Building Telephony Systems With Opensips Second Edition

Open Networks v2 is module 3 of the Free Technology Academy (FTA) Masters programme. Its focus is on the use of GNU/Linux as a networking technology, switching, routing, IPv4 & IPv6, VPNs, services like IP Telephony plus a look at SDN and NFV.

Voice over IP (VoIP) and Internet Multimedia Subsystem technologies (IMS) are rapidly being adopted by consumers, enterprises, governments and militaries. These technologies offer higher flexibility and more features than traditional telephony (PSTN) infrastructures, as well as the potential for lower cost through equipment consolidation and, for the consumer market, new business models. However, VoIP systems also represent a higher complexity in terms of architecture, protocols and implementation, with a corresponding increase in the potential for misuse. In this book, the authors examine the current state of affairs on VoIP security through a survey of 221 known/disclosed security vulnerabilities in bug-tracking databases. We complement this with a comprehensive survey of the state of the art in VoIP security research that covers 245 papers. Juxtaposing our findings, we identify current areas of risk and deficiencies in research focus. This book should serve as a starting point for understanding the threats and risks in a rapidly evolving set of technologies that are seeing increasing deployment and use. An additional goal is to gain a better understanding of the security landscape with respect to VoIP toward directing future research in this and other similar emerging technologies.

This is a practical, hands-on book based around a fictitious case study VoIP Provider that you will build on a development server using OpenSIPS 1.6. The case study grows chapter by chapter, from installing your local development server, right up to the finished VoIP provider. This book is for readers who want to understand how to build a SIP provider from scratch using OpenSIPS. It is suitable for VoIP providers, large enterprises, and universities. Telephony and Linux experience will be helpful but is not essential. Readers need not have prior knowledge of OpenSIPS. This book will also help readers who were using OpenSER but are now confused with the new OpenSIPS.

This book is full of practical code examples aimed at a beginner to ease his or her learning curve.This book is written for IT professionals and enthusiasts who are interested in quickly getting a powerful telephony system up and running using the free and open source application, FreeSWITCH.Telephony experience will be helpful, but not required.

R.H. Sin’s second volume continues the passion and vigor of his previous publication. His stanzas inspire strength through the pure emotional energy and the vulnerability of his poems. Relationships, love, pain, and fortitude are powerfully rendered in his poetry, and his message of perseverance in the face of emotional turmoil cuts to the heart of modern-day life. R.H. Sin's poems are often only a few lines long, and yet the emotional punch of his language gives these words an enduring power beyond the short page. He doesn't back away from the pains and struggles of life and love, and yet his determined, unapologetic voice provides a measure of comfort and a message of perseverance that is at once realistic and indomitable. This blend of determination and painful vulnerability gives his poetry a distinctive, engaging flavor.

Complete Asterisk Training is a new edition of the Configuration Guide for Asterisk PBX. The reason for change the name is to match the name of the online training available on Udemy. So this book is part of a three part training system, eLearning, Text Book and Lab Guide. Why a different book about Asterisk? Most books are not oriented to teach the reader on how to build a complete PBX. They present many concepts, but not with a story, an objective. I have based this book in the old training guides from Novell. So it has a start where you install Asterisk, then you create extensions, trunks, dialplan until you complete a fully functional free and open source PBX. Then we go to more advanced concepts. In this book you will learn: - How to install Asterisk- How to register extensions- How to connect SIP trunks- How to create a dial plan to send and receive calls- How to configure analog and digital channels- How to configure SIP, IAX and PJSIP- How to use Asterisk behind NAT and clients behind NAT- How to use PBX features such as tranfer, capture, parking, conference- How to configure Call queues and Agents - How to generate CDRs to a database using ODBC drivers- How to develop simple AGIs and AMIs to connect your programs- How to secure Asterisk using Fail2Ban, IPTABLES, TLS and SRTP- How to use Asterisk Real Time to read the configuration from a databaseWe cover the latest version, Asterisk 16, a Long Term Support version. I hope you use this version for a long time. This book has more than 10 years, the first edition was in 2006 and since then it has been updated once each 4 or 5 years. This book has two companions. A training on Udemy with the same name and a Lab Guide on github, more details inside the book . I sincerely hope you enjoy. Flavio E. Goncalves

