

Table of Contents

Preface	xiii
Chapter 1. Speech Analysis	1
Christophe D’ALESSANDRO	
1.1. Introduction.	1
1.1.1. Source-filter model	1
1.1.2. Speech sounds.	2
1.1.3. Sources	6
1.1.4. Vocal tract	12
1.1.5. Lip-radiation.	18
1.2. Linear prediction.	18
1.2.1. Source-filter model and linear prediction	18
1.2.2. Autocorrelation method: algorithm	21
1.2.3. Lattice filter	28
1.2.4. Models of the excitation	31
1.3. Short-term Fourier transform	35
1.3.1. Spectrogram	35
1.3.2. Interpretation in terms of filter bank.	36
1.3.3. Block-wise interpretation	37
1.3.4. Modification and reconstruction	38
1.4. A few other representations	39
1.4.1. Bilinear time-frequency representations	39
1.4.2. Wavelets	41
1.4.3. Cepstrum	43
1.4.4. Sinusoidal and harmonic representations	46
1.5. Conclusion	49
1.6. References	50

Chapter 2. Principles of Speech Coding	55
Gang FENG and Laurent GIRIN	
2.1. Introduction.	55
2.1.1. Main characteristics of a speech coder	57
2.1.2. Key components of a speech coder	59
2.2. Telephone-bandwidth speech coders	63
2.2.1. From predictive coding to CELP.	65
2.2.2. Improved CELP coders	69
2.2.3. Other coders for telephone speech	77
2.3. Wideband speech coding.	79
2.3.1. Transform coding.	81
2.3.2. Predictive transform coding.	85
2.4. Audiovisual speech coding.	86
2.4.1. A transmission channel for audiovisual speech	86
2.4.2. Joint coding of audio and video parameters	88
2.4.3. Prospects	93
2.5. References	93
Chapter 3. Speech Synthesis.	99
Olivier BOËFFARD and Christophe D’ALESSANDRO	
3.1. Introduction.	99
3.2. Key goal: speaking for communicating	100
3.2.1. What acoustic content?	101
3.2.2. What melody?	102
3.2.3. Beyond the strict minimum	103
3.3 Synoptic presentation of the elementary modules in speech synthesis systems	104
3.3.1. Linguistic processing.	105
3.3.2. Acoustic processing	105
3.3.3. Training models automatically	106
3.3.4. Operational constraints	107
3.4. Description of linguistic processing	107
3.4.1. Text pre-processing.	107
3.4.2. Grapheme-to-phoneme conversion	108
3.4.3. Syntactic-prosodic analysis	110
3.4.4. Prosodic analysis	112
3.5. Acoustic processing methodology	114
3.5.1. Rule-based synthesis	114
3.5.2. Unit-based concatenative synthesis	115
3.6. Speech signal modeling.	117
3.6.1. The source-filter assumption	118
3.6.2. Articulatory model	119
3.6.3. Formant-based modeling	119

3.6.4. Auto-regressive modeling	120
3.6.5. Harmonic plus noise model	120
3.7. Control of prosodic parameters: the PSOLA technique	122
3.7.1. Methodology background	124
3.7.2. The ancestors of the method	125
3.7.3. Descendants of the method	128
3.7.4. Evaluation	131
3.8. Towards variable-size acoustic units	131
3.8.1. Constitution of the acoustic database	134
3.8.2. Selection of sequences of units	138
3.9. Applications and standardization	142
3.10. Evaluation of speech synthesis.	144
3.10.1. Introduction	144
3.10.2. Global evaluation	146
3.10.3. Analytical evaluation	151
3.10.4. Summary for speech synthesis evaluation.	153
3.11. Conclusions	154
3.12. References.	154
Chapter 4. Facial Animation for Visual Speech	169
Thierry GUIARD-MARIGNY	
4.1. Introduction.	169
4.2. Applications of facial animation for visual speech.	170
4.2.1. Animation movies	170
4.2.2. Telecommunications	170
4.2.3. Human-machine interfaces	170
4.2.4. A tool for speech research.	171
4.3. Speech as a bimodal process.	171
4.3.1. The intelligibility of visible speech	172
4.3.2. Visemes for facial animation	174
4.3.3. Synchronization issues.	175
4.3.4. Source consistency	176
4.3.5. Key constraints for the synthesis of visual speech.	177
4.4. Synthesis of visual speech	178
4.4.1. The structure of an artificial talking head.	178
4.4.2. Generating expressions	178
4.5. Animation.	180
4.5.1. Analysis of the image of a face.	180
4.5.2. The puppeteer	181
4.5.3. Automatic analysis of the speech signal	181
4.5.4. From the text to the phonetic string	181
4.6. Conclusion	182
4.7. References	182

