

Index

μ -Law PCM 32–3
3.7 kb/s ACELPC coder 319
4 kb/s ACELP coder 332
6 kb/s ACELP coder 332–3
8 kb/s ACELP coder 333–4
4.15 kb/s IMBE coder 270–1
2.4 kb/s MELP coder 113
4 kb/s MELP/CELP coder 283, 285
1.2 kb/s SB-LPC coder 128
4 kb/s SB-LPC coder 128, 271–5, 319
4.8 kb/s STC coder 268–70
1-tap pitch filter 81–2
3-tap pitch filter 82

A

AaS, *see* analysis-and-synthesis

AbS-LPC coder

block diagram 201

CELP 212, 213, 214, 219–60

excitation signal 200, 206–8, 208–12

generally 199–200

MPLPC 215–17, 218

perceptually-based error

minimization procedure 200,
203–6

procedure 201–2

RPELPC 217–19

SELP 212–15

time-varying filter 200, 202–3

weighting filter 204–5

AC, *see* autocorrelation

ACELP mode 298

ACELP mode error 348, 350

ACELP transition quantization 332–4

acoustic noise, performance of hybrid
coder 336, 337–45

acoustic noise, robustness of SWPM
342

active noise suppression 2

adaptive codebook 52, 53

adaptive differential pulse code

modulation (ADPCM) 1, 2, 8, 12, 13

adaptive multi-rate (AMR) coder 13, 14

adaptive multi-rate (ETSI) coding

standard 360, 362–3, 364–8, 374–5

adaptive normalized least mean squared
(ANLMS) algorithm 416–18

adaptive post-filtering 257–8

adaptive quantizer 33–6, 36–9

adaptive sequential LMS method
100–1

adaptive transversal filter in echo

cancellation 413–14

ADPCM, *see* adaptive differential pulse
code modulation

A-Law PCM 32–3

algebraic codebook excitation 247–51

algorithm

adaptive normalized least mean
squared 416–18

Burg 74

Durbin 69, 72

frequency-domain pitch

determination 155–8, 158–66, 177

- algorithm (*continued*)
 K-means 43–4
 normalized least mean squared 416
 pitch measurement 81
 time-domain pitch determination
 151–5, 158–66, 177
 voice activity detector 11, 281, 311,
 341, 357–75
- aliasing distortion 25
- all-pole digital filter 90
- all-pole modelling 269
- all-pole synthesis filter 205
- Al-Naimi 131, 416
- AMBE speech coding standard 15, 16
- AMDF, *see* average magnitude
 difference function
- AMR, *see* adaptive multi-rate
- analogue speech signal bandwidth 25
- analogue telephony system 1
- analysis-and-synthesis (AaS) coder 199
- analysis-by-synthesis coder, *see*
 AbS-LPC coder
- ANLMS, *see* adaptive normalized least
 mean squared
- anti-alias filter 130–46
- APC 78
- Application Specific Integrated Circuit
 (ASIC) chip 11
- Atal 78, 200, 204, 206
- Atal, Singhal and 235
- Atkinson 299
- autocorrelation PDA 152–5
- autocorrelation method 68–70
- autocorrelation pitch measurement
 algorithm 81
- average magnitude difference function
 (AMDF) PDA 81, 151–2
- average spectral distortion 107
- B**
- background noise 10
- backward adaptation 34, 35, 37
- Bartlett window function 58–9, 60, 61
- Bayes 373
- Berouti 383
- binary search codebook 46–8
- binary voicing, *see* hard decision voicing
- bit rate conflict with speech quality 9
- bit rate reduction xiii
- bit rate, VAD and 359
- Blackman window function 59, 60, 61
- block quantization, *see* vector
 quantization
- Burg's algorithm 74
- burst error 336
- C**
- capacity increase, VAD and 358, 359
- cascaded codebook 48–9, 53
- Cattermole 32
- CCITT regulatory body 9
- CCSR, *see* Centre for Communication
 Systems Research
- CDMA, *see* code division multiple access
- CELP 8, 12
- CELP coder
 codebook excitation 240–54
 excitation behaviour 212, 213, 214
 excitation signal 208–12
 generally 206, 207, 279
 LPC prediction 221–2
 multi-pulse excitation 230–2, 233
 operation 219–21
 pitch prediction 222–8
 post-filtering 257–60
 SNR 228–30
see also 4 kb/s MELP/CELP
- Centre for Communication Systems
 Research (CCSR), University of
 Surrey xiii
- centre-clipped codebook 241–2, 243
- centre-clipping PDA 169–72
- Cepstrum pitch measurement algorithm
 81
- channel dependent mode decision 9
- channel error 10, 54, 130, 336, 345–50
- Chebyshev Series method 100
- Chen 258
- Cho 371
- Cholesky decomposition 71, 81, 234
- closed-loop mode selection of voicing
 311–12, 315–18