Build high-speed and highly scalable telephony systems using OpenSIPSAbout This Book• Install and configure OpenSIPS to authenticate, route, bill, and monitor VoIP calls• Gain a competitive edge using the most scalable VoIP technology• Discover the latest features of OpenSIPS with practical examples and case studiesWho This Book Is ForIf you want to understand how to build a SIP provider from scratch using OpenSIPS, then this book is ideal for you. It is beneficial for VoIP providers, large enterprises, and universities. This book will also help readers who were using OpenSER but are now confused with the new OpenSIPS.Telephony and Linux experience will be helpful to get the most out of this book but is not essential. Prior knowledge of OpenSIPS is not assumed.What You Will Learn• Learn to prepare and configure a Linux system for OpenSIPS• Familiarise yourself with the installation and configuration of OpenSIPS• Understand how to set a domain and create users/extensions• Configure SIP endpoints and make calls between them• Make calls to and from the PSTN and create access control lists to authorize calls• Install a graphical user interface to simplify the task of provisioning user and system information• Implement an effective billing system with OpenSIPS• Monitor and troubleshoot OpenSIPS to keep it running smoothlyIn DetailOpenSIPS is a multifunctional, multipurpose signalling SIP server. SIP (Session Initiation Protocol) is nowadays the most important VoIP protocol and OpenSIPS is the open source leader in VoIP platforms based on SIP. OpenSIPS is used to set up SIP Proxy servers. The purpose of these servers is to receive, examine, and classify SIP requests. The whole telecommunication industry is changing to an IP environment, and telephony as we know it today will completely change in less than ten years. SIP is the protocol leading this disruptive revolution and it is one of the main protocols on next generation networks. While a VoIP provider is not the only kind of SIP infrastructure created using OpenSIPS, it is certainly one of the most difficult to implement.This book will give you a competitive edge by helping you to create a SIP infrastructure capable of handling tens of thousands of subscribers.Starting with an introduction to SIP and OpenSIPS, you will begin by installing and configuring OpenSIPS. You will be introduced to OpenSIPS Scripting language and OpenSIPS Routing concepts, followed by comprehensive coverage of Subscriber Management. Next, you will learn to install, configure, and customize the OpenSIPS control panel and explore dialplans and routing. You will discover how to manage the dialog module, accounting, NATTraversal, and other new SIP services. The final chapters of the book are dedicated to troubleshooting tools, SIP security, and advanced scenarios including TCP/TLS support, load balancing, asynchronous processing, and more.A fictional VoIP provider is used to explain OpenSIPS and by the end of the book, you will have a simple but complete system to run a VoIP

provider.Style and approachThis book is a step-by-step guide based on the example of a VoIP provider. You will start with OpenSIPS installation and gradually, your knowledge depth will increase.
Build high-speed and highly scalable telephony systems using OpenSIPS About This Book Install and configure OpenSIPS to authenticate, route, bill, and monitor VoIP calls Gain a competitive edge using the most scalable VoIP technology Discover the latest features of OpenSIPS with practical examples and case studies Who This Book Is For If you want to understand how to build a SIP provider from scratch using OpenSIPS, then this book is ideal for you. It is beneficial for VoIP providers, large enterprises, and universities. This book will also help readers who were using OpenSER but are now confused with the new OpenSIPS. Telephony and Linux experience will be helpful to get the most out of this book but is not essential. Prior knowledge of OpenSIPS is not assumed. What You Will Learn Learn to prepare and configure a Linux system for OpenSIPS Familiarise yourself with the installation and configuration of OpenSIPS Understand how to set a domain and create users/extensions Configure SIP endpoints and make calls between them Make calls to and from the PSTN and create access control lists to authorize calls Install a graphical user interface to simplify the task of provisioning user and system information Implement an effective billing

