Sip Understanding The Session Initiation Protocol Second Edition

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Packet Guide to Voice Over IP Pearson Education

This book constitutes the thoroughly refereed proceedings of the 10th International Workshop on Principles, Systems and Applications of IP Telecommunications, held in Heidelberg, Germany, in July 2008. The 16 full papers presented were carefully reviewed and selected from a total of 56 submissions. Topics covered include recent advances in the domains of convergent networks, VoIP security, and multimedia service environments for next generation networks.

Session Initiation Protocol (SIP): Controlling Convergent Networks

Artech House

State-of-the-art SIP primer SIP (Session Initiation Protocol) is the open standard that will make IP telephony an irresistible force in communications, doing for converged services what http does for the Web. SIP Demystified – authored by Gonzalo Camarillo, one of the contributors to SIP development in the IETF—gives you the tools to keep your company and career competitive. This guide tells you why the standard is needed, what architectures it supports, and how it interacts with other protocols. As a bonus, you even get a context-setting background in data networking. Perfect if you ' re moving from switched voice into a data networking environment, here ’ s everything you need to understand: * Where, why, and how SIP is used * What SIP can do and deliver * SIP ’ s fit with other standards and systems * How to plan implementations of SIP-enabled services * How to size up and choose from available SIP products

And Other Dispatches Morgan Kaufmann

Software-defined networking (SDN) technologies powered by the OpenFlow protocol provide viable options to address the bandwidth needs of next-generation computer networks. And, since many large corporations already produce network devices that support the OpenFlow standard, there are opportunities for those who can manage complex and large-scale networks using these technologies. Network Innovation through OpenFlow and SDN: Principles and Design explains how you can use SDN and OpenFlow to build networks that are easy to design, less expensive to build and operate, and more agile and customizable. Among the first books to systematically address the design aspects in SDN/OpenFlow, it presents the insights of expert contributors from around the world. The book’s four sections break down basic concepts, engineering design, QoS (quality-of-service), and advanced topics. Introduces the basic principles of SDN/OpenFlow and its applications in network systems Illustrates the entire design process of a practical

OpenFlow/SDN Addresses the design issues that can arise when applying OpenFlow to cloud computing platforms Compares various solutions in QoS support Provides an overview of efficient solutions to the integration of SDN with optical networks Identifies the types of network attacks that could occur with OpenFlow and outlines possible solutions for overcoming them Supplying a cutting-edge look at SDN and OpenFlow, this book gives you the wide-ranging understanding required to build, deploy, and manage OpenFlow/SDN products and networks. The book’s comprehensive coverage includes system architectures, language and programming issues, switches, controllers, multimedia support, security, and network operating systems. After reading this book you will understand what it takes to make a smooth transition from conventional networks to SDN/OpenFlow networks.

Asterisk SIPUnderstanding the Session Initiation Protocol

The first book published on deploying Voice Over IP (VoIP) products from Nortel Networks, the largest supplier of voice products in the world. This book begins with a discussion of the current protocols used for transmitting converged data over IP as well as an overview of Nortel ’ s hardware and software solutions for converged networks. In this section, readers will learn how H.323 allows dissimilar communication devices to communicate with each other, and how SIP (Session Initiation Protocol) is used to establish, modify, and terminate multimedia sessions including VOIP telephone calls. This section next introduces the reader to the Multimedia

Concentration Server 5100, and Nortel’s entire suite of Multimedia Communications Portfolio (MCP) products. The remaining chapters of the book teach the reader how to design, install, configure, and troubleshoot the entire Nortel product line. · If you are tasked with designing, installing, configuring, and troubleshooting a converged network built with Nortel's Multimedia Concentration Server 5100, and Multimedia Communications Portfolio (MCP) products, then this is the only book you need. · It shows how you'll be able to design, build, secure, and maintaining a cutting-edge converged network to satisfy all of your business requirements · Also covers how to secure your entire multimedia network from malicious attacks

A Solutions Manual for Network Professionals

BoD – Books on Demand

This book gives a detailed overview of SIP specific security issues and how to solve them While the standards and products for VoIP and SIP services have reached market maturity, security and regulatory aspects of such services are still being discussed. SIP itself specifies only a basic set of security mechanisms that cover a subset of possible security issues. In this book, the authors survey important aspects of securing SIP-based services. This encompasses a description of the problems themselves and the standards-based solutions for such problems. Where a standards-based solution has not been defined, the alternatives are discussed and the benefits and constraints of the different solutions are highlighted. Key Features: Will help the readers to understand the actual problems of using and developing VoIP services, and to distinguish between real problems and the general hype of VoIP security Discusses key aspects of SIP security including authentication, integrity, confidentiality, non-repudiation and signalling Assesses the real security issues facing users of SIP, and details the latest theoretical and practical solutions to SIP Security issues Covers secure SIP access, inter-provider secure communication, media security, security of the IMS infrastructures as well as VoIP services vulnerabilities and countermeasures against Denial-of-Service attacks and VoIP spam This book will be of interest to IT staff involved in deploying and developing VoIP, service users of SIP, network engineers, designers and managers. Advanced undergraduate and graduate students studying data/voice/multimedia communications as well as researchers in academia and industry will also find this book valuable.