- closed-loop optimization 199, 200
 - closed-loop prediction 235–6
 - CNG, *see* comfort noise generator
 - CNI, *see* comfort noise insertion
 - co-channel interference, VAD and 357
 - code division multiple access (CDMA) 14, 358
 - code-excited linear prediction, *see* CELP
 - codebook
 - adaptive 52, 53
 - binary search 46–8
 - cascaded 48–9, 53
 - centre-clipped 241–2, 243
 - comparison of codebooks 117–21
 - design 40, 43–4
 - design of simultaneous joint 116
 - full search 44–6
 - gain-shape 50–2
 - generally 40–2
 - multi-stage vector quantization 110, 113–16
 - optimization 43–6
 - overlapping 241, 242, 243
 - robustness 53–4
 - split vector quantization 49–50, 110, 111–12
 - testing 53
 - training 52, 116, 246
 - training database size 117
 - see also* vector quantization
 - codebook excitation
 - algebraic 247–51
 - CELP coder 240–54
 - Gaussian 241–3
 - generally 206–7, 222
 - LTP and 255–7
 - pitch adaptive mixed 251–4
 - vector sum 243–7
 - codebook vector 40
 - coder
 - AbS-LPC 199–200, 200–2
 - analysis-and-synthesis 199–200
 - CELP 206, 207, 279
 - combined low bit-rate 282–3
 - enhanced variable rate 286
 - harmonic 277–9
 - hybrid 8–9, 280–5
 - hybrid encoder 298–311
 - improved multi-band excitation 268, 270–1
 - 4.15 kb/s INMARSAT-M 270–1
 - low bit-rate 2
 - MBE 189
 - MELP 189
 - 4 kb/s MELP/CELP 283, 285
 - MPLPC 230–7
 - multi-band excitation 264–5, 277
 - parametric 6, 7–8
 - prototype waveform interpolation 282
 - 4.8 kb/s SB-LPC 128, 271–5
 - selection of 15–16
 - sinusoidal transform 261–75, 268–70
 - speech-specific 2
 - split-band LPC 261, 268, 271–5
 - 4.8 kb/s STC 268–70
 - variable rate 9
 - waveform approximating 6
 - see also* CELP coder, hybrid coder
 - coding harmonic speech 261–75
 - coding delay 10
 - comb filter 156
 - combined low bit-rate coder 282–3
 - comfort noise generator (CNG) 357
 - comfort noise insertion (CNI) 357
 - companded quantizer 32–3
 - complex root method 95
 - compression
 - μ -Law 32
 - A-Law 32–3
 - compression of signal 1–2
 - covariance method 70–1
 - Cox 90
- D**
- DFT, *see* discrete Fourier transform
 - DFT–LSF method 99
 - differential quantizer (DQ) 36–9, 122–4
 - differential vector quantization 52
 - digital signal processor (DSP) chip xiii, 11

- digital speech interpolation (DSI) 11
- digitally-encoded speech, advantages of 1
- direct expansion method 101–2
- direct similarity measure 153
- discrete cosine transform (DCT) 380
- discrete Fourier transform (DFT) 380
- distortion measure 42–3, 106
- DoD speech coding standard 14–15
- DQ, *see* differential quantizer
- DQ predictor 122–4
- DSI, *see* digital speech interpolation
- DSP chip, *see* Digital Signal Processor chip
- Durbin's algorithm 69, 72