Chapter 5. Computational Auditory Scene Analysis	189
Alain DE CHEVEIGNÉ	
5.1. Introduction.	189
5.2. Principles of auditory scene analysis	191
5.2.1. Fusion versus segregation: choosing a representation	191
5.2.2. Features for simultaneous fusion.	191
5.2.3. Features for sequential fusion.	192
5.2.4. Schemes	193
5.2.5. Illusion of continuity, phonemic restoration	193
5.3. CASA principles.	193
5.3.1. Design of a representation.	193
5.4. Critique of the CASA approach.	200
5.4.1. Limitations of ASA.	201
5.4.2. The conceptual limits of “separable representation”	202
5.4.3. Neither a model, nor a method?	203
5.5. Perspectives	203
5.5.1. Missing feature theory	203
5.5.2. The cancellation principle.	204
5.5.3. Multimodal integration	205
5.5.4. Auditory scene synthesis: transparency measure	205
5.6. References	206
 Chapter 6. Principles of Speech Recognition	 213
Renato DE MORI and Brigitte BIGI	
6.1. Problem definition and approaches to the solution.	213
6.2. Hidden Markov models for acoustic modeling	216
6.2.1. Definition.	216
6.2.2. Observation probability and model parameters	217
6.2.3. HMM as probabilistic automata	218
6.2.4. Forward and backward coefficients	219
6.3. Observation probabilities.	222
6.4. Composition of speech unit models	223
6.5. The Viterbi algorithm.	226
6.6. Language models	228
6.6.1. Perplexity as an evaluation measure for language models	230
6.6.2. Probability estimation in the language model	232
6.6.3. Maximum likelihood estimation	234
6.6.4. Bayesian estimation	235
6.7. Conclusion	236
6.8. References	237

Chapter 7. Speech Recognition Systems	239
Jean-Luc GAUVAIN and Lori LAMEL	
7.1. Introduction.	239
7.2. Linguistic model.	241
7.3. Lexical representation.	244
7.4. Acoustic modeling.	247
7.4.1. Feature extraction.	247
7.4.2. Acoustic-phonetic models.	249
7.4.3. Adaptation techniques	253
7.5. Decoder	256
7.6. Applicative aspects	257
7.6.1. Efficiency: speed and memory	257
7.6.2. Portability: languages and applications	259
7.6.3. Confidence measures.	260
7.6.4. Beyond words	261
7.7. Systems	261
7.7.1. Text dictation	262
7.7.2. Audio document indexing.	263
7.7.3. Dialog systems	265
7.8. Perspectives	268
7.9. References	270
 Chapter 8. Language Identification	 279
Martine ADDA-DECKER	
8.1. Introduction.	279
8.2. Language characteristics	281
8.3. Language identification by humans.	286
8.4. Language identification by machines.	287
8.4.1. LId tasks	288
8.4.2. Performance measures	288
8.4.3. Evaluation	289
8.5. LId resources	290
8.6. LId formulation	295
8.7. LId modeling	298
8.7.1. Acoustic front-end	299
8.7.2. Acoustic language-specific modeling	300
8.7.3. Parallel phone recognition.	302
8.7.4. Phonotactic modeling	304
8.7.5. Back-end optimization.	309
8.8. Discussion	309
8.9. References	311

Chapter 9. Automatic Speaker Recognition	321
Frédéric BIMBOT.	
9.1. Introduction.	321
9.1.1. Voice variability and characterization.	321
9.1.2. Speaker recognition	323
9.2. Typology and operation of speaker recognition systems	324
9.2.1. Speaker recognition tasks	324
9.2.2. Operation.	325
9.2.3. Text-dependence	326
9.2.4. Types of errors	327
9.2.5. Influencing factors	328
9.3. Fundamentals.	329
9.3.1. General structure of speaker recognition systems	329
9.3.2. Acoustic analysis	330
9.3.3. Probabilistic modeling	331
9.3.4. Identification and verification scores	335
9.3.5. Score compensation and decision	337
9.3.6. From theory to practice	342
9.4. Performance evaluation.	343
9.4.1. Error rate	343
9.4.2. DET curve and EER	344
9.4.3. Cost function, weighted error rate and HTER	346
9.4.4. Distribution of errors	346
9.4.5. Orders of magnitude	347
9.5. Applications	348
9.5.1. Physical access control.	348
9.5.2. Securing remote transactions	349
9.5.3. Audio information indexing.	350
9.5.4. Education and entertainment	350
9.5.5. Forensic applications.	351
9.5.6. Perspectives	352
9.6. Conclusions.	352
9.7. Further reading.	353
Chapter 10. Robust Recognition Methods	355
Jean-Paul HATON	
10.1. Introduction	355
10.2. Signal pre-processing methods.	357
10.2.1. Spectral subtraction.	357
10.2.2. Adaptive noise cancellation	358
10.2.3. Space transformation	359
10.2.4. Channel equalization	359
10.2.5. Stochastic models	360
10.3. Robust parameters and distance measures	360

