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Sommario/riassunto

"The new generation of voice services and telephony will be based on packet networks rather than TDM transmission and switching. This book addresses the evolution of telephony to Voice over IP (VoIP) and Unified Communications (UC), bringing email, voice mail, fax, and telephone services to one user interface. Concise and to the point, this text tells readers what they need to know to deal with vendors, network engineers, data center gurus, and top management with the confidence and clear understanding of how things really work. It serves as a useful tool for engineers just entering the field, as well as for experienced engineers and technical managers who want to deal effectively with sales people"--