- E**
- echo cancellation
 - adaptive transversal filter 413–14
 - digital echo canceller 411–13
 - duplex connection 411
 - G.165 (ITU-T) 413, 417–18
 - generally 406–11
 - near-end speech detection 413, 415
 - performance 415–23
 - residual error suppression 413, 415
 - transversal filter 412
- echo canceller with noise suppressor 418–9
- EFR, *see* ETSI GSM enhanced full rate
- EFR coder 251
- EFR weighting method 109, 110, 120, 121
- encoder, hybrid 298–311
- enhanced variable rate coder (EVRC) 286
- enhancing speech 379–426
- environmental variability in signal 53–4
- Ephraim and Malah 369, 381–387
- European Telecommunications Standards Institute (ETSI)
 - GSM enhanced full rate (EFR) speech coding standard 13, 14, 360, 361–2, 364–8
 - GSM full rate (FR) speech coding standard 13, 14, 360, 361–2, 364–8
 - GSM half rate (HR) speech coding standard 13, 14, 360, 361–2, 364–8
 - speech coding standard 10, 13–14
 - UMTS speech coding standard 360
- excitation of white noise 309–11
- excitation signal, determination of optimum 208–12
- excitation signal in AbS-LPC coder 200, 206–8

- F**
- FEC, *see* forward error correction
- filter, finite impulse response 283
- filter, finite length 226–8
- filter memory 203
- finite impulse response (FIR) filter 283
- finite length filter 226–8
- forward adaptation 33, 34, 37
- forward error correction (FEC) 10
- fractional-delay LTP 225–6
- frame energy of speech 185–6
- frame-to-frame interpolation 90
- frequency-domain analysis 57–8
- frequency-domain pitch determination 155–8, 158–66, 177–8
- frequency-domain voicing 263
- frequency response of LPC filter 77
- frequency, sampling 25
- FS-1015 speech coding standard 14, 15
- FS-1016 speech coding standard 14, 15
- full search codebook 44–6

- G**
- G.165 (ITU-T) speech coding standard 413, 417–18
- G.711 (ITU-T) speech coding standard 12, 13
- G.721 (ITU-T) speech coding standard 12, 13
- G.722 (ITU-T) speech coding standard 13
- G.722.1 (ITU-T) speech coding standard 13

- G.723.1 (ITU-T) speech coding standard
 - 12, 13
 - G.723.1 Annex A (ITU-T) speech coding standard 360, 361, 364–8
 - G.723.1 coder 251
 - G.726 (ITU-T) speech coding standard
 - 12, 13
 - G.728 (ITU-T) speech coding standard
 - 12, 13
 - G.729 (ITU-T) speech coding standard
 - 12, 13
 - G.729 Annex B (ITU-T) speech coding standard 360, 361, 364–8, 374–5
 - G.729 coder 251
 - gain in SWPM 323
 - gain-shape codebook 50–2
 - Gaussian codebook excitation 241–3
 - generalized cubic phase interpolation 297–8
 - Gibson 380
 - Gram–Schmidt orthogonalization
 - process 247
 - Griffin 299
 - group delay weighting method
 - 109–10, 121
 - GSM, *see* ETSI speech coding standard
 - GSM enhanced full rate (EFR) ETSI speech coding standard 13, 14, 360, 361–2, 364–8
 - GSM full rate (FR) ETSI speech coding standard 13, 14, 360, 361–2, 364–8
 - GSM half rate (HR) ETSI speech coding standard 13, 14, 360, 361–2, 364–8
 - GSS, *see* spectral subtraction, generalized
- H**
- Hamming window 59, 60–5, 165, 190, 262
 - Hard-decision noise adaptation 402–3
 - hard-decision voicing 150, 178–89
 - harmonic amplitude
 - estimation 266–8
 - generally 272, 299–301
 - quantization in AbS 327–9
 - in SWPM 323
 - harmonic band 194
 - harmonic coder 8
 - see also* coder
 - harmonic excitation mode 298
 - harmonic excitation quantization
 - amplitude 327–9
 - gain 329–30
 - onset parameter 330–1
 - pitch 324, 325, 326
 - pitch pulse location 325–7
 - pitch pulse shape 327
 - transition detection 324–5
 - harmonic excitation, synchronized 299–301
 - harmonic gain quantization in AbS 329–30
 - harmonic memory initialization at onset 308–9
 - harmonic mode error 346–7, 348
 - harmonic peak detection PDA 156
 - harmonic speech coding 261–75
 - harmonic voicing 264–6
 - hierarchical clustering, *see* binary search
 - Hilbert envelope 286
 - human speech emulation 7
 - hybrid coder
 - ACELP transition excitation quantization 331
 - burst error 336
 - decoder 319–20
 - design 280–1
 - encoder 298–311
 - generally 6–7, 8–9
 - harmonic excitation quantization 323–31
 - limitations 284–5
 - LPC vocoder 281
 - performance 320–2
 - performance under acoustic noise 336, 337–45
 - performance under channel errors 336, 345–50
 - quantization issues 322–31
 - random channel error 336
 - speech classification 311–19
 - speech quality 334–5
 - transition detection 315–18

hybrid coder (*continued*)
 unvoiced excitation quantization
 323, 324
 voicing 281
 hybrid mode selection of voicing
 311–12

