Peer-to-Peer (P2P) networks enable users to directly share digital content (such as audio, video, and text files) as well as real-time data (such as telephony traffic) with other users without depending on a central server. Although originally popularized by unlicensed online music services such as Napster, P2P networking has recently emerged as a viable multimillion dollar business model for the distribution of information, telecommunications, and social networking. Written at an accessible level for any reader familiar with fundamental Internet protocols, the book explains the conceptual operations and architecture underlying basic P2P systems using well-known commercial systems as models and also provides the means to improve upon these models with innovations that will better performance, security, and flexibility. Peer-to-Peer Networking and Applications is thus both a valuable starting point and an important reference to those practitioners employed by any of the 200 companies with approximately $400 million invested in this new and lucrative technology. Uses well-known commercial P2P systems as models, thus demonstrating real-world applicability. Discusses how current research trends in wireless networking, high-def content, DRM, etc. will intersect with P2P, allowing readers to account for future developments in their designs. Provides online access to the Overlay Weaver P2P emulator, an open-source tool that supports a number of peer-to-peer applications with which readers can practice.
Now that virtually every leading telecommunications service provider has committed to delivering IP-based telephony services, communications professionals face the enormous challenge of implementation. This hands-on guide brings together today's best known answers and solutions for delivering VoIP services with the quality customers demand. No other book covers the combined issues of protocol signaling, media transport methodology, reference topological considerations and voice quality testing in service offerings. No matter what your role in delivering Voice Over IP (VoIP) services, IP Telephony delivers the specifics you need to speed deployment, improve reliability, ensure quality, and simplify troubleshooting. Precise, thorough, and based firmly in the real-world, it is simply indispensable. The accompanying CD-ROM contains an extensive library of IP telephony-related RFCs, pertinent white papers and application notes that include real-world IP Telephony measurement examples.

Guide to Voice and Video over IP

How prepared are you to build fast and efficient web applications? This eloquent book provides what every web developer should know about the network, from fundamental limitations that affect performance to major innovations for building even more powerful browser applications—including HTTP 2.0 and XHR improvements, Server-Sent Events (SSE), WebSocket, and WebRTC. Author Ilya Grigorik, a web performance engineer at Google, demonstrates performance optimization best practices for TCP, UDP, and TLS protocols, and explains unique wireless and mobile network optimization requirements. You’ll then dive into performance characteristics of technologies such as HTTP 2.0, client-side network scripting with XHR, real-time streaming with SSE and WebSocket, and P2P communication with WebRTC. Deliver superlative TCP, UDP, and TLS performance Speed up network performance over 3G/4G mobile networks Develop fast and energy-efficient mobile applications Address bottlenecks in HTTP 1.x and other browser protocols Plan for and deliver the best HTTP 2.0 performance Enable efficient real-time streaming in the browser Create efficient peer-to-peer videoconferencing and low-latency applications with real-time WebRTC transports

Voice and Video Conferencing Fundamentals

The IMS is the foundation architecture for the next generation of mobile phones, wireless-enabled PDAs, PCs, and the like. IMS delivers multimedia content (audio, video, text, etc.) over all types of networks. For network engineers/administrators and telecommunications engineers it will be essential to not only understand IMS architecture, but to also be able to apply it at every stage of the network design process. This book will contain pragmatic information on how to engineer IMS networks as well as an applications-oriented approach for the engineering and networking professionals responsible for making IMS function in the real world. * Describes the convergence of wireless IMS (IP Multimedia Subsystem) with other networks, including wireline and cable * Discusses building interfaces for end users and IMS applications servers * Explores network management issues with IMS

Practical VoIP Security

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.

