integration method, 308, 309f

graphical representation, 313, 314f

Laplace transfer function, 309–310

mapping properties, 310, 310f

s-plane vs. z-plane, frequency mapping, 312, 312f

z-transform, 309–310

Bipolar quantizer, 38–40, 39f, 40t

Blackman window, 230, 231f

BLT design method, *see* Bilinear transformation (BLT)
design methodBounded-in and bounded-out (BIBO) stability,
71–72, 71f

Butterworth filters, 338–340, 338t, 339t

see also Digital Butterworth lowpass prototype functions

Butterworth magnitude frequency response, 25–26

C

Cascade (series) realization method, 192, 195, 196f
 Causal system, 66–67
 CD recording system, *see* Compact-disc (CD) recording system
 Chebyshev filters, 388–389
 see also Digital Chebyshev lowpass prototype functions
 Chebyshev polynomial approximation, 269
 Chrominance channels, 688–689
 Circular convolution
 forward filter coefficients, 670
 reversed filter coefficients, 668
 Comité Consultatif International Téléphonique et Télégraphique (CCITT), 754
 Compact-disc (CD) recording system
 decoder, 8f, 9
 encoder, 7–9, 8f
 Companding
 analog μ -law companding, *see* Analog μ -law companding
 digital μ -law companding, *see* Digital μ -law companding
 Component video, 746
 Composite video, 746
 Compression, *see* Discrete cosine transform (DCT)
 see also image compression
 Conjugate quadrature filter (CQF), 630
 Continuous wavelet transform (CWT), 638, 641
 Convolution, 72–80, 798–799
 impulse response, 69
 linear, 142–143
 Cyclic redundancy check (CRC) code, 526

D

DAC, *see* Digital-to-analog conversion (DAC)
 DCT, *see* Discrete cosine transform (DCT)
 Decimation, 556
 Decimation filter, 581, 581f, 582t
 commutative model, 582–583, 583f
 filter bank coefficients, 582
 implementation, 582, 582f, 584
 three-tap decimation filter, 581
 Decimation-in-frequency method
 bit reversal process, 126, 127f
 eight-point FFT
 12 complex multiplications, 124–126, 125f
 first iteration, 123–124, 125f
 inverse of, 127–128, 127f
 second iteration, 124, 125f
 graphical operations, 123–124, 125f
 index mapping for, 126, 126t
 inverse FFT, definition, 126
 twiddle factor, 123–124
 Decimation-in-time method

 eight-point FFT algorithm, 128–129, 130f
 eight-point IFFT, 129–131, 131f
 first iteration, 128–129, 130f
 frequency bins, 128–129
 second iteration, 128–129, 130f
 Decomposition, *see* Two-channel perfect reconstruction
 quadrature mirror filter bank
 Delta modulation (DM), 511
 Denoise, 668, 670f
 DFT, *see* Discrete Fourier transform (DFT)
 Difference equation, 67–68, 79
 DSP system, input and output, 162, 162f
 filter() function, 165
 filtic() function, 165
 nonzero initial conditions, 165
 transfer function
 impulse response, 169
 step response, 169
 system response, 169–172
 z-transfer function, 166–167, 166f
 zero initial conditions, 165
 Differential pulse code modulation (DPCM)
 3-bit quantizer, 509, 510t
 direct-current coefficients, 736
 encoder and decoder, 509, 509f
 quantization step size, 512
 Digital-to-analog conversion (DAC), 47
 anti-image filter and equalizer, 30–31, 30f
 process, 40, 41f
 quantization error, 40–42
 quantization noise, 42
 quantized vs. original signal, 44f
 R-2R ladder DAC, 36–37, 37f
 SNR, 42
 Digital audio equalizer, 341f
 audio spectrum, 343–344, 343f
 audio test signal, 343–344
 filter banks design, 342, 342t
 magnitude frequency responses, 342, 342f
 MATLAB program, 344
 specifications for, 341, 341t
 Digital Butterworth lowpass prototype functions, 318, 319t
 magnitude response function, 318, 320f
 prototype filter order, 318–320
 Digital Chebyshev lowpass prototype functions, 318, 319t, 320t
 analog filter specification conversion, 321, 322t
 analog lowpass and bandpass filters, 321, 322f
 lowpass prototype order, 321
 magnitude response function, 320–321, 321f
 Digital convolution, 72–80
 Digital crossover design

Digital crossover design (*Continued*)

- lowpass and highpass filters
 - impulse responses, 260, 261f
 - magnitude frequency responses, 260, 260f, 261f
- speaker drivers, 258–259, 259f
- specifications, 259–260

Digital filtering system

- analog filter steady-state frequency response, 178
- difference equation, *see* Difference equation
- digital filters, *see* Digital filters
- Euler's formula, 179
- FIR and IIR systems, 186
- frequency response properties, 180
- inverse z-transform, 179
- magnitude frequency response, 178, 181
- normalized digital frequency, 178
- signal enhancement
 - biomedical signals, 199
 - ECG signal, notch filtering, 205–206, 206f, 207f, 208f
 - speech signals, *see* Speech signals
- sinusoidal inputs, system response, 180, 181f
- steady-state frequency responses, 178, 178f, 180
- system transient response, 178, 178f
- types

- freqz() function, 188
- normalized bandpass filter, 187, 187f
- normalized bandstop filter, 188, 188f
- normalized highpass filter, 187, 187f
- normalized lowpass filter, 186f, 187
- passband, stopband and transition band, 186
- z-plane pole-zero plot, 172f
 - analog-to-digital conversion, 174
 - bounded-in/bounded-out stability, 175
 - features, 172
 - Laplace shift property, 174
 - Laplace vs. z-transform, 173, 174f
 - s-plane vs. z-plane mapping, 175, 175f
 - stability rules, 175, 176f

Digital filters

- cascade (series) realization method, 192, 195, 196f
- direct-form II realization method, 192–195, 195f
- direct-form I realization method, 192–193, 194f
- parallel realization method, 192, 196–199, 196f
- sinusoidal steady-state response, 813f
 - inverse z-transform, 814
 - magnitude and phase response, 814
 - properties of, 815–816
 - z-transform output, 813