I

improved multi-band excitation (IMBE)
 coder 268, 270–1
 improved multi-band excitation (IMBE)
 speech coding standard 15, 16
 impulse train generator 65
 INMARSAT speech coding standard
 15, 16
 INMARSAT-M coder 270–1
 inter-frame correlation 121–30
 interpolation
 generalized cubic phase 297–8
 generally 281
 linear 262
 filter 227
 technique overlap and add 262
 inverse filtering (LPC) 76, 77
 inverse sine function 88, 90, 92
 IS-54 (TIA/EIA) speech coding standard
 14, 15
 IS-96 (TIA/EIA) speech coding standard
 360, 363–4, 364–8
 IS-127 (TIA/EIA) speech coding
 standard 360, 363–4, 364–8
 IS-641-A (TIA/EIA) speech coding
 standard 14, 15
 IS-733 (TIA/EIA) speech coding
 standard 360, 363–4, 364–8
 Itakura 90
 Itakura–Saito distortion 389
 iterative sequential optimization
 79–80, 116
 ITU regulatory body
 G.165 speech coding standard 413,
 417–18
 G.711 speech coding standard 12, 13
 G.721 speech coding standard 12, 13
 G.722 speech coding standard 13
 G.722.1 speech coding standard 13

G.723.1 Annex A speech coding
 standard 360, 361, 364–8
 G.723.1 coder 251
 G.723.1 speech coding standard 12,
 13
 G.726 speech coding standard 12, 13
 G.728 speech coding standard 12, 13
 G.729 speech coding standard 12, 13
 G.729 Annex B speech coding
 standard 360, 361, 364–8, 374–5
 G.729 coder 251
 generally 9, 12–13
 P.862 measure of quality 18

J

Jayant (one word memory) quantizer
 34–6
 JQ-MSVQ quantizer 128–31

K

Kaiser window function 59, 60, 61
 Kalman filter 380
 Karhunen–Loève transform (KLT) 380
 Katugampala 285
 Kleijn 282
 K-means algorithm 43–4, 45–6
 Kroon 227

L

lag, pitch 78, 81, 83, 175–7
 LAR function, *see* log-area ratio function
 lattice method 72–4
 least mean square 67
 likelihood ratio 368–75
 line spectral frequency (LSF)
 derivation 94–5
 distribution plot 97
 generally 87, 90
 properties 103–5
 line spectral pair (LSP) 87, 90
 linear interpolation 262
 linear prediction filter coefficient 7
 linear prediction vocoder 7
 linear predictive analysis, *see* LPC
 analysis

