

Contents

List of Figures	xiii
List of Tables	xxi
Contributing Authors	xxiii
Introduction	xxix
<i>Karlheinz Brandenburg and Mark Kahrs</i>	
1	
Audio quality determination based on perceptual measurement techniques	1
<i>John G. Beerends</i>	
1.1 Introduction	1
1.2 Basic measuring philosophy	2
1.3 Subjective versus objective perceptual testing	6
1.4 Psychoacoustic fundamentals of calculating the internal sound representation	8
1.5 Computation of the internal sound representation	13
1.6 The perceptual audio quality measure (PAQM)	17
1.7 Validation of the PAQM on speech and music codec databases	20
1.8 Cognitive effects in judging audio quality	22
1.9 ITU Standardization	29
1.9.1 ITU-T, speech quality	30
1.9.2 ITU-R, audio quality	35
1. 10 Conclusions	37
2	
Perceptual Coding of High Quality Digital Audio	39
<i>Karlheinz Brandenburg</i>	
2.1 Introduction	39

2.2	Some Facts about Psychoacoustics	42
2.2.1	Masking in the Frequency Domain	42
2.2.2	Masking in the Time Domain	44
2.2.3	Variability between listeners	45
2.3	Basic ideas of perceptual coding	47
2.3.1	Basic block diagram	48
2.3.2	Additional coding tools	49
2.3.3	Perceptual Entropy	50
2.4	Description of coding tools	50
2.4.1	Filter banks	50
2.4.2	Perceptual models	59
2.4.3	Quantization and coding	63
2.4.4	Joint stereo coding	68
2.4.5	Prediction	72
2.4.6	Multi-channel: to matrix or not to matrix	73
2.5	Applying the basic techniques: real coding systems	74
2.5.1	Pointers to early systems (no detailed description)	74
2.5.2	MPEG Audio	75
2.5.3	MPEG-2 Advanced Audio Coding (MPEG-2 AAC)	79
2.5.4	MPEG-4 Audio	81
2.6	Current Research Topics	82
2.7	Conclusions	83
3		
Reverberation Algorithms		85
<i>William G. Gardner</i>		
3.1	Introduction	85
3.1.1	Reverberation as a linear filter	86
3.1.2	Approaches to reverberation algorithms	87
3.2	Physical and Perceptual Background	88
3.2.1	Measurement of reverberation	89
3.2.2	Early reverberation	90
3.2.3	Perceptual effects of early echoes	93
3.2.4	Reverberation time	94
3.2.5	Modal description of reverberation	95
3.2.6	Statistical model for reverberation	97
3.2.7	Subjective and objective measures of late reverberation	98
3.2.8	Summary of framework	100
3.3	Modeling Early Reverberation	100
3.4	Comb and Allpass Reverberators	105
3.4.1	Schroeder's reverberator	105
3.4.2	The parallel comb filter	108
3.4.3	Modal density and echo density	109
3.4.4	Producing uncorrelated outputs	111
3.4.5	Moorer's reverberator	112
3.4.6	Allpass reverberators	113
3.5	Feedback Delay Networks	116

3.5.1	Jot's reverberator	119
3.5.2	Unitary feedback loops	121
3.5.3	Absorptive delays	122
3.5.4	Waveguide reverberators	123
3.5.5	Lossless prototype structures	125
3.5.6	Implementation of absorptive and correction filters	128
3.5.7	Multirate algorithms	128
3.5.8	Time-varying algorithms	129
3.6	Conclusions	130
4		
Digital Audio Restoration		133
<i>Simon Godsill, Peter Rayner and Olivier Cappé</i>		
4.1	Introduction	134
4.2	Modelling of audio signals	135
4.3	Click Removal	137
4.3.1	Modelling of clicks	137
4.3.2	Detection	141
4.3.3	Replacement of corrupted samples	144
4.3.4	Statistical methods for the treatment of clicks	152
4.4	Correlated Noise Pulse Removal	155
4.5	Background noise reduction	163
4.5.1	Background noise reduction by short-time spectral attenuation	164
4.5.2	Discussion	177
4.6	Pitch variation defects	177
4.6.1	Frequency domain estimation	179
4.7	Reduction of Non-linear Amplitude Distortion	182
4.7.1	Distortion Modelling	183
4.7.2	Non-linear Signal Models	184
4.7.3	Application of Non-linear models to Distortion Reduction	186
4.7.4	Parameter Estimation	188
4.7.5	Examples	190
4.7.6	Discussion	190
4.8	Other areas	192
4.9	Conclusion and Future Trends	193
5		
Digital Audio System Architecture		195
<i>Mark Kahrs</i>		
5.1	Introduction	195
5.2	Input/Output	196
5.2.1	Analog/Digital Conversion	196
5.2.2	Sampling clocks	202
5.3	Processing	203
5.3.1	Requirements	204
5.3.2	Processing	207
5.3.3	Synthesis	208