Packet Broadband Network Handbook

Session Initiation Protocol (SIP) was conceived in 1996 as a signaling protocol for inviting users to multimedia conferences. With this development, the next big Internet revolution silently started. That was the revolution which would end up converting the Internet into a total communication system which would allow people to talk to each other, see each other, work collaboratively or send messages in real time. Internet telephony and, in general, Internet multimedia, is the new revolution today and SIP is the key protocol which allows this revolution to happen. The IMS is the foundation architecture for the next generation of mobile phones, wireless-enabled PDAs, PCs, and the like. IMS delivers multimedia content (audio, video, text, etc.) over all types of networks. For network engineers/administrators and telecommunications engineers it will be essential to not only understand IMS architecture, but to also be able to apply it at every stage of the network design process. This book will contain pragmatic information on how to engineer IMS networks as well as an applications-oriented approach for the engineering and networking professionals responsible for making IMS function in the real world.
Session Initiation Protocol (SIP) was conceived in 1996 as a signaling protocol for inviting users to multimedia conferences. With this development, the next big Internet revolution silently started. That was the revolution which would end up converting the Internet into a total communication system which would allow people to talk to each other, see each other, work collaboratively or send messages in real time. Internet telephony and, in general, Internet multimedia, is the new revolution today and SIP is the key protocol which allows this revolution to grow. The book explains, in tutorial fashion, the underlying technologies that enable real-time IP multimedia communication services in the Internet (voice, video, presence, instant messaging, online picture sharing, white-boarding, etc). Focus is on session initiation protocol (SIP) but also covers session description protocol (SDP), Real-time transport protocol (RTP), and message session relay protocol (MSRP). In addition, it will also touch on other application-related protocols and refer to the latest research work in IETF and 3GPP about these topics.

The book includes discussion of leading edge theory (which is key to really understanding the technology) accompanied by Java examples that illustrate the theoretical concepts. Throughout the book, in addition to the code snippets, the reader is guided to build a simple but functional IP soft-phone therefore demonstrating the theory with practical examples.

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Read Free Internet Communications Using Sip Delivering Voip And Multimedia Services With Session Initiation Protocol

This book covers IP multimedia from both a theoretical and practical point of view focusing on letting the reader understand the concepts and put them into practice using Java. It includes lots of drawings, protocol diagrams, UML sequence diagrams and code snippets that allow the reader to rapidly understand the concepts. Focus on HOW multimedia communications over the Internet works to allow readers to really understand and implement the technology. Explains how SIP works, including many programming examples so the reader can understand abstract concepts like SIP dialogs, SIP transactions, etc. It is not focused on just VoIP. It looks at a wide array of enhanced communication services related to SIP enabling the reader to put this technology into practice. Includes nearly 100 references to the latest standards and working group activities in the IETF, bringing the reader completely up to date. Provides a step-by-step tutorial on how to build a basic, though functional, IP soft-phone allowing the reader to put concepts into practice. For advanced readers, the book also explains how to build a SIP proxy and a SIP registrar to enhance one's expertise and marketability in this fast moving area.

SIP: Understanding the Session Initiation Protocol, Fourth Edition

Leading authorities deliver the commandments for designing high-speed networks. There are no end of books touting the virtues of one or another high-speed networking technology, but until now, there were none offering networking professionals a framework for choosing and integrating the best ones for their organization's networking needs. Written by two world-renowned experts in the field of high-speed network design, this book outlines a total strategy for designing high-bandwidth, low-latency systems. Using real-world implementation examples to illustrate their points, the authors cover all aspects of network design, including network components, network architectures, topologies, protocols, application interactions, and more.

SIP Demystified

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.

VoIP and Unified Communications

Deliver rich audio and video real-time communication and peer-to-peer data exchange right in the browser, without the need for proprietary plug-ins. This concise hands-on guide shows you how to use the emerging Web Real-Time Communication (WebRTC) technology to build a browser-to-browser application, piece by piece. The authors' learn-by-example approach is perfect for web programmers looking to understand real-time communication, and telecommunications architects unfamiliar with HTML5 and JavaScript-based client-server web programming. You'll use a ten-step recipe to create a complete WebRTC system, with exercises that you can apply to your own projects. Tour the WebRTC development cycle and trapezoid architectural model. Understand how and why VoIP is shifting from standalone functionality to a browser component. Use mechanisms that let client-side web apps interact with browsers through the WebRTC API. Transfer streaming data between browser peers with the RTCPeerConnection API. Create a signaling channel between peers for setting up a WebRTC session. Put everything together to create a basic WebRTC system from scratch. Learn about conferencing, authorization, and other advanced WebRTC features.

The British National Bibliography
Internet Communications Using SIP Delivering VoIP And Multimedia Services With Session Initiation Protocol

Chat, interactive games, and others to run all at the same time. Now that the deployment of real SIP networks is about to take off, two leaders of the commercial rollout deliver complete guidance on this exciting new technology. Geared to IT and networking professionals and decision-makers at Internet service providers (ISPs), as well as networking (NSPs) and application (ASPs) service providers, this book helps readers sort through the available vendor offerings and services to discover how to integrate and maximize SIP's power across their networks.