- log spectral distortion, *see* spectral distortion
 - log-area ratio (LAR) function 88, 90, 91
 - logarithmic scalar quantizer 32–3
 - Log-PCM system 1, 2
 - long term prediction 202, 203
 - long term predictor in CELP 222–8
 - low bit-rate coder 2
 - low-band to full-band energy of speech 184
 - low-pass filtering 134–46
 - LP filter coefficient 7
 - LPC analysis
 - autocorrelation method 68–70
 - covariance method 70–1
 - generally 65–77, 90
 - lattice method 72–4
 - least mean square approach 67
 - maximum likelihood approach 67
 - in other fields 67
 - performance 74–7
 - LPC difference equation 66
 - LPC filter 87–90
 - LPC frequency response 77
 - LPC inverse filtering 76, 77
 - LPC mode 298
 - LPC predictor 202–3, 221–2
 - LPC quantization process 87, 90, 94–5, 97
 - LPC residual 272
 - LPC spectral envelope 77
 - LPC synthesis 93–4, 102–3, 104
 - LPC transformation to LSF
 - adaptive sequential LMS method 100–1
 - Chebyshev Series method 100
 - complex root method 95
 - generally 90–101
 - ratio filter method 98–100
 - real root method 95–6
 - LSF estimation 130–46
 - LSF inverse distance weighting method 109, 110, 121
 - LSF prediction 122–4
 - LSF quantization process 105–10, 121–30, 128–30
 - LSF quantizer 107, 110–16
 - LSF transformation to LPC 101–2, 102–3, 104
 - LSF, *see* line spectral frequency
 - LSP, *see* line spectral pair
 - LTP analysis 228–30
 - LTP and codebook excitation 255–7
- ## M
- MA-MSVQ quantizer 129–31
 - Markov 380
 - Max quantizer 30–2
 - maximally smooth criterion 297
 - maximum likelihood 67
 - maximum likelihood pitch measurement algorithm 81
 - maximum likelihood STSA estimation 380, 384, 389–92, 398
 - MBE mixed voicing 190–3
 - MBE-based coder 189
 - M-best tree search 115–16
 - McAulay 261, 262, 265, 297, 299, 384
 - mean opinion score (MOS) scale 17
 - mean square error distortion measure 42
 - mean square error measurement 106, 107–10
 - MELP coder 189
 - see also* 4 kb/s MELP/CELP coder
 - memory, filter 203
 - meta-frame 128
 - minimum mean square error STSA (MMSE-STSA) estimation 380, 386–7, 389–92, 400, 401
 - MIRS, *see* modified intermediate response system
 - mixed decision noise adaptation 403–4
 - mode decision 9
 - mode error 345–50
 - modified intermediate response system (MIRS) 164
 - moving average (MA) predictor
 - generally 123–4
 - joint quantization and 129–31
 - low-pass filtering and 135–46
 - optimal order 124–5
 - performance 126–8

- moving average (MA) predictor
 (*continued*)
 prediction factor 125–6
 training 125–6
- MPLPC coder
 amplitude, optimum excitation
 232–5
 excitation behaviour 215–17, 218
 generally 230–2, 233
 pitch prediction 235–7
 pulse location 216, 217
- MSVQ, *see* multi-stage vector
 quantization
- MSVQ quantizer 125–6
- multi-band excitation (MBE) coder
 149, 264–5, 277
- multi-band voicing 264
- Multimedia Communications Research
 Group of CCSR xiii
- multimode coder, *see* hybrid coder
- multi-pulse excitation 207–8, 230–2,
 233
- multipulse LPC (MPLPC) 200, 207
- multi-stage vector quantization
 codebook training 116
 comparison with SVQ 117–21
 generally 111
 M-best tree search 115–16
 2.4 kb/s MELP coder 113
 performance 117–21
 search strategy 114–16
- N**
- narrowband speech coding standard
 12–13
- NATO speech coding standard 15
- near-end speech detection in echo
 cancellation 413, 415
- network dependent mode decision 9
- neural network 283
- Nguyen 153
- noise adaptation
 hard decision 402–3
 mixed decision 403–4
 performance comparison of methods
 404–6, 407, 408, 409, 410
- soft decision 402, 403
- voice activity detector 402
- noise suppression 2
- noise suppression rules 369
- noise suppressor with echo canceller
 418–9
- noisy speech 189
- non-uniform scalar quantizer 29–30
- normalized least mean squared (NLMS)
 algorithm 416
- Nyquist criterion xiii, 24, 25, 131
- O**
- offset target modification in SWPM
 304–8
- one word memory adaptation 34–6
- one-shot optimization 79, 80–1
- onset harmonic memory initialization
 308–9
- onset harmonic parameter quantization
 in AbS 330–1
- open-loop mode selection of voicing
 311–14
- open-loop search scheme 51
- optimization closed-loop 199, 200
- optimization of codebook 43–6, 116
- optimization of pitch predictor 79–81
- optimum scalar quantizer 29–32
- ordering of LSF parameters 100, 103–5
- outlier 107
see also performance
- overlapping codebook 241, 242, 243
- P**
- P.862 measure of quality 18
- packet loss, VAD and 359
- Paliwal–Atal weighting method 108,
 110, 121
- PAME, *see* pitch adaptive mixed
 excitation
- pan-European digital mobile radio
 system, *see* GSM
- Panter and Dite 30
- parametric coder 6, 7–8
- partial correlation (PARCOR) coefficient
 73, 87–90, 93–4

