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Autore	Johnston Alan B.
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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 4.2 URI and URL Schemes Used by SIP 4.2.1 SIP and SIPS URIs; 4.2.2 Telephone URLs; 4.2.3 Presence and Instant Messaging URLs; 4.3 Tags; 4.4 Message Bodies; 4.5 Conclusion; 4.6 Questions; References; 5 SIP Response Messages; 5.1 Informational; 5.1.1 100 Trying; 5.1.2 180 Ringing; 5.1.3 181 Call is Being Forwarded; 5.1.4 182 Call Queued; 5.1.5 183 Session Progress; 5.2 Success; 5.2.1 200 OK; 5.2.2 202 Accepted; 5.2.3 204 No Notification; 5.3 Redirection; 5.3.1 300 Multiple Choices; 5.3.2 301 Moved Permanently; 5.3.3 302 Moved Temporarily; 5.3.4 305 Use Proxy; 5.3.5 380 Alternative Service 5.4 Client Error 5.4.1 400 Bad Request; 5.4.2 401 Unauthorized; 5.4.3 402 Payment Required; 5.4.4 403 Forbidden; 5.4.5 404 Not Found; 5.4.6 405 Method Not Allowed; 5.4.7 406 Not Acceptable; 5.4.8 407 Proxy Authentication Required; 5.4.9 408 Request Timeout; 5.4.10 409 Conflict; 5.4.11 410 Gone; 5.4.12 411 Length Required; 5.4.13 412 Conditional Request Failed; 5.4.14 413 Request Entity Too Large; 5.4.15 414 Request-URI Too Long; 5.4.16 415 Unsupported Media Type; 5.4.17 416 Unsupported URI Scheme; 5.4.18 417 Unknown Resource Priority; 5.4.19 420 Bad Extension; 5.4.20 421 Extension Required

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.