system with OpenSIPS Monitor and troubleshoot OpenSIPS to keep it running smoothly In Detail OpenSIPS is a multifunctional, multipurpose signalling SIP server. SIP (Session Initiation Protocol) is nowadays the most important VoIP protocol and OpenSIPS is the open source leader in VoIP platforms based on SIP. OpenSIPS is used to set up SIP Proxy servers. The purpose of these servers is to receive, examine, and classify SIP requests. The whole telecommunication industry is changing to an IP environment, and telephony as we know it today will completely change in less than ten years. SIP is the protocol leading this disruptive revolution and it is one of the main protocols on next generation networks. While a VoIP provider is not the only kind of SIP infrastructure created using OpenSIPS, it is certainly one of the most difficult to implement. This book will give you a competitive edge by helping you to create a SIP infrastructure capable of handling tens of thousands of subscribers. Starting with an introduction to SIP and OpenSIPS, you will begin by installing and configuring OpenSIPS. You will be introduced to OpenSIPS Scripting language and OpenSIPS Routing concepts, followed by comprehensive coverage of Subscriber Management. Next, you will learn to install, configure, and customize the OpenSIPS control panel and explore dialplans and routing. You will discover how to manage the dialog module, accounting, NATTraversal, and other new SIP services. The final chapters of the book are dedicated to troubleshooting tools, SIP security, and advanced scenarios including TCP/TLS support, load balancing, asynchronous processing, and more. A fictional VoIP provider is used to explain OpenSIPS and by the end of the book, you will have a simple but complete system to run a VoIP provider. Style and approach This book is a step-by-step guide based on the example of a VoIP provider. You will start with OpenSIPS installation and gradually, your knowledge depth will increase.

This book presents the state of the art technologies and solutions to tackle the critical challenges faced by the building and development of the WSN and ecological monitoring system but also potential impact on society at social, medical and technological level. This book is dedicated to Sensing systems for Sensors, Wireless Sensor Networks and Ecological Monitoring. The book aims at Master and PhD degree students, researchers, practitioners, especially WSN engineers involved with ecological monitoring. The book will provide an opportunity of a dedicated and a deep approach in order to improve their knowledge in this specific field.

This book constitutes the refereed proceedings of the 5th International Conference on Autonomous Infrastructure, Management and Security, AIMS 2011, held in Nancy, France, in June 2011. The 11 revised full papers presented together 11 papers of the AIMS PhD workshops were carefully reviewed and selected from numerous submissions. The papers are organized in topical sections on security management, autonomic network and service management (PhD workshop), policy management, P2P and aggregation schemes, and monitoring and security (PhD workshop).

This new book is the first to deliver a comprehensive yet concise critical appraisal of broadband systems for high-speed Internet access. It gives you a demystifying, lightly-technical overview of existing and future mobile and fixed systems that use gigahertz or terahertz frequencies, and serves as a handy reference for information on most high-speed Internet support systems that are currently in use or will come into use before 2015.

This bestselling guide makes it easy to learn how to design a complete Voice over IP (VoIP) or traditional PBX system with Asterisk, with a detailed roadmap that shows readers how to install and configure this open source software, whether upgrading an existing phone system or starting from scratch.

This book is a well illustrated, step-by-step guide to building a SIP based network using OpenSER. This book is for readers who want to understand how to build a SIP provider from scratch using OpenSER. Telephony and Linux experience will be helpful but is not essential. Readers need not have prior knowledge of OpenSER.

Build robust high-performance telephony systems using FreeSWITCH

How to Start a VoIP Business is the first book which explains in plain English how to become a VoIP provider and start different services, based on a VoIP technology. This simple six-stage guide will give you the know-how of launching services, such as mobile VoIP, callback, calling cards, call shops, residential VoIP, virtual PBX, SIP trunking, wholesale transit, call origination and call termination.

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.

