Sip Understanding The Session Initiation Protocol Fourth Edition

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Seven Deadliest Unified Communications Attacks John Wiley & Sons The 3rd edition of this highly successful text builds onthe achievement of the first two editions to provide comprehensiv ecoverage of IMS. It continues to explore the concepts,architecture, protocols and functionalities of IMS while providinga wealth of new and updated information. It is written in a mannerthat allows readers to choose the level of knowledge andunderstanding they need to gain about the IMS. With 35% new material, The IMS,IP Multimedia Concepts andServices, 3rd Edition has been completely revised toinclude updated chapters as well as totally new chapters on IMSmultimedia telephony and IMS voice call continuity. Additional newmaterial includes IMS transit, IMS local numbering, emergencysessions, identification of communication services in IMS, newauthentication model for fixed access, NAT traversal and globallyroutable user agents URI. Detailed descriptions of protocolbehaviour are provided on a level that can be used forimplementation and testing. Key features of the 3rd edition: Two new chapters on IMS multimedia telephony service and IMSVoice Call Continuity Updated information on Third Generation Partnership Project(3GPP) Release 7 level, including architecture, reference pointsand concepts

Substantially extended coverage on IMS detailed procedures Completely rewritten and extended chapters on IMSservices **International Conference of Computational Methods in Sciences and Engineering (ICCMSE 2004)** Elsevier 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 WebRTC reference. Table of Contents 1 Introduction to Web Real-Time Communications 1.1 WebRTC Introduction 1.2 Multiple Media Streams in WebRTC 1.3 Multi-Party Sessions in WebRTC 1.4 WebRTC Standards 1.5 What is New in WebRTC 1.6 Important Terminology Notes 1.7 References 2 How to Use WebRTC 2.1 Setting Up a WebRTC Session 2.2 WebRTC Networking and Interworking Examples 2.3 WebRTC Pseudo-Code Example 2.4 References 3 Local Media 3.1 Media in WebRTC 3.2 Capturing Local Media 3.3 Media Selection and Control 3.4 Media Streams

Wi-Fi Telephony John Wiley & Sons

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. *Second International Service Availability Symposium, ISAS 2005, Berlin, Germany,* *April 25-26, 2005, Revised Selected Papers* Addison-Wesley Professional 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.

A Modern Approach Including Java® Practice 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. Session Initiation Protocol (SIP): Controlling Convergent Networks John Wiley & Sons 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 Next Generation Teletraffic and Wired/Wireless Advanced Networking SIPUnderstanding the Session Initiation Protocol More and more businesses today have their receive phone service through Internet instead of local phone company lines. Many businesses are also using their internal local and wide-area network infrastructure to replace legacy enterprise telephone networks. This migration to a single network carrying voice and data is called convergence, and it's revolutionizing the world of telecommunications by slashing costs and empowering users. The technology of families driving this convergence is called VoIP, or Voice over IP. VoIP has advanced Internet-based telephony to a viable solution, piquing the interest of companies small and large. The primary reason for migrating to VoIP is cost, as it equalizes the costs of long distance calls, local calls, and e-mails to fractions of a penny per use. But the real enterprise turn-on is how VoIP empowersbusinesses to mold and customize telecom and datacom solutions using a single, cohesive networking platform. These business drivers are so compelling that legacy telephony is going the way of the dinosaur, yielding to Voice over IP as the dominant enterprise communications paradigm. Developed from real-world experience by a senior developer, O'Reilly's Switching to VoIP provides solutions for the most common VoIP migration challenges. So if you're a network professional who is migrating from a traditional telephony system to a modern, feature-rich network, this book is a must-have. You'lldiscover the strengths and weaknesses of circuit-switched and packet-switched networks, how VoIP systems impact network infrastructure, as well as solutions for common challenges involved with IP voice migrations. Among the challenges discussed and projects presented: building a softPBX configuring IP phones ensuring quality of service scalability standards-compliance topological considerations coordinating a complete system ?switchover? migrating applications like voicemail and directoryservices retro-interfacing to traditional telephony supporting mobile users security and survivability dealing with the challenges of NAT To help you grasp the core principles at work, Switching to VoIP uses a combination of strategy and hands-on "how-to" that introduce VoIP routers and media gateways, various makes of IP telephone equipment, legacy analog phones, IPTables and Linux firewalls, and the Asterisk open source PBX software by Digium.You'll learn how to build an IP-based or legacy-compatible phone system and voicemail system complete with e-mail integration while becoming familiar with VoIP protocols and devices. Switching to VoIP remains vendor-neutral and advocates standards, not brands. Some of the standards explored include: SIP H.323, SCCP, and IAX Voice codecs 802.3af Type of Service, IP precedence, DiffServ, and RSVP 802.1a/b/g WLAN If VoIP has your attention, like so many others, then Switching to VoIP will help you build your own system, install it, and begin making calls. It's the only thing left between you and a modern telecom network. *Delivering VoIP and Multimedia Services with Session Initiation Protocol* Syngress 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.