- pattern-matching quantization, *see*
 - vector quantization
- PCM, *see* pulse code modulation
- PDA, *see* pitch determination algorithm
- peak detector 177
- peakiness of speech 179–80
- peak-picking of the magnitude spectrum 266–7
- perceptual evaluation of speech quality 18
- perceptually-based error minimization
 - procedure in AbS-LPC coder 200, 203–6
- perceptually-determined distortion measure 42–3
- performance of LTP analysis methods 228–30
- performance of echo canceller 415–23
- performance of hybrid coder 320–2
- performance of JQ-MSVQ quantizer 129
- performance of low-pass filtering 142–6
- performance of LPC analysis 74–7
- performance of MA-MSVQ quantizer 129, 130
- performance of moving average predictor 126–8
- performance of multi-stage vector quantization 117–21, 125–6
- performance of noise adaptation methods 404–6, 407, 408, 409, 410
- performance of pitch determination algorithms 164–6, 167, 168
- performance of pitch tracking process 175
- performance of speech coding standards 15–18
- performance of speech enhancement methods 389–402
- performance of voice activity detector (VAD) 364–8
- periodicity in speech signal 77–8, 178–9
- phase synchronization 281
- pitch adaptive mixed excitation (PAME) 251–4
- pitch determination 149, 150–78, 263
- pitch determination algorithm (PDA)
 - autocorrelation 152–5
 - average magnitude difference (AMDF) 151–2
 - centre-clipping 169–72
 - generally 149
 - harmonic peak detection 156
 - peak detector 177
 - performance comparison 164–6, 167, 168
 - spectral autocorrelation 158–62, 163–4
 - spectral synthesis 163–4
 - spectrum similarity 156–8
- pitch determination preprocessing 166–77
- pitch error 165, 177–8
- pitch estimation, *see* pitch determination
- pitch filter 81–2
- pitch gain, optimum 153, 154
- pitch lag 78, 81, 83, 175–7
- pitch measurement algorithm 81
- pitch period
 - generally 149, 150–1, 165
 - LP filter coefficient 7
 - SWPM 323
- pitch prediction 235–7
 - see also* long term prediction
- pitch predictor 77–83
 - see also* long term predictor
- pitch pulse location in AbS 325–7
- pitch pulse location in SWPM 286–91, 302–4, 323
- pitch pulse shape in AbS 327
- pitch pulse shape in SWPM 292–7, 302–4, 323
- pitch quantization in AbS 324, 325, 326
- pitch smoothing 172–7
- pitch tracking 172–7
- pitch-LPC formulation model 79
- plosive detection 318–19
- polyphase structure 227
- post-filtering in a CELP coder 257–60

power spectrum 99
 power-saving, VAD and 358
 PPL, *see* pitch pulse location
 PPS, *see* pitch pulse shape
 prediction gain 78, 80, 124–5, 140–1
 prediction of LPC parameters in CELP
 221–2
 prediction of pitch in CELP 222–8
 predictive vector quantization 52
 predictor codebook 52
 pre-emphasis of the signal 75
 pre-emphasized energy of speech 183
 probability density 33
 prototype waveform interpolation (PWI)
 coder 282
 public switched telephone network
 (PSTN) 5, 9, 10
 pulse amplitude coding 237–8
 pulse amplitude quantization, joint
 238–40
 pulse code modulation (PCM) xiii, 5,
 32–3
 pulse excitation 202
 pulse location 211–12, 216, 217, 248
 pulse position coding 237

Q

quality measurements 16
 quantization 23, 238–40
see also types of quantization:
 differential vector, LSF, multi-stage
 vector, predictive vector, scalar,
 split vector, vector
 quantization error 27–8, 29, 106
 quantization issues of hybrid coder
 322–31
 quantization noise 7
 quantization process
 scalar 26–39, 106
 vector 39–50, 106
see also LPC quantization process, LSF
 quantization process
 quantizer
 adaptive 33–9
 companded 32–3
 differential 36–9

Jayant 34–6
 JQ-MSVQ 128–31
 logarithmic scalar 32–3
 LPC 87, 90, 94–5, 97
 LSF 107, 110–16
 Max 30–2
see also scalar quantizer
 quantizer input/output 31
 quantizer step size 26