Convergence Technologies for 3G Networks John Wiley & Sons

SIPUnderstanding the Session Initiation ProtocolArtech House

Understanding the Session Initiation Protocol Springer Science & Business Media

"This book is like a good tour guide.It doesn't just describe themajor attractions; you share in the history, spirit, language, andculture of the place." --Henning Schulzrinne, Professor, Columbia University Since its birth in 1996, Session Initiation Protocol (SIP) hasgrown up. As a richer, much more robust technology, SIP today isfully capable of supporting the communication systems that powerour twenty-first century work and life. This second edition handbook has been revamped to cover the neweststandards, services, and products. You'll find the latest on SIPusage beyond VoIP, including Presence, instant messaging (IM),mobility, and emergency services, as well as peer-to-peer SIPapplications, quality-of-service, and security issues--everythingyou need to build and deploy today's SIP services. This book will help you * Work with SIP in Presence and event-based communications * Handle SIP-based application-level mobility issues * Develop applications to facilitate communications access forusers with disabilities * Set up Internet-based emergency services * Explore how peer-to-peer SIP systems may change VoIP * Understand the critical importance of Internet transparency * Identify relevant standards and specifications * Handle potential quality-of-service and security problems

IP Multimedia Concepts and Services CRC Press

Now in its fourth 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 fourth edition incorporates changes in SIP from the last five years with new chapters on internet threats and attacks, WebRTC and SIP, and substantial updates throughout. This cutting-edge book shows how SIP provides a highly-scalable and cost-effective way to offer new and exciting telecommunication feature sets, helping practitioners design “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.

Services, Technologies, and Security of Session Initiation Protocol Elsevier

The first complete guide to planning, evaluating, and implementing high-value SIP trunking solutions Most large enterprises have switched to IP telephony, and service provider backbone networks have largely converted to VoIP transport. But there's a key missing link: most businesses still connect to their service providers via old-fashioned, inflexible TDM trunks. Now, three Cisco® experts show how to use Session Initiation Protocol (SIP) trunking to eliminate legacy interconnects and gain the full benefits of end-to-end VoIP. Written for enterprise decision-makers, network architects, consultants, and service providers, this book demystifies SIP trunking technology and trends and brings unprecedented clarity to the transition from TDM to SIP interconnects. The authors separate the true benefits of SIP trunking from the myths and help you systematically evaluate and compare service provider offerings. You will find detailed cost analyses, including guidance on identifying realistic, achievable savings. SIP Trunking also introduces essential techniques for optimizing network design and security, introduces proven best practices for implementation, and shows how to apply them through a start-to-finish case study. Discover the advanced Unified Communications solutions that SIP trunking facilitates Systematically plan and prepare your network for SIP trunking Generate effective RFPs for SIP trunking Ask service providers the right questions—and make sense of their answers Compare SIP deployment models and assess their tradeoffs Address key network design issues, including security, call admission control, and call flows Manage SIP/TDM interworking throughout the transition This IP communications book is part of the Cisco Press® Networking Technology Series. IP communications titles from Cisco Press help networking professionals understand voice and IP telephony technologies, plan and design converged networks, and implement network solutions for increased productivity.

Challenges and Solutions for Voice over WLANs CRC Press

This book constitutes the thoroughly refereed post-proceedings of the Second International Service Availability Symposium, ISAS 2005, held in Berlin, Germany in April 2005. The 15 revised full papers presented together with a keynote talk were carefully selected for inclusion in the book. The papers are organized in topical sections on data and computation availability, specifying, modeling and verifying service availability, high-availability by service-oriented architectures, modeling and composition, and verification and availability assessment.

SIP Demystified Elsevier

Wi-Fi telephony is the latest, most cost effective, and clearest way of carrying voice data wirelessly. The great news is that it can be integrated seamlessly into the same infrastructures as currently used for computer and telephone data.The digital quality is far above current cellular technologies. This book will be among the first to discuss Session Initiation Protocol (SIP), Quality of Service (QoS), and interoperability in connection with Wi-Fi telephony. Security challenges are also presented and solved along these malleable wireless boundaries. In short, this book provides all the information necessary for effective, reliable, crystal clear Wi-Fi telephony service and implementation.