Wireless and Mobile All-IP Networks

Pearson Education

With contributions by Michael Ashikhmin, Michael Gleicher, Naty Hoffman, Garrett Johnson, Tamara Munzner, Erik Reinhard, Kelvin Sung, William B. Thompson, Peter Willemsen, Brian Wyvill. The third edition of this widely adopted text gives students a comprehensive, fundamental introduction to computer graphics. The authors present the mathematical foundations of computer graphics with a focus on geometric intuition, allowing the programmer to understand and apply those foundations to the development of efficient code. New in this edition: Four new contributed chapters, written by experts in their fields: Implicit Modeling, Computer Graphics in Games, Color, Visualization, including information visualization Revised and updated material on the graphics pipeline, reflecting a modern viewpoint organized around programmable shading. Expanded treatment of viewing that improves clarity and consistency while unifying viewing in ray tracing and rasterization. Improved and expanded coverage of triangle meshes and mesh data structures. A new organization

for the early chapters, which concentrates foundational material at the beginning to increase teaching flexibility.
SIP Handbook
John Wiley & Sons
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. Human-Computer Interaction McGraw Hill Professional In Securing VoIP Networks, two leading experts systematically review the security risks and vulnerabilities associated with VoIP networks and offer proven, detailed recommendations for securing them. Drawing on case studies from their own fieldwork, the authors address VoIP security from the perspective of real-world network implementers, managers, and security specialists. The authors identify key threats to VoIP networks, including eavesdropping, unauthorized access, denial of service, masquerading, and fraud; and review vulnerabilities in protocol design, network architecture, software, and system configuration that place networks at risk. They discuss the advantages and tradeoffs associated with protection mechanisms built into SIP, SRTP, and other VoIP protocols; and review key management solutions such as MIKEY and ZRTP. Next, they present a complete security framework for enterprise VoIP networks, and provide detailed architectural guidance for both service providers and enterprise users. 1 Introduction 2 VoIP Architectures and Protocols 3 Threats and Attacks 4 VoIP Vulnerabilites 5 Signaling Protection Mechanisms 6 Media Protection Mechanisms 7 Key Management Mechanisms 8 VoIP and Network Security Controls 9 A Security Framework for Enterprise VoIP Networks 10 Provider Architectures and Security 11 Enterprise Architectures and Security IP, UMTS, EGPRS and ATM Packt Publishing Ltd 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.
Service Availability
VON Books
This book constitutes the refereed proceedings of the 8th International Conference on Next Generation Teletraffic and Wired/Wireless Advanced Networking, NEW2AN 2008, held in St. Petersburg, Russia in September 3-5, 2008 in conjunction with the First ruSMART 2008. The 21 revised full papers presented were carefully reviewed and selected from a total of 60 submissions. The NEW2AN papers are organized in topical sections on wireless networks, multi-hop wireless networks, cross-layer design, teletraffic theory, multimedia communications, heterogeneous networks, network security. The ruSMART papers start with three keynote talks followed by seven articles on Smart Spaces.
Practical VoIP Security
Springer Science & Business Media
• The expert author speaks on the topic of SIP at conferences worldwide
Understanding the Session Initiation Protocol
Profile Books Limited

"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 *SIP Beyond VoIP* John Wiley & Sons This book is for programmers who want to learn about real-time communication and utilize the full potential of WebRTC. It is assumed that you have working knowledge of setting up a basic telecom infrastructure as well as basic programming and scripting knowledge. Elsevier Voice Over IP (VoIP) phone lines now represent over 50% of all new phone line installations. Every one of these new VoIP phone lines and handsets must now be protected from malicious hackers because these devices now reside on the network and are accessible from the Internet just like any server or workstation. This book will cover a wide variety of the publicly available exploit tools and how they can be used specifically against VoIP (Voice over IP) Telephony systems. The book will cover the attack methodologies that are used against the SIP and H.323 protocols as well as VoIP network infrastructure. Significant emphasis will be placed on

both attack and defense techniques. This book is designed to be very hands on and scenario intensive · More VoIP phone lines are being installed every day than traditional PBX phone lines · VoIP is vulnerable to the same range of attacks of any network device · VoIP phones can receive as many Spam voice mails as your e-mail can receive Spam e-mails, and as result must have the same types of anti-spam capabilities
SIP John Wiley & Sons
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.
WebRTC Integrator's Guide Morgan Kaufmann
Providing a thorough overview to SIP (session initiation protocol) servlets 1.1, this unique

resource serves as a practical guide to this exciting and emerging communications network technology. Covering all key concepts and their links into Java Enterprise Edition (JEE), the book discusses the construction, deployment and lifecycle of the SIP servlet. You find a detailed presentation of the role, responsibilities, and convergence of the SIP servlet container. Further, the book addresses the application router, addressing its role, container interactions, routing regions, and application sequencing.

Principles, Systems and Applications of IP Telecommunications. Services and Security for Next Generation Networks

CRC Press Provides information on Asterisk, an open source telephony application.