10.3.1. Spectral representations	361
10.3.2. Auditory models.	364
10.3.3 Distance measure	365
10.4. Adaptation methods	366
10.4.1 Model composition	366
10.4.2. Statistical adaptation	367
10.5. Compensation of the Lombard effect.	368
10.6. Missing data scheme.	369
10.7. Conclusion	369
10.8. References.	370
 Chapter 11. Multimodal Speech: Two or Three senses are Better than One	377
Jean-Luc SCHWARTZ, Pierre ESCUDIER and Pascal TEISSIER	
11.1. Introduction	377
11.2. Speech is a multimodal process	379
11.2.1. Seeing without hearing.	379
11.2.2. Seeing for hearing better in noise.	380
11.2.3. Seeing for better hearing... even in the absence of noise.	382
11.2.4. Bimodal integration imposes itself to perception	383
11.2.5. Lip reading as taking part to the ontogenesis of speech.	385
11.2.6. ...and to its phylogenesis ?	386
11.3. Architectures for audio-visual fusion in speech perception	388
11.3.1. Three paths for sensory interactions in cognitive psychology	389
11.3.2. Three paths for sensor fusion in information processing	390
11.3.3. The four basic architectures for audiovisual fusion	391
11.3.4. Three questions for a taxonomy	392
11.3.5. Control of the fusion process	394
11.4. Audio-visual speech recognition systems	396
11.4.1. Architectural alternatives	397
11.4.2. Taking into account contextual information	401
11.4.3. Pre-processing	403
11.5. Conclusions	405
11.6. References.	406
 Chapter 12. Speech and Human-Computer Communication	417
Wolfgang MINKER & Françoise NÉEL	
12.1. Introduction	417
12.2. Context.	418
12.2.1. The development of micro-electronics.	419
12.2.2. The expansion of information and communication technologies and increasing interconnection of computer systems	420

12.2.3. The coordination of research efforts and the improvement of automatic speech processing systems	421
12.3. Specificities of speech.	424
12.3.1. Advantages of speech as a communication mode	424
12.3.2. Limitations of speech as a communication mode	425
12.3.3. Multidimensional analysis of commercial speech recognition products	427
12.4. Application domains with voice-only interaction.	430
12.4.1. Inspection, control and data acquisition	431
12.4.2. Home automation: electronic home assistant	432
12.4.3. Office automation: dictation and speech-to-text systems	432
12.4.4. Training.	435
12.4.5. Automatic translation	438
12.5. Application domains with multimodal interaction	439
12.5.1. Interactive terminals	440
12.5.2. Computer-aided graphic design.	441
12.5.3. On-board applications	442
12.5.4. Human-human communication facilitation	444
12.5.5. Automatic indexing of audio-visual documents	446
12.6. Conclusions	446
12.7. References.	447
Chapter 13. Voice Services in the Telecom Sector	455
Laurent COURTOIS, Patrick BRISARD and Christian GAGNOULET	
13.1. Introduction	455
13.2. Automatic speech processing and telecommunications	456
13.3. Speech coding in the telecommunication sector	456
13.4. Voice command in telecom services	457
13.4.1. Advantages and limitations of voice command	457
13.4.2. Major trends	459
13.4.3. Major voice command services	460
13.4.4. Call center automation (operator assistance)	460
13.4.5. Personal voice phonebook	462
13.4.6. Voice personal telephone assistants	463
13.4.7. Other services based on voice command	463
13.5. Speaker verification in telecom services	464
13.6. Text-to-speech synthesis in telecommunication systems	464
13.7. Conclusions	465
13.8. References.	466
List of Authors	467
Index	471