R

Rabiner 169, 172
 random channel error 336
 random noise excitation 202
 random noise generator 65
 ratio filter method 95–6, 98–100
 real-time coder 108
 real-time system 74
 rectangular window function 58–9,
 60–5, 165
 Reeves 5
 reference template 40
 regular pulse excitation 207–8
 regulatory body 9
 residual error suppression in echo
 cancellation 413, 415
 rms energy 317, 318, 333
 robustness 10
 RPELPC coder 217–19

S

sampling 23–5
 satellite telephony 15
 SB-LPC, *see* split-band LPC
 scalar quantization process 26–39, 106
 scalar quantizer 54
 scalar quantizer, logarithmic 32–3
 scalar quantizer, non-uniform 29–30
 scalar quantizer, optimum 29–32
 scalar quantizer, uniform 26–9
 secure communication 14–15
 segmental SNR 389
 self-excitation 207
 SELP 207
 SELP coder 212–15
 sequential optimization 116

- Shlomot 282
- short term predictor, *see* LPC analysis
- short-time spectral analysis 57–65
- signal compression 1–2
- signal power LP filter coefficient 7
- signal processing 1–2
- signal reconstruction 5
- signal to noise ratio (SNR)
 - CELP coder 228–30
 - generally 7
 - RPELPC coder 218–19
 - segmental 389
- signal variability 53–4
- simultaneous joint codebook design 116
- Singhal and Atal 235
- sinusoidal analysis 262–3
- sinusoidal coder 8, 261–75
- sinusoidal model voicing 265–6
- sinusoidal speech coder 149, 150, 156
- sinusoidal speech-model matching 299
- sinusoidal transform coder (STC) 149, 261–75
- smoothed likelihood ratio (SLR) 371–2, 373, 374–5
- SNR, *see* signal to noise ratio
- soft-decision noise adaptation 402, 403
- soft-decision voicing 150, 189–96
- Sohn 368
- Sondhi 169, 172
- source dependent mode decision 9
- source-filter model 65–7
- speaker variability in signal 53
- spectral analysis, short-time 57–65
- spectral autocorrelation PDA 158–62, 163–4
- spectral correlation 267–8
- spectral distortion 106–7, 131–4
- spectral envelope 77, 149, 202
- spectral subtraction 380, 382–4, 389–92, 396, 397
- spectral synthesis method 150
- spectral synthesis PDA 163–4
- spectral tilt of speech 182, 266
- spectrum flattening 166–72
- spectrum similarity PDA 156–8
- speech characteristic
 - frame energy 185–6
 - low-band to full-band energy 184
 - peakiness 179–80
 - periodic similarity 178–9
 - pre-emphasized energy 183
 - spectrum tilt 182
 - weighting 186–7
 - zero crossing rate 180–1
- speech classification in a hybrid coder 311–19
- speech coder, *see* coder
- speech coding standard
 - DoD 14–15
 - ETSI 13–14
 - INMARSAT 15, 16
 - ITU-T 12–13
 - NATO 15
 - performance 15–18
 - TIA/EIA 14, 15
- speech enhancement
 - adaptive filtering 380
 - discrete cosine transform (DCT) 380
 - discrete Fourier transform (DFT) 380
 - echo cancellation 406–23, 424–6
 - generally 379–80
 - guidelines 402
 - Kalman filter 380
 - Karhunen–Loève transform (KLT) 380
 - maximum likelihood STSA
 - estimation 380, 389–92, 398
 - minimum mean square error STSA
 - estimation 380, 389–92, 400, 401
 - model-based 380
 - noise adaptation 402–6
 - performance comparison of methods 389–402
 - short-time spectral amplitude 381–402
 - spectral subtraction 389–92, 396, 397
 - transform domain 380
 - uncertainty of speech presence 387–9, 401

