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Sommario/riassunto

Speech processing and speech transmission technology are expanding fields of active research. New challenges arise from the 'anywhere, anytime' paradigm of mobile communications, the ubiquitous use of voice communication systems in noisy environments and the convergence of communication networks toward Internet based transmission protocols (such as Voice over IP). As a consequence, new speech coding, new enhancement and error concealment, and new quality assessment methods are emerging. Advances in Digital Speech Transmission provides an up-to-date overview of the field, including topics such as speech coding in heterogeneous communication networks, wideband coding, and the quality assessment of modern speech transmission systems. Provides an insight into the latest developments in speech processing and speech transmission, making it an essential reference to those working in these fields Offers a balanced overview of technology and applications Presents a unique in-depth discussion of soft-decision source decoding, joint source-channel coding and Turbo DeCodulation techniques Explains speech signal processing in hearing instruments and human-machine interfaces from applications point of view Covers speech coding for Voice over IP, source localization, digital hearing aids and speech processing for automatic speech recognition Advances in Digital Speech Transmission serves as an essential link between the basics and the type of technology and applications (prospective) engineers work on in industry labs and academia. The book will also be of interest to advanced students, researchers, and other professionals who need to brush up their knowledge in this field.