Digital μ -law companding

- 8-bit compressed PCM code format, 505–506, 506t, 508–509, 508f
- characteristics, 505, 506f

compressor and expander, 504, 505f

decoding table, 506–508, 507t

encoding table, 505–506, 507t

Digital signal processing (DSP), 1, 2f

- aliasing distortion, 2
- analog input signal, 2
- audio signals and spectrums, 3, 5f
- digital filtering, 3, 3f, 4f
- DS processor, 2
- real-world applications, 12, 12t
 - CD recording system, 7–9, 8f
 - data compressor, 7, 8f
 - data expander, 7, 8f
 - digital image enhancement, 9–12, 12f
 - interference cancellation, electrocardiography, 5–7, 7f
 - software audio players, 9
 - two-band digital crossover, 5, 6f
 - vibration signature analysis, 9, 10f, 11f
- signal frequency (spectrum) analysis, 3, 4f
- speech samples and spectrum, 4, 6f

Digital signal (DS) processor

- adder output, 429
- ASIC, 412
- features, 406, 411
- FIR filter, direct-form I implementation, 430, 430f
- fixed-point format, 411–412
 - 3-bit 2's complement number, 412–413, 412t, 413t
 - computational units, 427
 - C program, 445–446, 446t, 447f, 448f
 - fractional binary 2's complement system, 414
 - program control unit, 427
 - Q-30 format, 418, 418f
 - Q-format number, 415, 415f, 418
 - TMSC320C54x family architecture, 426–427, 426f

floating-point format, 411–412, 419, 419f

- advantages, 427
- ARAUs, 428–429
- C programs, 445, 445f
- IEEE format, 423–426, 423f, 425f
- overflow, 422
- rules for, 420
- speech quality applications, 429
- TMS320C3x processor, 427–428, 428f
- underflow, 423

hardware units

- address generators, *see* Address generators
- MAC, 408–409, 409f
- shifters, 409

Harvard architecture, 407, 407f

- execution cycle, 407, 408f
- pipelining operation, 408

- IIR filter
 - direct-form II implementation, 432, 432f
 - transfer function, 433
- linear buffering, *see* Linear buffering
- manufactures, 411
- real-time processing
 - input and output sample clock, 438, 439f
 - program segment, 438, 440f
 - TMS320C6713 DSK setup, 438, 440f
- scale factor, 429
- second-order section filters, 434
- TMS320C67x DSK, 436f, 437
 - C6713 DSK board, 434–436, 435f
 - memory and internal buses, 438
 - peripherals, 438
 - registers of, 437, 437f
 - software tool, 438
 - Texas Instruments Veloci™ architecture, 437
 - TMS320C6713 DSK, 438, 439f
- Von Neumann architecture, 406, 406f
 - applications, 408
 - execution cycles, 407, 408f
 - opcode and operand, 406
- Digital signals
 - BIBO stability, 71–72, 71f, 80
 - causal system, 66–67
 - difference equation format, 67–68, 79
 - digital convolution, 72–80
 - digital samples, 58, 58f
 - digital sequences, 61, 61f
 - analog signal function, 62, 79
 - exponential function, 60, 60f, 61t
 - sampling rate, 61
 - shifted unit-impulse and unit-step sequences, 59, 59f
 - sinusoidal function, 60, 60f, 60t
 - unit-impulse sequence, 58–59, 59f
 - unit-step sequence, 59, 59f, 62
- DS processor, 58
- impulse response
 - digital convolution sum, 69
 - FIR system, 69
 - IIR system, 71
 - unit-impulse response, 68, 68f
- linear system, 63–65, 64f
- notation of, 57–58, 58f
- time-invariant system, 65–66, 65f
- Direct-form II realization method, 192–195, 195f
- Direct-form I realization method, 192–193, 194f
- Discrete cosine transform (DCT), 519–522, 524–525, 525f
 - coefficients, 731, 732t
 - scan order, 732, 733t
- image compression
 - 2D-DCT, 729–731
 - JPEG image compression, *see* JPEG image compression
 - lossless/lossy compression, 728
 - principle of, 729
 - wavelet transform, *see* Wavelet transform
- Discrete Fourier transform (DFT), 625
 - amplitude spectrum, 87, 88f, 97–101
 - data window time, 97–101
 - definition, 88
- FFT
 - applications of, 97, 97f
 - data sequence, 101–102
 - decimation-in-frequency method, *see* Decimation-in-frequency method
 - decimation-in-time method, *see* Decimation-in-time method
 - digital sequence sample, 123
 - interpolated spectrum, 102–103
 - zero-padding effect, 102–103, 102f
- fft() and ifft() MATLAB functions, 93, 93t
- formula development, 91, 92f
- Fourier series, 132
- see also* Fourier series
 - amplitude spectral components, 90
 - coefficients, 88–89
 - periodic digital signal, 88, 89f
 - two-side line amplitude spectrum, 89, 89f
- frequency bin, 95
- frequency resolution, 96–101
- inverse of, 93
- phase spectrum, 97–101
- power spectrum, 97–101
- signal amplitude vs. sampling time instant, 87
- spectral estimation, window functions
 - Hamming window, 109–110, 111f
 - see also* Hamming window function
 - Hanning window, 109–110, 111f
 - periodic, continuous and band limited data, 107, 107f
 - rectangular window, 109–110, 111f
 - signal samples and spectra, 107–108, 108f
 - spectral leakage, 108
 - triangular window, 109–110, 111f
 - window operation, 108–109, 109f, 110f
- twiddle factor, 92–93
- Discrete wavelet transform (DWT)
 - discrete time function, 656–657
 - dwt() function, 671
 - dyadic subband coding structure, 657, 658f
 - IDWT, 656

Discrete wavelet transform (DWT) (*Continued*)
 idwt() function, 671
 lowpass and highpass filter coefficients, 656
 signal amplitude, 657
 4-tap Daubechies filters, frequency response, 656, 657f
 time-frequency plane, 661–664, 662f
 time-frequency plot, 661–664, 661f
 wavelet coefficients, 656
 wavelet expansion, 655

Downsampling, 557f
 data sequence, 556
 definition, 556
 MATLAB program, 559, 609
 normalized stop frequency edge, 556–557
 Nyquist sampling theorem, 556
 spectral plots, 556–557, 558f
 TMS320C6713 DSK, 608, 612f
 using anti-aliasing filter, spectral plots, 558–559, 560f
 without using anti-aliasing filter, spectral plots, 558, 559f
 z-transform, 556–557

DPCM, *see* Differential pulse code modulation (DPCM)

DSP, *see* Digital signal processing (DSP)