- speech enhancement (*continued*)
 wavelet transform 380
 Wiener filtering 389–92, 399
- speech presence, uncertainty of 387–9, 401
- speech quality xiii, 9, 10, 334–5
- speech signal
 average spectral distortion 107, 121
 LPC analysis 65–77
 outlier 107, 121
 periodicity 77–8
 requirements for good quality 107
 spectral analysis 57–65
 transition region 57
 unvoiced 57, 58, 65
 voiced 57, 58, 65
- speech stationarity assumption 131
- split-band LPC (SB-LPC) coder 128, 261, 268, 271–5, 342
- split vector codebook 49–50
- split vector quantization 111–12, 117–21
- split-band mixed voicing 193–6
- Stachurski 283
- STANAG speech coding standard 15
- statistical multiplexing, VAD and 359
- STC, *see* sinusoidal transform coder
- STP, *see* short term predictor
- STSA, *see* short-time spectral analysis
- STSA estimation 380, 389–92, 398, 400, 401
- Sundberg 174
- SWPM, *see* synchronized waveform-matched phase model
- synchronized harmonic excitation 299–301
- synchronized waveform-matched phase model (SWPM)
 advantages 301–4
 generally 285–98
 offset target modification 304–8
 robustness to acoustic noise 342
- T**
- tandem connection 11
- telephony system, analogue 1
- threshold function, voicing 191–2, 196
- TIA regulatory body
 enhanced variable rate coder (EVRC) 286
 IS-54 speech coding standard 14, 15
 IS-96 speech coding standard 360, 363–4, 364–8
 IS-127 speech coding standard 360, 363–4, 364–8
 IS-641-A speech coding standard 14, 15
 IS-733 speech coding standard 360, 363–4, 364–8
- time division multiple access (TDMA) 14
- time-domain pitch determination 151–5, 158–66, 177
- time-varying codebook 52
- time-varying filter in AbS-LPC coder 200, 202–3
- Toeplitz matrix 69, 249
- training a codebook 52, 116, 246
- training a moving average predictor 125–6
- Trancoso 282
- transcoding 11
- transition 298
- transition detection 315–18
- transition quantization in ACELP 331, 332–4
- transition region 150
- transmission channel errors 54
- tree search, M-best 115–16
- tree search codebook, *see* binary search codebook
- U**
- UMTS (ETSI) speech coding standard 360
- uncertainty of speech presence 387–9, 401
- uniform scalar quantizer 26–9, 33
- unvoiced excitation 202
- unvoiced speech signal 57, 58, 65, 150, 298
- up-sampling 226–8

V

- VAD, *see* voice activity detector
- variable bit-rate coding 331–5
- variable rate coder 9
- vector quantization
 - harmonic amplitude 272
 - multi-stage 111, 113–21
 - split 111–12
 - see also* codebook
- vector quantization process 39–50, 106
- vector quantizer 54
- vector sum codebook excitation 243–7
- Villette 299
- vocoder 6–7, 149, 150, 202
- voice activity detector (VAD)
 - benefits 357–9
 - ETSI speech coding standards 360, 361–2, 362–3, 364–8, 374–5
 - hard decision noise adaptation 402
 - ITU-T speech coding standards 360, 361, 364–8, 374–5
 - likelihood ratio 368–75
 - performance 364–8
 - TIA/EIA speech coding standards 360, 363–4, 364–8
 - voicing decision 359
- voice activity detector (VAD) algorithm 11, 281, 311, 341, 357
- voiceband data handling 11–12
- voiced excitation 202
- voiced speech signal 57, 58, 65, 150, 298
- voiced–unvoiced classification 149
- voicing
 - frequency-domain 263
 - generally 281
 - harmonic 264–6
 - multi-band excitation 264

- sinusoidal model 265–6
- threshold function 191–2, 196

W

- waveform coder 6–7, 8
- waveform, equation for sampled 23
- wavelet transform 380
- weighted mean square error
 - measurement 106
- weighted mean square error distortion measure 42
- weighting filter of AbS-LPC coder 204–5
- weighting method
 - EFR 109, 110, 120, 121
 - group delay 109–10, 121
 - LSF inverse distance 109–10, 121
 - Paliwal–Atal 108, 110, 121
 - performance 119–21
- white noise excitation 309–11
- white noise excitation mode error 346, 347
- wideband speech coding standard 13
- wide-sense stationary assumption 133
- Wiener filtering 380, 385–6, 389–92, 399
- window length 81
- window function
 - Bartlett 58–9, 60, 61
 - Blackman 59, 60, 61
 - generally 58–65, 75
 - Hamming 59, 60–5, 165, 190, 262
 - Kaiser 59, 60, 61
 - rectangular 58–9, 60–5, 165
- window position test 132

Z

- zero-crossing rate of speech 180–1, 313