For courses in fluid mechanics. Introduces engineering students to the principles of fluid mechanics. Written and conceived by an author with decades of relevant experience in the fields of fluid mechanics, engineering, and related disciplines, this First Edition of Fluid Mechanics for Engineers effectively introduces engineering students to the principles of fluid mechanics. With the understanding that fluid mechanics is a required core course for most engineering students, the author focuses first and foremost on the most essential topics of the field. Practical applications for several engineering disciplines are considered, with a special focus on civil engineering. Elective topics are also included for instructors' consideration with regard to specific courses. Written in a stimulating style, Fluid Mechanics for Engineers fulfills the requirements of a core course while keeping students engaged. Pearson Mastering Engineering™ not included. Students, if Pearson Mastering Engineering is a recommended/mandatory component of the course, please ask your instructor for the correct ISBN and course ID. Pearson Mastering Engineering should only be purchased when required by an instructor. Instructors, contact your Pearson representative for more information. Pearson Mastering Engineering is an online homework, tutorial, and assessment program designed to work with this text to engage students and improve results. Interactive, self-paced tutorials provide individualized coaching to help students stay on track. With a wide range of activities available, students can actively learn, understand, and retain even the most difficult concepts.

Formerly released as a three-volume set, this authoritative reference remains the most comprehensive single source on spread spectrum systems-which are just beginning to find important new applications such as CDMA cellular networks & wireless Personal Communication Networks. Although theory is covered where appropriate, readers will find the focus is on practical issues, including an in-depth look at multiple access communications & positioning systems applications.

FreeSWITCH is an open source carrier-grade telephony platform designed to facilitate the creation of voice, chat, and video applications, via phones and web browsers. It is scalable, carrier-ready, and easy-to-program for converged communication and

VoIP. The technology serves SIP, WebRTC, PSTN, FAX, PBX, VERTO, and all the relevant channels essential to stay connected in today's world. In the FreeSWITCH 1.6 Cookbook, members of the FreeSWITCH development team share some of their hard-earned knowledge with you. Use this knowledge to improve and expand your FreeSWITCH installations.

The classic reader that has introduced millions of students to the essay as a genre.

"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
Provides information on Asterisk, an open source telephony application.
Build a robust, high-performance telephony system with FreeSWITCH About This Book Learn how to install and configure a complete telephony system of your own, from scratch, using FreeSWITCH 1.6 Get in-depth discussions of important concepts such as dialplan, user directory, NAT handling, and the powerful FreeSWITCH event socket Discover expert tips from the FreeSWITCH experts, including the creator of FreeSWITCH—Anthony Minessale Who This Book Is For This book is for beginner-level IT professionals and enthusiasts who are interested in quickly getting a powerful telephony system up and running using FreeSWITCH. It would be good if you have some telephony experience, but it's not a must. What You Will Learn Build a complete WebRTC/SIP VoIP platform able to interconnect and process audio and video in real time Use advanced PBX features to create powerful dialplans Understand the inner workings and architecture of FreeSWITCH

Real time configuration from database and webserver with mod_xml_curl Integrate browser clients into your telephony service Use scripting to go beyond the dialplan with the power and flexibility of a programming language Secure your FreeSWITCH connections with the help of effective techniques Deploy all FreeSWITCH features using best practices and expert tips Overcome frustrating NAT issues Control FreeSWITCH remotely with the all-powerful event socket Trace packets, check debug logging, ask for community and commercial help In Detail FreeSWITCH is an open source telephony platform designed to facilitate the creation of voice and chat-driven products, scaling from a soft-phone to a PBX and even up to an enterprise-class soft-switch. This book introduces FreeSWITCH to IT professionals who want to build their own telephony system. This book starts with a brief introduction to the latest version of FreeSWITCH. We then move on to the fundamentals and the new features added in version 1.6, showing you how to set up a basic system so you can make and receive phone calls, make calls between extensions, and utilize basic PBX functionality. Once you have a basic system in place, we'll show you how to add more and more functionalities to it. You'll learn to deploy the features on the system using unique techniques and tips to make it work better. Also, there are changes in the security-related components, which will affect the content in the book, so we will make that intact with the latest version. There are new support libraries introduced, such as SQLite, OpenSS, and more, which will make FreeSWITCH more efficient and add more functions to it. We'll cover these in the new edition to make it more appealing for you. Style and approach This easy-to-follow guide helps you understand every topic easily using real-world examples of FreeSWITCH tasks. This book is full of practical code so you get a gradual learning curve. Design a complete Voice over IP (VoIP) or traditional PBX system with Asterisk, even if you have only basic telecommunications knowledge. This bestselling guide makes it easy with a detailed roadmap that shows you how to install and configure this open source software, whether you’re upgrading your existing phone system or starting from scratch. Ideal for Linux administrators, developers, and power users, this updated fifth edition shows you how to write a basic dialplan step-by-step and brings you up to speed on the features in Asterisk 16, the latest long-term support release from Digium. You’ll quickly gain working knowledge to build a simple yet inclusive system. Integrate Asterisk with analog, VoIP, and digital telephony systems Build an interactive dialplan using best practices for more advanced features Delve into voicemail options such as storing messages in a database Connect to external services including