Dual-tone multifrequency (DTMF) tone generator, 442
 Goertzel algorithm, 392
 advantages, 380
 DFT algorithm, 377
 DFT coefficient, 378–379
 Euler's identity, 378–379
 MATLAB function, 382
 modified second-order Goertzel IIR filter, 380–381, 381f
 second-order Goertzel IIR filter, 310–311, 377
 transfer function, 377
 MATLAB program, 377
 modified Goertzel algorithm, 384f
see also Modified Goertzel algorithm
 ASCII code, 385–386
 design principles, 383
 frequency bins, 383, 384t
 MATLAB simulation, 385–386, 386f
 telephone touch keypads, 373–375, 373f, 376f

DWT, *see* Discrete wavelet transform (DWT)

E

Echo cancellation, 479–480, 479f

Edge detection, 717, 718f
 differential convolution kernel, 715–716
 grayscale image, 717, 719f
 horizontal Sobel edge detector, 716
 Laplacian edge detector, 716–717
 Laplacian of Gaussian filter, 717, 719f
 MATLAB functions, 718–721, 720f

vertical Sobel edge detector, 716

Electrocardiography (ECG)
 60-Hz hum eliminator and heart rate detection, 392
 cascaded frequency responses, 365, 366f
 characteristics of, 362, 363f
 design specifications, 364–365
 harmonics, 364
 heart rate, definition, 367–368
 MATLAB program, 368
 QRS complex, 362–364
 signal enhancement system, 364, 364f
 signal processing results, 366, 367f
 signal spectrum, 362, 363f
 transfer function and difference equation, 365
 zero-crossing algorithm, 366–367, 368f
 interference cancellation, 476–478, 477f, 478f

Equalizer, *see* Anti-image filter

Euler's identity, 378–379

Exponent, floating-point format, 419

F

Fast Fourier transform (FFT), 3–4
 applications of, 97, 97f
 data sequence, 101–102
 decimation-in-frequency method, *see* Decimation-in-frequency method
 decimation-in-time method, *see* Decimation-in-time method
 digital sequence sample, 123
 interpolated spectrum, 102–103
 zero-padding effect, 102–103, 102f

Father wavelet, 642, 644f

fconv() function, 670

fft() and ifft() MATLAB functions, 93, 93t

Finite impulse response (FIR) filter design, 69, 286t, 287
 coefficient accuracy effects, 282–285

Fourier transform design, 221t, 222, 290
 coefficient symmetry, 220
 desired impulse response, 220, 221f
 Fourier coefficients, 219
 Gibbs oscillatory behavior, 224, 229
 ideal lowpass filter, 219, 219f
 ideal lowpass frequency response, 219, 219f
 linear phase response, 223, 224f, 226–227, 226f, 227f
 magnitude and phase frequency responses, 224, 225f
 nonlinear phase response, 226, 227f
 periodic frequency response, 219
 sinusoidal sequence, 225
 symmetric coefficients, 224–225
 17-tap FIR lowpass filter coefficients, 224, 225t
 z-transfer function, 220

frequency sampling, 286, 817–820

- design procedure, 263
 - desired filter frequency response, 262, 262f
 - DFT, 262
 - features, 262
 - IDFT, 262
 - magnitude frequency response, 269
 - input-output relationship, 217
 - linear phase form, 281, 282f
 - noise reduction
 - clean signal and spectrum, 254, 255f
 - data acquisition process, 253
 - MATLAB program, 255
 - noise signal and spectrum, 254, 254f
 - passband frequency range, 254
 - speech noise reduction, 256–257, 256f, 257f
 - stopband frequency range, 254
 - vibration signals, 257–258, 258f, 259f
 - optimal design method, 286–287
 - see also* Parks-McClellan algorithm
 - transfer function, 218
 - transversal form, 280–281, 280f
 - two-band digital crossover
 - impulse responses, lowpass and highpass filters, 260, 261f
 - magnitude frequency responses, lowpass and highpass filters, 260, 260f, 261f
 - speaker drivers, 258–259, 259f
 - specifications, 259–260
 - window method, 285–286
 - see also* Window method
 - Finite precision, 35, 40–42
 - First-order IIR filter transfer function, 369–371
 - Fixed-point DS processor, 411–412
 - 3-bit 2's complement number, 412, 412t
 - fractional representation, 413, 413t
 - computational units, 427
 - direct-form II implementation, C code, 445–446, 446t, 447f, 448f
 - fractional binary 2's complement system, 414
 - program control unit, 427
 - Q-30 format, 418, 418f
 - Q-format number, 415, 415f, 418
 - TMSC320C54x family architecture, 426–427, 426f
 - Floating-point DS processor, 411–412, 419, 419f
 - advantages, 427
 - ARAUs, 428–429
 - direct-form I implementation, C code, 445, 445f
 - IEEE format
 - double precision format, 425, 425f
 - single precision format, 423–424, 423f
 - overflow, 422
 - rules for, 420
 - speech quality applications, 429
 - TMS320C3x processor, 427–428, 428f
 - underflow, 423
 - Folding frequency, 20, 47
 - Fourier series, 132
 - amplitude-phase form, 776
 - amplitude and phase spectrum, 779
 - amplitude spectral components, 90
 - coefficients, 88–89
 - complex exponential form, 776–782
 - waveform signals, 784t
 - Fourier transform, 786–791
 - properties, 790t
 - waveform signals, 789t
 - periodic digital signal, 88, 89f
 - rectangular waveform, 780, 780f
 - sine-cosine form, 775, 780
 - waveform signals, 783t
 - two-side line amplitude spectrum, 89, 89f
 - Frequency modulation (FM), 751
 - Frequency resolution, 96–101
 - Frequency sampling method, 286
 - design procedure, 263
 - desired filter frequency response, 262, 262f
 - DFT, 262, 817
 - Euler formula, 817
 - features, 262
 - frequency response, 819–820
 - IDFT, 262
 - L'Hospital's rule, 817
 - magnitude frequency response, 269
 - Frequency warping effect, 312, 313f
 - digital frequency, 312
 - digital integration method, 308, 309f
 - graphical representation, 313, 314f
 - Laplace transfer function, 309–310
 - mapping properties, 310, 310f
 - s-plane vs. z-plane, frequency mapping, 312, 312f
 - z-transform, 309–310
- ## G
- Gaussian filter kernel, 711–712
 - Gibbs effect, 224
 - Goertzel algorithm, 392
 - advantages, 380
 - DFT algorithm, 377
 - DFT coefficient, 378–379
 - Euler's identity, 378–379
 - MATLAB function, 382
 - modified second-order Goertzel IIR filter, 380–381, 381f
 - second-order Goertzel IIR filter, 310–311, 377
 - transfer function, 377

