1. Record Nr. UNINA9910830012803321

Titolo	Spoken language processing [[electronic resource] /] / edited by Joseph Mariani
Pubbl/distr/stampa	London, : ISTE Hoboken, NJ, : John Wiley and Sons, 2009
ISBN	1-282-25394-8 9786613814593 0-470-61118-9 0-470-39381-5
Descrizione fisica	1 online resource (505 p.)
Collana	ISTE ; ; v.34
Altri autori (Persone)	MarianiJoseph
Disciplina	006.4/54 006.454
Soggetti	Automatic speech recognition Speech processing systems
Lingua di pubblicazione	Inglese
Formato	Materiale a stampa
Livello bibliografico	Monografia
Note generali	Description based upon print version of record.
Nota di bibliografia	Includes bibliographical references and index.
Nota di contenuto	Spoken Language Processing; Table of Contents; Preface; Chapter 1. Speech Analysis; 1.1. Introduction; 1.1.1. Source-filter model; 1.1.2. Speech sounds; 1.1.3. Sources; 1.1.4. Vocal tract; 1.1.5. Lip-radiation; 1.2. Linear prediction; 1.2.1. Source-filter model and linear prediction; 1.2.2. Autocorrelation method: algorithm; 1.2.3. Lattice filter; 1.2.4. Models of the excitation; 1.3. Short-term Fourier transform; 1.3.1. Spectrogram; 1.3.2. Interpretation in terms of filter bank; 1.3.3. Block-wise interpretation; 1.3.4. Modification and reconstruction; 1.4. A few other representations 1.4.1. Bilinear time-frequency representations1.4.2. Wavelets; 1.4.3. Cepstrum; 1.4.4. Sinusoidal and harmonic representations; 1.5. Conclusion; 1.6. References; Chapter 2. Principles of Speech Coding; 2.1. Introduction; 2.1.1. Main characteristics of a speech coder; 2.1.2. Key components of a speech coder; 2.2. Telephone-bandwidth speech coders; 2.2.1. From predictive coding to CELP; 2.2.2. Improved CELP coders; 2.2.3. Other coders for telephone speech; 2.3. Wideband speech coding; 2.3.1. Transform coding; 2.3.2. Predictive transform

coding; 2.4. Audiovisual speech coding
2.4.1. A transmission channel for audiovisual speech2.4.2. Joint coding of audio and video parameters; 2.4.3. Prospects; 2.5. References; Chapter 3. Speech Synthesis; 3.1. Introduction; 3.2. Key goal: speaking for communicating; 3.2.1. What acoustic content?; 3.2.2. What melody?; 3.2.3. Beyond the strict minimum; 3.3 Synoptic presentation of the elementary modules in speech synthesis systems; 3.3.1. Linguistic processing; 3.3.2. Acoustic processing; 3.3.3. Training models automatically; 3.3.4. Operational constraints; 3.4. Description of linguistic processing; 3.4.1. Text pre-processing
3.4.2. Grapheme-to-phoneme conversion3.4.3. Syntactic-prosodic analysis; 3.4.4. Prosodic analysis; 3.5. Acoustic processing methodology; 3.5.1. Rule-based synthesis; 3.5.2. Unit-based concatenative synthesis; 3.6. Speech signal modeling; 3.6.1. The source-filter assumption; 3.6.2. Articulatory model; 3.6.3. Formant-based modeling; 3.6.4. Auto-regressive modeling; 3.6.5. Harmonic plus noise model; 3.7. Control of prosodic parameters: the PSOLA technique; 3.7.1. Methodology background; 3.7.2. The ancestors of the method; 3.7.3. Descendants of the method; 3.7.4. Evaluation
3.8. Towards variable-size acoustic units3.8.1. Constitution of the acoustic database; 3.8.2. Selection of sequences of units; 3.9. Applications and standardization; 3.10. Evaluation of speech synthesis; 3.10.1. Introduction; 3.10.2. Global evaluation; 3.10.3. Analytical evaluation; 3.10.4. Summary for speech synthesis evaluation; 3.11. Conclusions; 3.12. References; Chapter 4. Facial Animation for Visual Speech; 4.1. Introduction; 4.2. Applications of facial animation for visual speech; 4.2.1. Animation movies; 4.2.2. Telecommunications; 4.2.3. Human-machine interfaces
4.2.4. A tool for speech research

Sommario/riassunto

Speech processing addresses various scientific and technological areas. It includes speech analysis and variable rate coding, in order to store or transmit speech. It also covers speech synthesis, especially from text, speech recognition, including speaker and language identification, and spoken language understanding.This book covers the following topics: how to realize speech production and perception systems, how to synthesize and understand speech using state-of-the-art methods in signal processing, pattern recognition, stochastic modelling computational linguistics and human factor st