*Using current telephone and computer infrastructure this technology can be implemented at low cost *The importance of Quality of Service (QoS) and security of Wi-Fi telephony is considered *Enhances the clarity of a call beyond a basic cellular phone using digital data transfer

SIP Trunking Artech House

The transportation of multimedia over the network requires timely and errorless transmission much more strictly than other data. This had led to special protocols and to special treatment in multimedia applications (telephony, IP-TV, streaming) to overcome network issues. This book begins with an overview of the vast market combined with the user's expectations. The base mechanisms of the audio/video coding (H.26x etc.) are explained to understand characteristics of the generated network traffic. Further chapters treat common specialized underlying IP network functions which cope with multimedia data in conjunction which special time adaption measures. Based on those standard functions these chapters can treat uniformly SIP, H.248, High-End IP-TV, Webcast, Signage etc. A special section is devoted to home networks which challenge high-end service delivery due to possibly unreliable management. The whole book treats concepts described in accessible IP-based standards and which are implemented broadly. The book is aimed at graduate students/practitioners with good basic knowledge in computer networking. It provides the reader with all concepts of currently used IP technologies of how to deliver multimedia efficiently to the end user.

A Modern Approach Including Java® Practice McGraw Hill Professional

WebRTC, Web Real-Time Communications, is revolutionizing the way web users communicate, both in the consumer and enterprise worlds. WebRTC adds standard APIs (Application Programming Interfaces) and built-in real-time audio and video capabilities and codecs to browsers without a plug-in. With just a few lines of JavaScript, web developers can add high quality peer-to-peer voice, video, and data channel communications to their collaboration, conferencing, telephony, or even gaming site or application. New for the Third Edition The third edition has an enhanced demo application which now shows the use of the data channel for real-time text sent directly between browsers. Also, a full description of the browser media negotiation process including actual SDP session descriptions from Firefox and Chrome. Hints on how to use Wireshark to monitor WebRTC protocols, and example captures are also included. TURN server support for NAT and firewall traversal is also new. This edition also features a step-by-step introduction to WebRTC, with concepts such as local media, signaling, and the Peer Connection introduced through separate runnable demos. Written by experts involved in the standardization effort, this book contains the most up to date discussion of WebRTC standards in W3C and IETF. Packed with figures, example code, and summary tables, this book is the ultimate

Protocol Overview 10.3 References 11 IETF Documents 11.1 Request For Comments 11.2 Internet-Drafts 11.3 RTCWEB Working Group Internet-Drafts 11.4 Individual Internet-Drafts 11.5 RTCWEB Documents in Other Working Groups 11.6 References 12 IETF Related RFC Documents 12.1 Real-time Transport Protocol 12.2 Session Description Protocol 12.3 NAT Traversal RFCs 12.4 Codecs 12.5 Signaling 12.6 References 13 Security and Privacy 13.1 Browser Security Model 13.2 New WebRTC Browser Attacks 13.3 Communication Security 13.4 Identity in WebRTC 13.5 Enterprise Issues 14 Implementations and Uses INDEX ABOUT THE AUTHORS

Understanding the Session Initiation Protocol CRC Press

This newly revised edition of the ground-breaking Artech House bestseller, SIP: Understanding the Session Initiation Protocol gives you a thorough and up-to-date understanding of this revolutionary protocol for call signaling and IP Telephony. The second edition includes brand new discussions on the use of SIP for wireless multimedia communications. It explains how SIP is powerful "rendezvous" protocol that leverages mobility and presence to allow users to communicate using different devices, modes, and services anywhere they are connected to the Internet You learn why SIP has been chosen by the 3GPP (3rd Generation Partnership Program for wireless cell phones) as the core signaling, presence, and instant messaging protocol.

Internet Communications Using SIP "O'Reilly Media, Inc."

Internet Protocol (IP) telephony is an alternative to the traditional Public Switched Telephone Networks (PSTN), and the Session Initiation Protocol (SIP) is quickly becoming a popular signaling protocol for VoIP-based applications. SIP is a peer-to-peer multimedia signaling protocol standardized by the Internet Engineering Task Force (IETF), and it plays a vital role in providing IP telephony services through its use of the SIP Proxy Server (SPS), a software application that provides call routing services by parsing and forwarding all the incoming SIP packets in an IP telephony network. SIP Proxy Server Performance closely examines key aspects to the efficient design and implementation of SIP proxy server architecture. Together, a strong design and optimal implementation can enable significant enhancements to the performance characteristics of SPS. Since SPS performance can be characterized by the transaction states of each SIP session, the book analyzes an existing M/M/1-network performance model for SIP proxy servers in light of key performance benchmarks, such as the average response time for processing the SIP calls and the average number of SIP calls in the system. It also presents several other real-world industrial case studies to aid in further optimizations. This book is intended for researchers, practitioners and professionals interested in optimizing SIP proxy server performance. Professionals working on other VoIP solutions will also find the book valuable.