Grayscale histogram and equalization
 equalized grayscale image, human neck, 695, 697f
 new pixel value, 693–695
 original grayscale image, human neck, 695, 696f
 pixel value distribution, 692–693

H

Hamming window function, 230, 231f
 ECG data, 118f, 119
 seismic data, 116, 118f
 speech data, 116, 117f
 vibration signal, 119, 119f, 121f, 122f
 vibration signature analysis, gearbox, 119–120, 120f

Hanning window, 230, 231f

Harvard architecture, 407, 407f

 execution cycle, 407, 408f
 pipelining operation, 408

High-definition TV (HDTV) formats, 754, 754t

Highpass filters

 coefficients, 656
 digital Butterworth lowpass prototype functions, 322–331
 digital Chebyshev lowpass prototype functions, 322–331, 322f
 impulse responses, 260, 261f
 lowpass prototype transformation, 305, 305f
 magnitude frequency responses, 260, 260f, 261f

Histogram equalization, 9–12

Horizontal Sobel edge detector, 716

Huffman coding, 528, 737, 737t

I

IDWT, *see* Inverse discrete wavelet transform (IDWT)

IEEE floating-point format

 double precision format, 425, 425f
 single precision format, 423–424, 423f

IIR filter design, *see* Infinite impulse response (IIR) filter design

Image processing

 24-bit color image equalization, 695, 698f
 equalized RGB color image, 698, 699f
 histogram equalization method, 698, 699f
 original RGB color image, 698, 698f
 RGB channels, equalization effects, 698–699, 700f
 8-bit indexed color image equalization, 700–701, 701f, 702f
 compression, DCT
 2D-DCT, 729–731
 JPEG image compression, *see* JPEG image compression
 lossless/lossy compression, 728
 principle of, 729

 wavelet transform coding, *see* Wavelet transform coding

2D-DFT, 725

definition, histogram, 692

edge detection, 717, 718f

 differential convolution kernel, 715–716

 grayscale image, 717, 719f

 horizontal Sobel edge detector, 716

 Laplacian edge detector, 716–717

 Laplacian of Gaussian filter, 717, 719f

 MATLAB functions, 718–721, 720f

 vertical Sobel edge detector, 716

grayscale histogram and equalization

 equalized grayscale image, human neck, 695, 697f

 new pixel value, 693–695

 original grayscale image, human neck, 695, 696f

 pixel value distribution, 692–693

image level adjustment

 display level adjustment, 707

 linear level adjustment, 704–706, 705f, 706f

 MATLAB functions, 707, 708f

lowpass noise filtering

 average convolution kernel, 709

 Gaussian filter kernel, 711–712

 noisy and enhanced image, 711–712, 711f, 712f, 713f

MATLAB functions, equalization, 702–704, 703f

median filtering

 enhanced image, 714, 715f

 “pepper and salt” noise, 714, 715f

 principle of, 712–714

notation and data formats

 8-bit color image, 687, 687f

 24-bit color image, 686, 686f

 8-bit grayscale image, 684–685, 685f

 chrominance channels, 688–689

 format conversion, 690–691, 691f

 grayscale image conversion, RGB-to-YIQ transformation, 690, 690f

 image pixel notation, 684, 685f

 intensity image, 688, 688f

 luminance channel, 688–689

 spatial resolution, 684

 transformation and inverse transformation, 688–689

pseudo-color generation and detection

 grayscale to pseudo-color pixel, 722, 722f

 MATLAB code, 725

 procedure for, 724f

 sine functions, RGB transformations, 722, 723f

video sequence creation, 745–746, 746f, 747f

video signals, *see* Video signals

Impulse function, 374

- Impulse-invariant design method, 345f, 389, 392
 - filter DC gain, 348
 - inverse Laplace transform, analog impulse function, 345–346
 - rectangular approximation, 346–348
 - sampling interval effect, 348, 349f
 - scaled magnitude frequency response, 347f, 348
 - second-order filter design, 348–351
- Impulse response system
 - digital convolution sum, 69
 - FIR system, 69
 - IIR system, 71
 - unit-impulse response, 68, 68f
- Infinite impulse response (IIR) filter design, 71, 389, 390t, 391f
 - bandpass filter design specifications, 389
 - BLT design method, 388
 - see also* Bilinear transformation (BLT) design method
 - C code
 - direct-form II implementation, 445–446, 446t, 447f, 448f
 - direct-form I structure, 445, 445f
 - difference equation, 302–303
 - digital audio equalizer, 341f
 - audio spectrum, 343–344, 343f
 - audio test signal, 343–344
 - filter banks design, 342, 342t
 - magnitude frequency responses, 342, 342f
 - MATLAB program, 344
 - specifications for, 341, 341t
 - digital Butterworth lowpass prototype functions, 318, 319t
 - bandpass and bandstop filter, 331–337
 - lowpass and highpass filters, 322–331
 - magnitude response function, 318, 320f
 - prototype filter order, 318–320
 - digital Chebyshev lowpass prototype functions, 318, 319t, 320t
 - analog filter specification conversion, 321, 322t
 - bandpass filters, 321, 322f, 331–337
 - bandstop filters, 331–337
 - highpass filters, 322–331, 322f
 - lowpass filters, 321–331, 322f
 - lowpass prototype order, 321
 - magnitude response function, 320–321, 321f
 - direct-form I and direct-form II, realization structure, 358–360
- DTMF tone generator
 - Goertzel algorithm, *see* Goertzel algorithm
 - MATLAB program, 377
 - modified Goertzel algorithm, *see* Modified Goertzel algorithm
 - telephone touch keypads, 373–375, 373f, 376f
- first-order IIR filter transfer function, 369–371
- fixed-point system, 432, 432f
- format of, 302–303
- higher order IIR filter design, cascade method, 338–340, 338t, 339t
 - realization structure, 361–362
- 60-Hz hum eliminator and heart rate detection,
 - electrocardiography, 392
- cascaded frequency responses, 365, 366f
- characteristics of, 362, 363f
- design specifications, 364–365
- harmonics, 364
- heart rate, definition, 367–368
- MATLAB program, 368
- QRS complex, 362–364
- signal enhancement system, 364, 364f
- signal processing results, 366, 367f
- signal spectrum, 362, 363f
- transfer function and difference equation, 365
- zero-crossing algorithm, 366–367, 368f
- impulse-invariant design method, 345f, 389, 392
 - filter DC gain, 348
 - inverse Laplace transform, analog impulse function, 345–346
 - rectangular approximation, 346–348
 - sampling interval effect, 348, 349f
 - scaled magnitude frequency response, 347f, 348
 - second-order filter design, 348–351
- pole-zero placement method, 389, 392
 - first-order highpass filter, 357–358, 357f
 - first-order lowpass filter, 355–357, 355f, 356f
 - magnitude response, 351, 352f
 - Nyquist limit, 351–352
 - second-order bandpass filter, 352–354, 353f
 - second-order bandstop (notch) filter, 354–355, 354f
- second-order IIR filter transfer function, 369–371
- single-tone generator, 374–375, 374f, 375f
- transfer function, 433
- Infinite precision, 35, 282
- Interlaced raster scan, 747, 748f
- Interpolation filter, 579, 579t
 - commutative model, 580, 580f
 - filter bank coefficients, 579–580
 - four-tap interpolation filter, 578, 578f
 - implementation, 579, 579f, 584
- Inverse discrete cosine transform (IDCT), 729
- Inverse discrete Fourier transform (IDFT), 262
- Inverse discrete wavelet transform (IDWT),
 - 656, 671
- Inverse fast Fourier transform (IFFT)
 - definition, 126
 - eight-point IFFT, 129–131, 131f

