Automatic speech recognition (ASR) systems are finding increasing use in everyday life. Many of the commonplace environments where the systems are used are noisy, for example users calling up a voice search system from a busy cafeteria or a street. This can result in degraded speech recordings and adversely affect the performance of speech recognition systems. As the use of ASR systems increases, knowledge of the state-of-the-art in techniques to deal with such problems becomes critical to system and application engineers and researchers who work with or on ASR technologies. This book presents a comprehensive survey of the state-of-the-art in techniques used to improve the robustness of speech recognition systems to these degrading external influences.

Key features:

- Reviews all the main noise robust ASR approaches, including signal separation, voice activity detection, robust feature extraction, model compensation and adaptation, missing data techniques and recognition of reverberant speech.
- Acts as a timely exposition of the topic in light of more widespread use in the future of ASR technology in challenging environments.
- Addresses robustness issues and signal degradation which are both key requirements for practitioners of ASR.
- Includes contributions from top ASR researchers from leading research units in the field.
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Dr. Virtanen is a senior researcher at Tampere University of Technology. Previously, he has worked at Cambridge University, UK as a research associate. His main research contributions are in sound source separation and its application to robust speech recognition, audio content analysis, and music information retrieval. He is well-known for his work on non-negative matrix factorization based source separation, which is currently widely used in the field. He has published numerous journal and conference articles related to above topics.

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