Google Hangouts, XMPP, and calendars Incorporate Asterisk features and functions into a relational database to facilitate information sharing Learn how to use Asterisk's security, call routing, and faxing features Monitor and control your system with the Asterisk Manager Interface (AMI)

Building Telephony Systems with OpenSIPSPackt Publishing Ltd

Design a complete Voice over IP (VoIP) or traditional PBX system with Asterisk, even if you have only basic telecommunications knowledge. This bestselling guide makes it easy, with a detailed roadmap that shows you how to install and configure this open source software, whether you're upgrading your existing phone system or starting from scratch. Ideal for Linux administrators, developers, and power users, this updated edition shows you how to write a basic dialplan step-by-step, and brings you up to speed on the features in Asterisk 11, the latest long-term support release from Digium. You'll quickly gain working knowledge to build a simple yet inclusive system. Integrate Asterisk with analog, VoIP, and digital telephony systems Build an interactive dialplan, using best practices for more advanced features Delve into voicemail options, such as storing messages in a database Connect to external services including Google Talk, XMPP, and calendars Incorporate Asterisk features and functions into a relational database to facilitate information sharing Learn how to use Asterisk's security, call routing, and faxing features Monitor and control your system with the Asterisk

Manager Interface (AMI) Plan for expansion by learning tools for building distributed systems

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.

The book includes the insights that reflect ‘Advances in Computer and Computational Sciences’ from upcoming researchers and leading academicians across the globe. It contains the high-quality peer-reviewed papers of ‘International Conference on Computer, Communication and Computational Sciences (IC4S 2017), held during 11–12 October, 2017 in Thailand. These papers are arranged in the form of chapters. The content of this book is divided into two volumes that cover variety of topics such as intelligent hardware and software design, advanced communications, intelligent computing techniques, intelligent image processing, and web and informatics. This book helps the perspective readers’ from computer industry and academia to derive the advances of next generation computer and communication technology and shape them into real life applications.

This book constitutes the refereed conference proceedings of the 20th International Symposium on Research in Attacks, Intrusions, and Defenses, RAID 2017, held in Atlanta, GA, USA, in September 2017. The 21 revised full papers were selected from 105 submissions. They are organized in the following topics: software security, intrusion detection, systems security, android security, cybercrime, cloud security, network security.

The award-winning debut novel by young Mexican author Aura Xilonen, The Gringo Champion is a thrillingly inventive story about crossing borders that the Los Angeles Review of Books called "one of the must-read books of 2017." Liborio has to leave Mexico, a land that has taught him little more than a keen instinct for survival. He crosses the Rio Bravo, like so many others, to reach "the promised land." And in a barrio like any other, in some gringo city, this illegal immigrant tells his story. As Liborio narrates his memories we discover a childhood scarred by malnutrition and abandonment, an adolescence lived with a sense of having nothing to lose. In his new home, he finds a job at a bookstore. He falls in love with a woman so intensely that his fantasies of her verge on obsession. And, finally, he finds himself on a path that just might save him: he becomes a boxer. This is a migrant's story of deracination, loneliness, fear, and finally, love told in a sparkling, innovative prose. It's Million Dollar Baby meets The Brief Wondrous Life of Oscar Wao, and a story of migration and hope that is as topical as it is timeless.

