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3 SIP Clients and Servers 3.1 SIP User Agents; 3.2 Presence Agents; 3.3 Back-to-Back User Agents; 3.4 SIP Gateways; 3.5 SIP Servers; 3.5.1 Proxy Servers; 3.5.2 Redirect Servers; 3.5.3 Registrar Servers; 3.6 Uniform Resource Indicators; 3.7 Acknowledgment of Messages; 3.8 Reliability; 3.9 Multicast Support; 3.10 Conclusion; 3.11 Questions; References; 4 SIP Request Messages; 4.1 Methods; 4.1.1 INVITE; 4.1.2 REGISTER; 4.1.3 BYE; 4.1.4 ACK; 4.1.5 CANCEL; 4.1.6 OPTIONS; 4.1.7 SUBSCRIBE; 4.1.8 NOTIFY; 4.1.9 PUBLISH; 4.1.10 REFER; 4.1.11 MESSAGE; 4.1.12 INFO; 4.1.13 PRACK; 4.1.14 UPDATE
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Sommario/riassunto

Now in its third edition, the ground-breaking Artech House bestseller SIP: Understanding the Session Initiation Protocol offers you the most comprehensive and current understanding of this revolutionary protocol for call signaling and IP Telephony. The third edition has been significantly expanded with brand new chapters on NAT traversal, SIP security, services in SIP, presence and instant messaging, Peer-to-Peer SIP, and an introduction to ABNF and XML. This cutting-edge book shows you how SIP provides a highly-scalable and cost-effective way to offer new and exciting telecommunication feature sets, helping you design your "next generation" network and develop new applications and software stacks. Other key discussions include SIP as a key component in the Internet multimedia conferencing architecture, request and response messages, devices in a typical network, types of servers, SIP headers, comparisons with existing signaling protocols including H.323, related protocols SDP (Session Description Protocol) and RTP (Real-time Transport Protocol), and the future direction of SIP. Detailed call flow diagrams illustrate how this technology works with other protocols such as H.323 and ISUP. Moreover, this book covers SIP RFC 3261 and the complete set of SIP extension RFCs.