Building a VoIP Network with Nortel's Multimedia Communication Server 5100

John Wiley & Sons

bull; Demonstrates how real-time audio and video is packetized for transmission. bull; Explains the details of the RTP standards and related concepts. bull; How to implement RTP to work around network problems and limitations

Measuring SIP Proxy Server Performance VON Books

Go under the hood of an operating Voice over IP network, and build your knowledge of the protocols and architectures used by this Internet telephony technology. With this concise guide, you’ll learn about services involved in VoIP and get a first-hand view of network data packets from the time the phones boot through calls and subsequent connection teardown. With packet captures available on the companion website, this book is ideal whether you’re an instructor, student, or professional looking to boost your skill set. Each chapter includes a set of review questions, as well as practical, hands-on lab exercises. Learn the requirements for deploying packetized voice and video Understand traditional telephony concepts, including local loop, tip and ring, and T carriers Explore the Session Initiation Protocol (SIP), VoIP’s primary signaling protocol Learn the operations and fields for VoIP’s standardized RTP and RTCP transport protocols Delve into voice and video codecs for converting analog data to digital format for transmission Get familiar with Communications Systems H.323, SIP’s widely used predecessor Examine the Skinny Client Control Protocol used in Cisco VoIP phones in networks around the world

Network Innovation through OpenFlow and SDN Pearson Education

The International Conference of Computational Methods in Sciences and Engineering

(ICCMSE) is unique in its kind. It regroups original contributions from all fields of the traditional Sciences, Mathematics, Physics, Chemistry, Biology, Medicine and all branches of Engineering. The aim of the conference is to bring together computational scientists from several disciplines in order to share methods and ideas. More than 370 extended abstracts have been submitted for consideration for presentation in ICCMSE 2004. From these, 289 extended abstracts have been selected after international peer review by at least two independent reviewers.

Second International Service Availability Symposium, ISAS 2005, Berlin, Germany, April 25-26, 2005, Revised Selected Papers CRC Press

The merging of voice and data on a single network opens powerful new possibilities in communications. Only a fundamental understanding of both technologies will ensure you are equipped to maximise their full potential. Convergence Technologies for 3G Networks describes the evolution from cellular to a converged network that integrates traditional telecommunications and the technology of the Internet. In particular, the authors address the application of both IP and ATM technologies to a cellular environment, including IP telephony protocols, the use of ATM/AAL2 and the new AAL2 signalling protocol for voice/multimedia and data transport as well as the future of the UMTS network in UMTS Release 5/6 All-IP architecture. Convergence Technologies for 3G Networks: Explains the operation and integration of GSM, GPRS, EDGE, UMTS, CDMA2000, IP, and ATM. Provides practical examples of 3G connection scenarios. Describes signalling flows and protocol stacks. Covers IP and ATM as used in a 3G context. Addresses issues of QoS and real-time application support. Includes IP/SS7 internetworking and IP softswitching. Outlines the architecture of the IP Multimedia Subsystem (IMS) for UMTS. Convergence Technologies for 3G Networks is suited for professionals from the telecommunications, data communications and computer networking industries..

Delivering VoIP and Multimedia Services with Session Initiation Protocol Elsevier Translates technical jargon into practical businesscommunications solutions This book takes readers from traditional voice, fax, video, anddata services delivered via separate platforms to a single, unifiedplatform delivering all of these services seamlessly via theInternet. With its clear, jargon-free explanations, the authorenables all readers to better understand and assess the growingnumber of voice over Internet protocol (VoIP) and unifiedcommunications (UC) products and services that are available forbusinesses. VoIP and Unified Communications is based on the author's carefulreview and synthesis of more than 7,000 pages of publishedstandards as well as a broad range of datasheets, websites, whitepapers, and webinars. It begins with an introduction to IPtechnology and then covers such topics as: Packet transmission and switching VoIP signaling and call processing How VoIP and UC are defining the future Interconnections with global services Network management for VoIP and UC This book features a complete chapter dedicated to cost analysesand payback calculations, enabling readers to accurately determinethe short- and long-term financial impact of migrating to variousVoIP and UC products and services. There's also a chapter detailingmajor IP systems hardware and software. Throughout the book,diagrams illustrate how various VoIP and UC components and systemswork. In addition, the author highlights potential problems andthreats to UC services, steering readers away from commonpitfalls. Concise and to the point, this text enables readers—fromnovices to experienced engineers and technical managers—tounderstand how VoIP and UC really work so that everyone canconfidently deal with network engineers, data center gurus, and topmanagement.