Inverse z-transform

- definition, 144
- partial fraction expansion method, 144–145
 - constant(s) formulas, 145, 146t
 - MATLAB function `residue()`, 150–152

J

JPEG image compression

- alternating-current coefficients, 738f
 - bit stream, 738
 - run-length coding, 736–738
- direct-current coefficients
 - DPCM, 736
 - Huffman coding, 737, 737t
- encoder, 735, 735f
- image blocks, 735
- lossless entropy coding, 737
- quantization, 735–736, 736t
- RGB to YIQ transformation, 735
- two-dimensional grayscale image, 733, 734f
 - coding error, 733, 734t
 - DCT coefficients, 731, 732t
 - DCT coefficient scan order, 732, 733t
 - JPEG vector, 732–733
 - normalized DCT coefficients, 732, 733t
 - original image, 731, 732f
 - quality factor, 731, 733t
 - recovered image subblock, 733, 734t
 - 8×8 subblock, 731, 731t

K

Kaiser window, 230

Kernel

- average convolution, 709
- differential convolution, 715–716
- Gaussian filter, 711–712

L

Laplace shift property, 174

Laplace transform

- differential equations, 793–794
- and table, 791–793, 791t
- transfer function, 794–795

Laplacian edge detector, 716–717

Laplacian of Gaussian filter, 717, 719f

Least mean square (LMS) algorithm, 461–462

- adaptive FIR filters
 - corrupted signal and noise reference, 455–457, 456f, 456t
 - desired signal spectrum, 453, 454f
 - noise canceller, 454, 454f
 - one-tap FIR filter, 455, 457

Linear buffering

- FIR filter, 441–442, 441f
- IIR filter, 442, 443f
 - coefficient buffer, 444, 444f
 - digital oscillation, 442

Linear convolution, 142–143

Linear midtread quantizer, 533

Linear phase response, 223, 224f, 226–227, 226f, 227f

Linear systems, 63, 64f

- digital amplifier, 64
- system output, 65

Linear time invariant system

- difference equation, 67
- FIR system, 69
- stability criterion, 71
- unit-impulse response, 68–69, 68f, 72

Lowpass filters, 304, 304f

- analog filters, 321, 322f
- coefficients, 656
- digital Butterworth lowpass prototype functions, 322–331
- digital Chebyshev lowpass prototype functions, 321–331, 322f
- impulse responses, 260, 261f
- magnitude frequency responses, 260, 260f, 261f
- Sallen-Key lowpass filter, 26–27, 26f
- 17-tap FIR lowpass filter coefficients, 224, 225t

Lowpass noise filtering

- average convolution kernel, 709
- Gaussian filter kernel, 711–712
- noisy and enhanced image, 711–712, 711f, 712f, 713f
- Luminance channel, 688–689

MMAC, *see* Multiplier and accumulator (MAC)

Maclaurin series expansion, 593–595

Macroblocks, 755, 755f

Mathematical formulas

- complex conjugate, 826
- complex number form, 825–826
 - addition and subtraction, 826
 - division, 826–828
 - multiplication, 826
- L'Hospital's rule, 828
- quadratic equation solution, 828
- simultaneous equation solution, 828
- simultaneous linear equation solution, 830

Matrix Laboratory (MATLAB) programs

- ADPCM coding, 539
 - decoding, 542
 - encoding, 539
- analog filters, lowpass prototype transformation, 307

- arrays and indexing, 770–771
- CD audio player, 572
- commands and syntax
 - array operations, 769
 - complex numbers, 768
 - numbers, variables and expressions, 768
 - sum() function, 767
 - variable names, 768
- DCT waveform coding, 546
- digital audio equalizer, 344
- digital μ -law compressor, 537
- digital μ -law encoding and decoding, 537
- digital μ -law expander, 538
- downsampling, 559, 609
- DTMF tone generator, 377
- edge detection, 718–721, 720f
- equalization, 702–704, 703f
- fft() and ifft() function, 93, 93t
- FIR filter design
 - noise reduction, 255
 - window method, 237–240, 237t, 288
- first-order SDM, 597
- Goertzel algorithm, 382
- 60-Hz hum eliminator and heart rate detection,
 - electrocardiography, 368
- image level adjustment, 707, 708f
- μ -law companding, 535
- μ -law encoding and decoding, 534
- μ -law expanding, 535
- midtread quantizer
 - decoding, 536
 - encoding, 536
 - linear, 533
- modified Goertzel algorithm, 385–386, 386f
- noise cancellation, 466
- noninteger factor L/M , 568
- one-level wavelet transform and compression, 742
- oversampling, 589
- plot functions, 771–772, 772f
- pseudo-color generation and detection, 725
- residue() function, 150–152
- script files, 164f, 772–773
- signal to quantization noise ratio, 48, 537
- sign function, 548
- speech signals
 - bandpass filtering, 204
 - pre-emphasis of, 201
- sumsub.m function, 773–774
- system modeling, adaptive filters, 470
- two-channel perfect reconstruction quadrature mirror filter
 - bank, 633
- two-level wavelet transform and compression, 744
- uniform quantization decoding, 48
- uniform quantization encoding, 48
- upsampling, 564, 611
- wavelet data compression, 667
- W-MDCT function, 545
 - inverse function, 545
 - waveform coding, 546
- MDCT, *see* Modified discrete cosine transform (MDCT)
- Mean square error quadratic function, 457–458, 458f
- Median filtering
 - enhanced image, 714, 715f
 - “pepper and salt” noise, 714, 715f
 - principle of, 712–714
- Modified discrete cosine transform (MDCT)
 - 1D-DCT, 522
 - decoding stage, 523
 - encoding stage, 523
 - W-MDCT, 522, 522f
 - waveform coding, 524–525, 525f
 - wmdeth() and wimdeth() functions, 523–524
- Modified Goertzel algorithm, 384f
 - ASCII code, 385–386
 - design principles, 383
 - frequency bins, 383, 384t
 - MATLAB simulation, 385–386, 386f
- Mother wavelet, 642, 644, 644f
- Motion estimation, 755–756, 755f
- Motion vector, 755
- MPEG audio
 - audio frame formats, 526, 527f
 - data frame types, 526, 526f
 - DCT, 519–522, 524–525, 525f
 - encoder, 527, 528f
 - Huffman coding, 528
 - MDCT, *see* Modified discrete cosine transform (MDCT)
- Multiplier and accumulator (MAC), 407–409, 409f
- Multirate digital signal processing, 555–556
 - CD audio player
 - interpolation filter design, 571–572, 573f
 - MATLAB program, 572
 - sample rate conversion, 571, 571f
 - signal plots, 572, 574f
 - multistage decimation, *see* Multistage decimation approach
 - sampling rate, integer factor, *see* Sampling rate
- Multiresolution analysis, 650–651
- Multistage decimation approach
 - sampling rate conversion, 578
 - two-stage decimator, 574, 575f
 - filter requirements, 576
 - stopband frequency edge, anti-aliasing filter, 575–576, 575f