SIP Trunking

Why the Internet was designed to be the way it is, and how it could be different, now and in the future. How do you design an internet? The architecture of the current Internet is the product of basic design decisions made early in its history. What would an internet look like if it were designed, today, from the ground up? In this book, MIT computer scientist David Clark explains how the Internet is actually put together, what requirements it was designed to meet, and why different design decisions would create different internets. He does not take today's Internet as a given but tries to learn from it, and from alternative proposals for what an internet might be, in order to draw some general conclusions about network architecture.

Clark discusses the history of the Internet, and how a range of potentially conflicting requirements—including longevity, security, availability, economic viability, management, and meeting the needs of society—shaped its character. He addresses both the technical aspects of the Internet and its broader social and economic contexts. He describes basic design approaches and explains, in terms accessible to nonspecialists, how networks are designed to carry out their functions. (An appendix offers a more technical discussion of network functions for readers who want the details.) He considers a range of alternative proposals for how to design an internet, examines in detail the key requirements a successful design must meet, and then imagines how to design a future internet from scratch. It's not that we should expect anyone to do this; but, perhaps, by conceiving a better future, we can push toward it.

Asterisk

“This book is like a good tour guide. It doesn't just describe the major attractions; you share in the history, spirit, language, and culture of the place.” --Henning Schulzrinne, Professor, Columbia University Since its birth in 1996, Session Initiation Protocol (SIP) has grown up. As a richer, much more robust technology, SIP today is fully capable of supporting the communication systems that power our twenty-first century work and life. This second edition handbook has been revamped to cover the newest standards, services, and products. You'll find the latest on SIP usage beyond VoIP, including Presence, instant messaging (IM), mobility, and emergency services, as well as peer-to-peer SIP applications, quality-of-service, and security issues—everything you 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 for users 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 Security

Scores of factors and considerations affect real-time audio and video transmission. This book addresses all of them: packetization for transmission, the details of the RTP standards, related concepts, and how to implement RTP to work around network problems and limitations.

Real-Time Communication with WebRTC

Voice and Video Conferencing Fundamentals Design, develop, select, deploy, and support advanced IP-based audio and video conferencing systems Scott Firestone, Thiya Ramalingam, Steve Fry As audio and video conferencing move rapidly into the mainstream, customers and end users are demanding unprecedented performance, reliability, scalability, and security. In Voice and Video Conferencing Fundamentals, three leading experts systematically introduce the principles, technologies, and protocols underlying today's state-of-the-art conferencing systems. Discover how to use these concepts and techniques to deliver unified, presence-enabled services that integrate voice, video, telephony, networks, and the Internet—and enable breakthrough business collaboration. The authors begin with a clear, concise overview of current voice and video conferencing, including system components, operational modes, endpoints, features, and user interactivity. Next, they illuminate conferencing architectures, offering practical insights for designing today's complex IP-based conferencing and communications systems.
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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

Designing an Internet

Overlay Networks

This book presents a review of the latest advances in speech and video compression, computer networking protocols, the assessment and monitoring of VoIP quality, and next generation network architectures for multimedia services. The book also concludes with three case studies, each presenting easy-to-follow step-by-step instructions together with challenging hands-on exercises.

- Features:
  - Provides illustrative worked examples and end-of-chapter problems
  - Examines speech and video compression techniques, together with speech and video compression standards
  - Describes the media transport protocols RTP and RTCP, as well as the VoIP signalling protocols SIP and SDP
  - Discusses the concepts of VoIP quality of service and quality of experience
  - Reviews next-generation networks based on the IP multimedia subsystem and mobile VoIP
  - Presents case studies on building a VoIP system based on Asterisk, setting up a mobile VoIP system based on Open IMS and Android mobile, and analysing VoIP protocols and quality.

IP Telephony

Provides information on Asterisk, an open source telephony application.
Within the set of many identifier-locator separation designs for the Internet, HIP has progressed further than anything else we have so far. It is time to see what HIP can do in larger scale in the real world. In order to make that happen, the world needs a HIP book, and now we have it.” - Jari Arkko, Internet Area Director, IETF

One of the challenges facing the current Internet architecture is the incorporation of mobile and multi-homed terminals (hosts), and an overall lack of protection against Denial-of-Service attacks and identity spoofing. The Host Identity Protocol (HIP) is being developed by the Internet Engineering Task Force (IETF) as an integrated solution to these problems. The book presents a well-structured, readable and compact overview of the core protocol with relevant extensions to the Internet architecture and infrastructure. The covered topics include the Bound End-to-End Tunnel Mode for IPsec, Overlay Routable Cryptographic Hash Identifiers, extensions to the Domain Name System, IPv4 and IPv6 interoperability, integration with SIP, and support for legacy applications. Unique features of the book:

- All-in-one source for HIP specifications
- Complete coverage of HIP architecture and protocols
- Base exchange, mobility and multihoming extensions
- Practical snapshots of protocol operation
- IP security on lightweight devices
- Traversal of middleboxes, such as NATs and firewalls
- Name resolution infrastructure
- Micromobility, multicast, privacy extensions
- Chapter on applications, including HIP pilot deployment in a Boeing factory
- HOWTO for HIP on Linux (HIPL) implementation

This book is an important compliment to the official IETF specifications, this book will be a valuable reference for practicing engineers in equipment manufacturing companies and telecom operators, as well as network managers, network engineers, network operators and telecom engineers. Advanced students and academics, IT managers, professionals and operating system specialists will also find this book of interest.

Take Part in the Future of Wireless/Wireline Convergence

The IP multimedia subsystem (IMS), established as the foundation for future wireless and wireline convergence, is the bedrock that will facilitate easy deployment on new, rich, personalized multimedia communication services that mix telecom and data services. Designers, planners, and researchers of communication systems will need to make full use of the technology occurring with this convergence if they want to be the ones providing end users with new and efficient services that are as cost-effective as they are innovative. To provide researchers and technicians with the tools they need to optimize their role in this communication revolution, the IP Multimedia Subsystem (IMS) Handbook presents all the technical aspects of the IMS needed to support the growth of digital traffic and the implementation of underlying networks. This guide covers everything from basic concepts to research-grade material, including the future direction of the architecture.

Learn How IMS Will Speed Innovation

Filling the gap between existing traditional telecommunications and Internet technologies, IMS has led to an environment in which new services and concepts are introduced more quickly than ever before, such as reusable service components and real-time integration. The technology promises to be a cost-effective evolutionary path to future wireless and wireline convergences that will meet next-generation service requirements.

Computer and Communication Networks, Second Edition, explains the modern technologies of networking and communications, preparing you to analyze and simulate complex networks, and to design cost-effective networks for emerging requirements. Offering uniquely balanced coverage of basic and advanced topics, it teaches through case studies, realistic examples and exercises, and intuitive illustrations. Nader F. Mir establishes a solid foundation in basic networking concepts; TCP/IP schemes; wireless and LTE networks; Internet applications, such as Web and e-mail; and network security. Then, he delves into both network analysis and advanced networking protocols, VoIP, cloud-based multimedia networking, SDN, and virtualized networks. In this new edition, Mir provides updated, practical, scenario-based information that many networking books lack, offering a uniquely effective blend of theory and implementation.

Drawing on extensive field experience, he presents many contemporary applications and covers key topics that other texts overlook, including P2P and voice/video networking, SDN, information-centric networking, and modern router/switch design. Students, researchers, and networking professionals will find up-to-date, thorough information here.
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In emergency and disaster scenarios, it is vital to have a stable and effective infrastructure for relaying communication to the public. With the advent of new technologies, more options are available for enhancing communication systems. Multimedia Services and Applications in Mission Critical Communication Systems is a comprehensive source of academic research on the challenges and solutions in creating stable mission critical systems and examines methods to improve system architecture and resources. Highlighting innovative perspectives on topics such as quality of service, performance metrics, and intrusion detection, this book is ideally designed for practitioners, professionals, researchers, graduate students, and academics interested in public safety communication systems.

IP Multimedia Subsystem (IMS) Handbook

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.

High-Speed Networking

This newly revised edition of the groundbreaking bestseller offers a thorough and up-to-date understanding of this revolutionary technology for IP Telephony. Essential reading for anyone involved in the development and operation of voice or data networks, this second edition includes brand-new discussions on the use of SIP as a wireless communications protocol and mobility technology.

Carrier Grade Voice Over IP, Third Edition

Find out how IAX can complement SIP to overcome complications encountered in current SIP-based communications. Written by an expert in the field of telecommunications, this book describes the Inter-Asterisk Exchange protocol (IAX) and its operations, discussing the main characteristics of the protocol including NAT traversal, security, IPv6 support, interworking between IPv4 and IPv6, interworking with SIP and many others. The author presents the ways in which IAX can be activated so as to avoid complications such as NAT and the presence of intermediary boxes in operational architectures. This book analytically demonstrates the added values of IAX protocol compared to existing ones, while proposing viable deployment scenarios that assess the behavior of the protocol in operational networks. Key Features:

- Promotes a viable alternative protocol to ease deployment of multimedia services
- Analyses the capabilities of the IAX protocol and its ability to meet VoIP service provider requirements, and provides scenarios of introducing IAX within operational architectures
- Addresses the advantages and disadvantages of SIP, and
- Details the features of IAX that can help, in junction with SIP, to overcome various disadvantages of SIP
- Explores the added values of IAX protocol compared to existing protocols
- Discusses the compatibility of new adopted architectures and associated protocols

This book will be a valuable reference for service providers,
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Internet Communications Using SIP

Multimedia Networking Technologies, Protocols, and Architectures

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.