Irrespective of whether we use economic or societal metrics, the Internet is one of the most important technical infrastructures in existence today. It will serve as a catalyst for much of our innovation and prosperity in the future. A competitive Europe will require Internet connectivity and services beyond the capabilities offered by current technologies. Future Internet research is therefore a must. The Future Internet Assembly (FIA) is a successful and unique bi-annual conference that brings together participants of over 150 projects from several distinct but interrelated areas in the EU Framework Programme 7. The 20 full papers included in this volume were selected from 40 submissions, and are preceded by a vision paper describing the FIA Roadmap. The papers have been organized into topical sections on the foundations of Future Internet, the applications of Future Internet, Smart Cities, and Future Internet infrastructures.

This book constitutes the proceedings of the 17th International Conference on Detection of Intrusions and Malware, and Vulnerability Assessment, DIMVA 2020, held in Lisbon, Portugal, in June 2020.* The 13 full papers presented in this volume were carefully reviewed and selected from 45 submissions. The contributions were organized in topical sections named: vulnerability discovery and analysis; attacks; web security; and detection and containment. ?*The conference was held virtually due to the COVID-19 pandemic.

This book proposes new technologies and discusses future solutions for ICT design infrastructures, as reflected in high-quality papers presented at the 4th International Conference on ICT for Sustainable Development (ICT4SD 2019), held in Goa, India, on 5–6 July 2019. The conference provided a valuable forum for cutting-edge research discussions among pioneering researchers, scientists, industrial engineers, and students from all around the world. Bringing together experts from different countries, the book explores a range of central issues from an international perspective. Master the art of advanced VoIP and WebRTC communication with the most dynamic application server, FreeSWITCH About This Book Forget the hassle - make FreeSWITCH work for you Discover how FreeSWITCH integrates with a range of tools and APIs From high availability to IVR development use this book to become more confident with this useful communication software Who This Book Is For SysAdmins, VoIP engineers – whoever you are, whatever you're trying to do, this book will help you get more from FreeSWITCH. What You Will Learn Get to grips with the core concepts of FreeSWITCH Learn FreeSWITCH high availability Work with SIP profiles, gateways, ITSPs, and Codecs optimization Implement effective security on your projects Master audio manipulation and recording Discover how FreeSWITCH works alongside WebRTC Build your own complex IVR and PBX applications Connect directly to PSTN/TDM Create your own FreeSWITCH module Trace SIP packets with the help of best open source tools Implement Homer Sipcapture to troubleshoot and debug all your platform traffic In Detail FreeSWITCH is one of the best tools around if you're looking for a modern method of managing communication protocols through a range of different media. From real-time browser communication with the WebRTC API to implementing VoIP (voice over internet protocol), with FreeSWITCH you're in full control of your projects. This book shows you how to unlock its full potential – more than just a tutorial, it's packed with plenty of tips and tricks to make it work for you. Written by members of the team who actually helped build FreeSWITCH, it will guide you through some of the newest features of version 1.6 including video transcoding and conferencing. Find out how FreeSWITCH interacts with other tools and APIs, learn how to tackle common (and not so common) challenges ranging from high availability to IVR development and programming advanced PBXs. Great communication functionality begins with FreeSWITCH – find out how and get your project up and running today. Style and approach Find out how it works, then put your knowledge into practice - that's how this advanced FreeSWITCH guide has been designed to help you learn. You'll soon master FreeSWITCH and be confident using it in your projects.

This is a problem-solution approach to take your FreeSWITCH skills to the next level, where everything is explained in a practical way. If you are a system administrator, hobbyist, or someone who uses FreeSWITCH on a regular basis, this book is for you. Whether you are a FreeSWITCH expert or just getting started, this book will take your skills to the next level.

Borrowing from nature is nothing new; designers have been looking to the natural world for inspiration for centuries. For some, this has been purely on an aesthetic level, for a select few on a functional level. Fewer still combine these considerations to produce designs that mimic both the forms and properties of natural structures. This volume chronicles the growing field of bionic architecture; structures that look to do just that.