N

Noise cancellation

- MATLAB program, 466
- MSE function vs. weights, 465, 465f
- one-tap adaptive filter, 462, 463f
- specifications, 466
- speech waveforms and spectral plots, 466, 466f, 467f
- two-tap adaptive filter, 463–465

Noise reduction systems

- clean signal and spectrum, 254, 255f
- data acquisition process, 253
- MATLAB program, 255
- noise signal and spectrum, 254, 254f
- passband frequency range, 254
- speech noise reduction, 256–257, 256f, 257f
- stopband frequency range, 254
- vibration signals, 257–258, 258f, 259f

Noncausal FIR filter coefficients, 233

Noncausal sequence, 141, 233

Normalized bandpass filter, 187, 187f

Normalized bandstop filter, 188, 188f

Normalized Butterworth function, 805–808

Normalized Chebyshev function, 808–812

Normalized highpass filter, 187, 187f

Normalized lowpass filter, 186f, 187

Notch filter, 354–355, 354f

NTSC TV standard, 750, 751f

Nyquist frequency, 20, 47

Nyquist limit, 351–352, 562–563, 564f

O

Optimal design method, 286–287

see also Parks-McClellan algorithm

Overflow, 422

Output digital signal, 2, 592

P

Parallel realization method, 192, 196–199, 196f

Parks-McClellan algorithm

- alternation theorem, 277–278
- approximation error, 269
- Chebyshev polynomial approximation, 269
- Chebyshev real magnitude function, 277
- design procedure, 270–279
- disadvantages, 279
- magnitude frequency response, 269–270, 270f
- minimax filters, 269
- Remez exchange algorithm, 269

Partial fraction expansion method, 144–145

- constant(s) formulas, 145, 146t
- MATLAB function `residue()`, 150–152

Perfect reconstruction, *see* Two-channel perfect

reconstruction quadrature mirror filter bank

Phase alternative line (PAL) system, 752

Plot functions, 771

Pole-zero placement method, 389, 392

- first-order highpass filter, 357–358, 357f
- first-order lowpass filter, 355–357, 355f, 356f
- magnitude response, 351, 352f
- Nyquist limit, 351–352

second-order bandpass filter, 352–354, 353f

second-order bandstop (notch) filter, 354–355, 354f

Pole-zero plot, *see* Z-plane pole-zero plot

Polyphase filters

- direct decimation process, 581, 581f, 582t
 - commutative model, 582–583, 583f
 - filter bank coefficients, 582
 - implementation, 582, 582f, 584
 - three-tap decimation filter, 581
- direct interpolation filter, 579, 579t
 - commutative model, 580, 580f
 - filter bank coefficients, 579–580
 - four-tap interpolation filter, 578, 578f
 - implementation, 579, 579f, 584
- properties, 581

Power spectrum, 97–101

Progressive scan, 754

Q

Quadrature amplitude modulation (QAM), 751–752

Quantization, 735–736, 736t

see also Waveform quantization and compression

- bipolar quantizer, 38–40, 39f, 40t
- definition, 35, 36f
- error, 37
- notations and rules, 38
- process, 37
- SNR, 47
- unipolar quantizer, 38, 38f, 39t

Quantization error

- analog μ -law companding, 501
- DAC, 40–42

R

Radix-2 FFT algorithm, 123

decimation-in-frequency method, *see* Decimation-in-frequency method

decimation-in-time method

- eight-point FFT algorithm, 128–129, 130f
- eight-point IFFT, 129–131, 131f
- first iteration, 128–129, 130f
- frequency bins, 128–129
- second iteration, 128–129, 130f

rconv() function, 668
 Realization structure
 direct-form I and direct-form II, 358–360
 higher order IIR filter design, cascade method, 361–362
 Real-time processing
 input and output sample clock, 438, 439f
 program segment, 438, 440f
 TMS320C6713 DSK setup, 438, 440f
 Rectangular window, 230
 Reference frame, 755, 755f
 Remez exchange algorithm, 269
 RGB components, 686, 686f
 RGB-to-YIQ transformation, 690, 690f
 Root mean square (RMS), 42, 499
 Rounded off error, 282
 Run-length coding, 736–738