Computer and Communication Networks

IP (internet protocol) Telephony, enabled by softswitches, is going to usher in a new era in telecommunications. By putting voice and data over one IP network, operators can enjoy lower costs and create new, revenue-generating "multimedia" services. This valuable reference offers a comprehensive overview of the technology behind IP telephony and offers essential information to network engineers, designers and managers who need to understand the protocols and explore the issues involved in migrating the existing telephony infrastructure to an IP-based real time communication service. Drawing on extensive research and practical development experience in VoIP from its earliest stages, the authors give access to all the relevant standards and cutting-edge techniques in a single resource.

IP Telephony: Deploying Voice-over-IP Protocols: Assumes a working knowledge of IP and networking and addresses the technical aspects of real-time communication over IP. Presents a high level overview of packet media transport technologies, covering all the major VoIP protocols – SIP, H323 and...
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MGCP Details specific strategies to design services for public networks where endpoints cannot be trusted and can be behind firewalls. Explores the problems that may arise from incomplete protocol implementations, or architectures optimized for private networks which fail in a public environment. This amply illustrated, state-of-the-art reference tool will be an invaluable resource for all those involved in the practical deployment of VoIP technology.

Location-Based Services

Translates technical jargon into practical business communications solutions This book takes readers from traditional voice, fax, video, and data services delivered via separate platforms to a single, unified platform delivering all of these services seamlessly via the Internet. With its clear, jargon-free explanations, the author enables all readers to better understand and assess the growing number of voice over Internet protocol (VoIP) and unified communications (UC) products and services that are available for businesses. VoIP and Unified Communications is based on the author's careful review and synthesis of more than 7,000 pages of published standards as well as a broad range of datasheets, websites, whitepapers, and webinars. It begins with an introduction to IP technology 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 analyses and payback calculations, enabling readers to accurately determine the short- and long-term financial impact of migrating to various VoIP and UC products and services. There's also a chapter detailing major IP systems hardware and software. Throughout the book, diagrams illustrate how various VoIP and UC components and systems work. In addition, the author highlights potential problems and threats to UC services, steering readers away from common pitfalls. Concise and to the point, this text enables readers—from novices to experienced engineers and technical managers—to understand how VoIP and UC really work so that everyone can confidently deal with network engineers, data center gurus, and top management.

Internet Multimedia Communications Using SIP


The most comprehensive book on the shelf about a family of technologies that are cornering the market in enhanced telecommunications services.

SIP Beyond VoIP

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.
Webrtc

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.

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 Example
3.5 Local Media Runnable Code Example

4 Signaling
4.1 The Role of Signaling
4.2 Signaling Transport
4.3 Signaling Protocols
4.4 Summary of Signaling Choices
4.5 Signaling Channel Runnable Code Example
4.6 References

5 Peer-to-Peer Media
5.1 WebRTC Media Flows
5.2 WebRTC and Network Address Translation (NAT)
5.3 STUN Servers
5.4 TURN Servers
5.5 Candidates

6 Peer Connection and Offer/Answer Negotiation
6.1 Peer Connections
6.2 Offer/Answer Negotiation
6.3 JavaScript Offer/Answer Control
6.4 Runnable Code Example: Peer Connection and Offer/Answer Negotiation

7 Data Channel
7.1 Introduction to the Data Channel
7.2 Using Data Channels
7.3 Data Channel Runnable Code Example
7.3.1 Client WebRTC Application

8 W3C Documents
8.1 WebRTC API Reference
8.2 WEBRTC Recommendations
8.3 WEBRTC Drafts
8.4 Related Work
8.5 References

9 NAT and Firewall Traversal
9.1 Introduction to Hole Punching
9.3 WebRTC and Firewalls
9.3.1 WebRTC Firewall Traversal
9.4 References

10 Protocols
10.1 Protocols
10.2 WebRTC Protocol Overview
10.3 References

11 IETF Documents
11.1 Request For Comments
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