5.3.4	Processors	209
5.4	Conclusion	234
6	Signal Processing for Hearing Aids	235
	<i>James M. Kates</i>	
6.1	Introduction	236
6.2	Hearing and Hearing Loss	237
6.2.1	Outer and Middle Ear	238
6.3	Inner Ear	239
6.3.1	Retrocochlear and Central Losses	247
6.3.2	Summary	248
6.4	Linear Amplification	248
6.4.1	System Description	249
6.4.2	Dynamic Range	251
6.4.3	Distortion	252
6.4.4	Bandwidth	253
6.5	Feedback Cancellation	253
6.6	Compression Amplification	255
6.6.1	Single-Channel Compression	256
6.6.2	Two-Channel Compression	260
6.6.3	Multi-Channel Compression	261
6.7	Single-Microphone Noise Suppression	263
6.7.1	Adaptive Analog Filters	263
6.7.2	Spectral Subtraction	264
6.7.3	Spectral Enhancement	266
6.8	Multi-Microphone Noise Suppression	267
6.8.1	Directional Microphone Elements	267
6.8.2	Two-Microphone Adaptive Noise Cancellation	268
6.8.3	Arrays with Time-Invariant Weights	269
6.8.4	Two-Microphone Adaptive Arrays	269
6.8.5	Multi-Microphone Adaptive Arrays	271
6.8.6	Performance Comparison in a Real Room	273
6.9	Cochlear Implants	275
6.10	Conclusions	276
7	Time and Pitch scale modification of audio signals	279
	<i>Jean Laroche</i>	
7.1	Introduction	279
7.2	Notations and definitions	282
7.2.1	An underlying sinusoidal model for signals	282
7.2.2	A definition of time-scale and pitch-scale modification	282
7.3	Frequency-domain techniques	285
7.3.1	Methods based on the short-time Fourier transform	285
7.3.2	Methods based on a signal model	293
7.4	Time-domain techniques	293

7.4.1	Principle	293
7.4.2	Pitch independent methods	294
7.4.3	Periodicity-driven methods	298
7.5	Formant modification	302
7.5.1	Time-domain techniques	302
7.5.2	Frequency-domain techniques	302
7.6	Discussion	303
7.6.1	Generic problems associated with time or pitch scaling	303
7.6.2	Time-domain vs frequency-domain techniques	308
8		
	Wavetable Sampling Synthesis	311
	<i>Dana C. Massie</i>	
8.1	Background and introduction	311
8.1.1	Transition to Digital	312
8.1.2	Flourishing of Digital Synthesis Methods	313
8.1.3	Metrics: The Sampling - Synthesis Continuum	314
8.1.4	Sampling vs. Synthesis	315
8.2	Wavetable Sampling Synthesis	318
8.2.1	Playback of digitized musical instrument events.	318
8.2.2	Entire note - not single period	318
8.2.3	Pitch Shifting Technologies	319
8.2.4	Looping of sustain	331
8.2.5	Multi-sampling	337
8.2.6	Enveloping	338
8.2.7	Filtering	338
8.2.8	Amplitude variations as a function of velocity	339
8.2.9	Mixing or summation of channels	339
8.2.10	Multiplexed wavetables	340
8.3	Conclusion	341
9		
	Audio Signal Processing Based on Sinusoidal Analysis/Synthesis	343
	<i>T.F. Quatieri and R. J. McAulay</i>	
9.1	Introduction	344
9.2	Filter Bank Analysis/Synthesis	346
9.2.1	Additive Synthesis	346
9.2.2	Phase Vocoder	347
9.2.3	Motivation for a Sine-Wave Analysis/Synthesis	350
9.3	Sinusoidal-Based Analysis/Synthesis	351
9.3.1	Model	352
9.3.2	Estimation of Model Parameters	352
9.3.3	Frame-to-Frame Peak Matching	355
9.3.4	Synthesis	355
9.3.5	Experimental Results	358
9.3.6	Applications of the Baseline System	362
9.3.7	Time-Frequency Resolution	364
9.4	Source/Filter Phase Model	366

9.4.1	Model	367
9.4.2	Phase Coherence in Signal Modification	368
9.4.3	Revisiting the Filter Bank-Based Approach	381
9.5	Additive Deterministic/Stochastic Model	384
9.5.1	Model	385
9.5.2	Analysis/Synthesis	387
9.5.3	Applications	390
9.6	Signal Separation Using a Two-Voice Model	392
9.6.1	Formulation of the Separation Problem	392
9.6.2	Analysis and Separation	396
9.6.3	The Ambiguity Problem	399
9.6.4	Pitch and Voicing Estimation	402
9.7	FM Synthesis	403
9.7.1	Principles	404
9.7.2	Representation of Musical Sound	407
9.7.3	Parameter Estimation	409
9.7.4	Extensions	411
9.8	Conclusions	411
10		
	Principles of Digital Waveguide Models of Musical Instruments	417
	<i>Julius O. Smith III</i>	
10.1	Introduction	418
10.1.1	Antecedents in Speech Modeling	418
10.1.2	Physical Models in Music Synthesis	420
10.1.3	Summary	422
10.2	The Ideal Vibrating String	423
10.2.1	The Finite Difference Approximation	424
10.2.2	Traveling-Wave Solution	426
10.3	Sampling the Traveling Waves	426
10.3.1	Relation to Finite Difference Recursion	430
10.4	Alternative Wave Variables	431
10.4.1	Spatial Derivatives	431
10.4.2	Force Waves	432
10.4.3	Power Waves	434
10.4.4	Energy Density Waves	435
10.4.5	Root-Power Waves	436
10.5	Scattering at an Impedance Discontinuity	436
10.5.1	The Kelly-Lochbaum and One-Multiply Scattering Junctions	439
10.5.2	Normalized Scattering Junctions	441
10.5.3	Junction Passivity	443
10.6	Scattering at a Loaded Junction of N Waveguides	446
10.7	The Lossy One-Dimensional Wave Equation	448
10.7.1	Loss Consolidation	450
10.7.2	Frequency-Dependent Losses	451
10.8	The Dispersive One-Dimensional Wave Equation	451
10.9	Single-Reed Instruments	455

	Contents	xi
10.9.1	Clarinet Overview	457
10.9.2	Single-Reed Theory	458
10.10	Bowed Strings	462
10.10.1	Violin Overview	463
10.10.2	The Bow-String Scattering Junction	464
10.11	Conclusions	466
	References	467
	Index	535