S

Sallen-Key lowpass filter, 26–27, 26f
 Sampling rate
 downsampling, 557f
 data sequence, 556
 definition, 556
 MATLAB program, 559, 609
 normalized stop frequency edge, 556–557
 Nyquist sampling theorem, 556
 spectral plots, 556–557, 558f
 TMS320C6713 DSK, 608, 612f
 using anti-aliasing filter, spectral plots, 558–559, 560f
 without using anti-aliasing filter, spectral plots, 558, 559f
 z-transform, 556–557
 noninteger factor L/M, 570–571
 anti-aliasing filter, 568, 569f
 interpolation filter, 567, 568f
 MATLAB program, 568
 sampling rate conversion, 567, 567f
 upsampling
 definition, 562
 interpolation filter, 563, 565f
 MATLAB program, 564, 611
 normalized stop frequency edge, 562–563
 Nyquist limit, 562–563, 564f
 process of, 562, 563f
 sampling frequency, 562–563
 TMS320C6713 DSK, 611, 612f
 Scaling functions, 649–650, 650f
 multiresolution analysis, 650–651
 SECAM system, *see* Séquentiel Couleur à Mémoire (SECAM) system
 Second-order bandpass filter, 352–354, 353f
 Second-order bandstop (notch) filter, 354–355, 354f
 Second-order Butterworth lowpass filter, 25–26
 Second-order IIR filter transfer function, 369–371
 Séquentiel Couleur à Mémoire (SECAM) system, 752, 752t
 Sequential search method, 755–756
 Shannon sampling theorem, 20
 Shaped-in-band noise power, 593–595
 Shifters, 409
 Sigma-delta modulation analog-to-digital conversion (SDM ADC)
 ADC resolution, 595–596
 CD player, 601–602, 601f, 602f
 continuous vs. regular sampled vs. oversampled signal amplitudes, 597, 599f
 discrete-time analog filter, 592, 593f
 DSP model, second-order SDM, 595, 596f
 extrapolation method, 592
 feedback control system, 592–593
 first-order SDM
 DSP model, 592, 593f
 MATLAB program, 597
 principles, 592, 592f
 frequency responses, 597, 597f
 MAX1402, functional diagram, 600, 600f
 noise shaping filter, 592–593, 594f
 shaped-in-band noise power, 593–595
 time vs. frequency domains, 597, 598f
 Signal denoising, 668, 670f
 Signal-to-noise power ratio, 499
 Signal reconstruction
 aliasing frequency component, 23–25
 anti-aliasing filtering, 35
 aliasing level percentage, 28
 Butterworth magnitude frequency response, 25–26
 Sallen-Key lowpass filter, 26–27, 26f
 sampled analog signal, 25, 26f
 anti-image filter and equalizer, *see* Anti-image filter and equalizer
 signal notations, 21–22, 22f
 signal spectrum recovery, 22–23, 22f, 23f
 Signal sampling
 ADC
 see also Analog-to-digital conversion (ADC)
 sample-and-hold analog voltage, 15, 16f
 analog (continuous) signal and digital samples vs. time instants, 15, 16f
 anti-image filter, 18
 DAC, *see* Digital-to-analog conversion (DAC)
 DSP, 15, 16f
 lowpass reconstruction filter, 20
 MATLAB function
 signal to quantization noise ratio calculation, 48
 uniform quantization decoding, 48

- Signal sampling (*Continued*)
 - uniform quantization encoding, 48
 - Nyquist frequency/folding frequency, 20, 47
 - sampling process, 18, 18f, 47
 - sampling rate, 16–17
 - sampling theorem condition, 17–18, 17f, 20, 47
 - Shannon sampling theorem, 20
 - signal reconstruction, *see* Signal reconstruction
 - spectral analysis, 18–19, 19f
 - Single-tone generator, 374–375, 374f, 375f
 - Smith-Barnwell PR-CQF filters, 630, 630t
 - Spectral leakage, 108
 - Speech coding
 - four-band compression, 637, 637f, 638f
 - seismic data, 637, 639f
 - two-band compression, 636–637, 636f
 - Speech noise reduction, 256–257, 256f, 257f
 - Speech signals
 - bandpass filtering
 - amplitude spectra, 203, 205f
 - digital fourth-order bandpass Butterworth filter, 203
 - frequency responses, 203, 203f
 - MATLAB program, 204
 - original and filtered speech plots, 203, 204f
 - pre-emphasis of
 - amplitude spectral plots, 201, 202f
 - magnitude and phase responses, 200, 200f
 - MATLAB program, 201
 - speech waveforms, 200, 201f
 - transfer function, 200
 - Stair functions, 771
 - Steepest descent algorithm, 459, 460f, 461–462
 - Stem functions, 771
 - Step response, 169
 - Subband coding
 - analysis and synthesis stages
 - channel 0, 622, 623f
 - channel 1, 622, 623f
 - channel 2, 622–624, 624f
 - channel 3, 624, 625f
 - 4-channel filter bank analyzer and synthesizer, 621–622, 622f
 - decomposition, *see* Two-channel perfect reconstruction
 - quadrature mirror filter bank
 - delta function, 624
 - discrete Fourier transform, 625
 - filter bank system, 621
 - impulse train, 625, 626f
 - signal flow, 624, 625f
 - speech coding, *see* Speech coding
 - two-band filter bank system, signal compression, 635–636, 636f
 - z-transform, 626
 - Subplot functions, 771
 - S-video, 746
 - Synthesis filter
 - channel 0, 622, 623f
 - channel 1, 622, 623f
 - channel 2, 622–624, 624f
 - channel 3, 624, 625f
 - 4-channel filter bank, 621–622, 622f
- ## T
- 17-Tap FIR lowpass filter coefficients, 224, 225t
 - Target frame, 755, 755f
 - Time-invariant system, 65–66, 65f
 - TMS320C6713 DSK
 - analog system program, 482
 - downsampling, 608, 612f
 - system modeling
 - LMS adaptive filter, 480–482, 480f, 482f
 - program segment, 481
 - tonal noise cancellation, 483, 483f, 484f
 - DSK1 program, 483
 - DSK2 program, 484
 - upsampling, 611, 612f
 - Transition band, 186
 - Translated function, 642, 643f
 - Transversal FIR filter, 280–281, 280f
 - Twiddle factor, 92–93, 123–124
 - Two-band digital crossover design
 - lowpass and highpass filters
 - impulse responses, 260, 261f
 - magnitude frequency responses, 260, 260f, 261f
 - speaker drivers, 258–259, 259f
 - specifications, 259–260
 - Two-channel perfect reconstruction quadrature mirror filter
 - bank, 626, 627f
 - analysis and synthesis filters, 627
 - autocorrelation function, 628
 - four-band implementation
 - binary tree structure, 634f, 635
 - dyadic tree structure, 635, 635f
 - frequency response, 629, 629f
 - lowpass filter equations, 630
 - MATLAB program, 633
 - N-tap FIR filters, 628
 - Smith-Barnwell PR-CQF filters, 630, 630t
 - two-band analysis and synthesis, 632, 633f
 - Two-dimensional discrete cosine transform (2D-DCT), 729–731
 - Two-dimensional discrete Fourier transform (2D-DFT), 725

U

Unipolar quantizer, 38, 38f, 39t
 Unit circle, 175
 Unit-impulse sequence, 58–59, 59f
 Unit-step sequence, 59, 59f, 62
 Underflow, 423
 Unstable system, 175
 Upsampling
 definition, 562
 interpolation filter, 563, 565f
 MATLAB program, 564, 611
 normalized stop frequency edge, 562–563
 Nyquist limit, 562–563, 564f
 process of, 562, 563f
 sampling frequency, 562–563
 TMS320C6713 DSK, 611, 612f

V

Vertical retrace, 747, 750
 Vertical Sobel edge detector, 716
 Vibration signature analysis, 9, 10f, 11f
 Video signals
 analog video
 “back porch”, 748
 electrical signal demodulation, 748–750, 749f
 frame via row-wise scanning, 747
 frequency modulation, 751
 interlaced raster scanning, 747, 748f
 NTSC TV standard, 750, 751f
 PAL system, 752
 QAM, 751–752
 SECAM system, 752, 752t
 vertical synchronization, 749f, 750
 video data, retrace and sync layout, 750, 750f
 video-modulated waveform, 747, 748f
 component video, 746
 composite video, 746
 digital video
 CCIR-601, chroma subsampling, 753, 753f
 HDTV formats, 754, 754t
 specifications, 754, 754t
 motion estimation, 755–756, 755f
 S-video, 746
 Von Neumann architecture, 406, 406f
 applications, 408
 execution cycles, 407, 408f
 opcode and operand, 406

W

Waveform coding, 7
 Waveform quantization and compression
 analog μ -law companding

 characteristics, 502, 502f
 compressor, 501, 501f
 expander, 501–502, 501f
 original speech data, 504, 505f
 quantization error, 501
 digital μ -law companding
 8-bit compressed PCM code format, 505–506, 506t,
 508–509, 508f
 characteristics, 505, 506f
 compressor and expander, 504, 505f
 decoding table, 506–508, 507t
 encoding table, 505–506, 507t
 DM, 511
 DPCM
 3-bit quantizer, 509, 510t
 encoder and decoder, 509, 509f
 quantization step size, 512
 G.721 modulation, *see* Adaptive differential pulse code
 modulation (ADPCM)
 linear midtread quantization
 characteristics of, 498–499, 498f
 quantization, definition, 497–498
 quantization error, 500
 quantized values, 498–499, 498t
 signal-to-noise power ratio, 499
 speech data plot, 500, 500f
 MATLAB programs, *see* MATLAB programs
 MPEG audio
 audio frame formats, 526, 527f
 data frame types, 526, 526f
 DCT, 519–522, 524–525, 525f
 encoder, 527, 528f
 Huffman coding, 528
 MDCT, *see* Modified discrete cosine transform (MDCT)
 TMS320C6713 DSK
 digital μ -law encoding and decoding, 530
 encoding and decoding, linear quantization,
 528–529
 Wavelet analysis
 analysis equations, 822–823
 properties, 821–822
 scaled function, 641, 642f, 643f
 synthesis equations, 823–824
 translated function, 642, 643f
 Wavelet transform
 amplitudes, 639–641, 641f
 analysis and synthesis stage, 664
 combined signal and spectrum, 639–641, 640f
 CWT, 638, 641
 Daubechies-4 filter coefficients, 653, 654t
 DWT, 638
 see also Discrete wavelet transform (DWT)

- Wavelet transform (*Continued*)
 - coefficient layout, 664, 665f
 - hard threshold, 668, 669f
 - signal denoising, 668, 670f
 - Haar father and mother wavelets, 642, 644, 644f, 652–653
 - individual signal components, 639, 640f
 - mother wavelet, definition, 641
 - one-level wavelet transform and compression, 741, 742f
 - MATLAB program, 742
 - scaled wavelet function, 641, 642f, 643f
 - scaling functions, 649–650, 650f
 - multiresolution analysis, 650–651
 - signal coding, 650, 651f
 - sinusoidal delaying function, 648, 648f
 - 4-tap Daubechies father wavelet, 654, 654f
 - 4-tap Daubechies mother wavelet, 655, 655f
 - translated wavelet function, 642, 643f
 - two-dimensional DWT, 738–741, 739f
 - two-level wavelet transform and compression, 741–742, 743f
 - MATLAB program, 744
 - types, 638
 - wavelet coefficients, 646–647
 - wavelet data compression
 - 16-bit ECG data, 668, 669f
 - 16-bit speech data, 667, 667f
 - MATLAB program, 667
 - Wiener filter theory
 - autocorrelation and cross-correlation, 459
 - LMS algorithm, 461–462
 - mean square error quadratic function, 457–458, 458f
 - noise cancellation, 457, 457f
 - statistical expectation, 457–458, 461–462
 - steepest descent algorithm, 459, 460f, 461–462
 - Windowed modified discrete cosine transform (W-MDCT), 522–523, 522f, 545
 - inverse function, 545
 - waveform coding, 524–525, 525f, 546
 - Window method
 - Blackman window, 230, 231f
 - cutoff frequency, 242
 - design procedure, 233
 - Gibbs oscillations, 230
 - Hamming window, 230, 231f
 - Hanning window, 230, 231f
 - Kaiser window, 230
 - length estimation, 241, 241t
 - magnitude frequency response, 240
 - MATLAB function, 237–240, 237t, 288
 - passband ripple, 241–242, 241f
 - rectangular window, 230
 - stopband attenuation, 241–242, 241f
 - triangular (Bartlett) window, 230, 231f
- ## Y
- YCbCr color space, 753
 - YIQ, 690, 690f
 - YUV color model, 752
- ## Z
- Zero-crossing algorithm, 366–367, 368f
 - Zigzag scan, 737
 - Z-plane pole-zero plot, 172f
 - analog-to-digital conversion, 174
 - bounded-in/bounded-out stability, 175
 - features, 172
 - Laplace shift property, 174
 - Laplace vs. z-transform, 173, 174f
 - s-plane vs. z-plane mapping, 175, 175f
 - stability rules, 175, 176f
 - Z-transform
 - definition, 137–138
 - difference equations, 152–156
 - exponential sequence, 138
 - inverse z-transform
 - definition, 144
 - partial fraction expansion method, *see* Partial fraction expansion method
 - lookup table, 156
 - one-sided/unilateral transform, 137–138
 - properties of, 144t
 - causal sequence, 141–143
 - linear convolution, 142–143
 - linearity, 140–141
 - time-shifted sequence, 141
 - region of convergence, 138
 - sequences for, 138–140, 139t