



Voice over IP Security

Security best practices derived from deep analysis of the latest VoIP network threats

ciscopress.com

Patrick Park

Voice over IP Security

Patrick Park

Copyright © 2009 Cisco Systems, Inc.

Published by: Cisco Press 800 East 96th Street Indianapolis, IN 46240 USA

All rights reserved. No part of this book may be reproduced or transmitted in any form or by any means, electronic or mechanical, including photocopying, recording, or by any information storage and retrieval system, without written permission from the publisher, except for the inclusion of brief quotations in a review.

Printed in the United States of America
First Printing September 2008
Library of Congress Cataloging-in-Publication data
Park, Patrick, 1971Voice over IP security / Patrick Park.
p. cm.
ISBN 978-1-58705-469-3 (pbk.)
1. Internet telephony--Security measures. I. Title. II. Title: VoIP security.

TK5105.8865.P37 2008 004.69'5--dc22

2008036070

ISBN-13: 978-1-58705-469-3 ISBN-10: 1-58705-469-8

Warning and Disclaimer

This book is designed to provide information about Voice over IP security. Every effort has been made to make this book as complete and as accurate as possible, but no warranty or fitness is implied.

The information is provided on an "as is" basis. The authors, Cisco Press, and Cisco Systems, Inc. shall have neither liability nor responsibility to any person or entity with respect to any loss or damages arising from the information contained in this book or from the use of the discs or programs that may accompany it.

The opinions expressed in this book belong to the author and are not necessarily those of Cisco Systems, Inc.

Trademark Acknowledgments

All terms mentioned in this book that are known to be trademarks or service marks have been appropriately capitalized. Cisco Press or Cisco Systems, Inc., cannot attest to the accuracy of this information. Use of a term in this book should not be regarded as affecting the validity of any trademark or service mark.

Corporate and Government Sales

The publisher offers excellent discounts on this book when ordered in quantity for bulk purchases or special sales, which may include electronic versions and/or custom covers and content particular to your business, training goals, marketing focus, and branding interests. For more information, please contact: **U.S. Corporate and Government Sales** 1-800-382-3419 corpsales@pearsontechgroup.com

For sales outside the United States please contact: International Sales international@pearsoned.com

Feedback Information

At Cisco Press, our goal is to create in-depth technical books of the highest quality and value. Each book is crafted with care and precision, undergoing rigorous development that involves the unique expertise of members from the professional technical community.

Readers' feedback is a natural continuation of this process. If you have any comments regarding how we could improve the quality of this book, or otherwise alter it to better suit your needs, you can contact us through email at feedback@ciscopress.com. Please make sure to include the book title and ISBN in your message.

Paul Boger

We greatly appreciate your assistance.

Publisher
Associate Publisher
Cisco Press Program Manager
Executive Editor
Managing Editor
Development Editor
Project Editor
Copy Editor
Technical Editors
Editorial Assistant
Designer
Composition
Indexer

Dave Dusthimer Jeff Brady Brett Bartow Patrick Kanouse Dan Young Seth Kerney Margaret Berson Bob Bell Dan Wing Vanessa Evans Louisa Adair Octal Publishing, Inc. WordWise Publishing Services LLC Water Crest Publishing, Inc.

ıllıılı cısco

Proofreader

Americas Headquarters Cisco Systems, Inc. 170 West Tasman Drive San Jose, CA 95134-1706 USA www.cisco.com Tel: 408 526-4000 800 553-NETS (6387) Fax: 408 527-0883 Asia Pacific Headquarters Cisco Systems, Inc. 168 Robinson Road #28-01 Capital Tower Singapore 068912 www.cisco.com Tel: +86 6317 7777 Fax: +65 6317 7799

Europe Headquarters Cisco Systems International BV Haarlerbergpark Haarlerbergweg 13-19 1101 CH Amsterdam The Netherlands www-europe.cisco.com Tet: +31 0 800 020 07911 Fax: +31 0 20 357 1100

Cisco has more than 200 offices worldwide. Addresses, phone numbers, and fax numbers are listed on the Cisco Website at www.cisco.com/go/offices.

©2007 Cisco Systems. Inc. All rights reserved. CCVP the Cisco logo, and the Cisco Square Bridge logo are trademarks of Cisco Systems. Inc: Changing the Way Wa Work, Live, Play, and Learn is a service mark of Cisco Systems. Inc: and Access Registrar. Aronet, BPX. Catalyst. CCDA. CCDP. CCIP. CCIP. CCNP. CCSP. Cisco. The Cisco Systems. Inc: Changing the Way Wa Work, Live, Play, and Learn is a service mark of Cisco Systems logo. Cisco Linity. Enterprise/Solve: EtherChannel. EtherFast: EtherSwitch. Fast Step, Follow Me Browsing. FormShare. GigaDrive, GigaStack. HomeLink. Internet Quotient. 105. IPTV. 10 Expertise. The I Ologo. I O Net Readiness Scorecard. JOuk Study, LightStream. Linksys. MeetingPlace. MGX. Networking Academy. Network: Registrar. Packet, PIX. ProConnett, RateMUX. ScriptShare. SildeCast. SMARTinet. StackWise. The Fastest Way to Increase Vour Internet Quotient. and TransPeth are registered trademarks of Cisco Systems. Inc. and/or its affiliates in the United States countries.

All other trademarks mentioned in this document or Website are the property of their respective owners. The use of the word partner does not imply a partnership relationship between Cisco and any other company. (0609R)

Introduction

Voice over Internet Protocol (VoIP) has been popular in the telecommunications world since its emergence in the late 90s, as a new technology transporting multimedia over the IP network. In this book, the multimedia (or rich media) includes not only voice, but also video, instant message, presence data, and fax data over the IP network.

Today people commonly make phone calls with IP phones or client software (such as Skype or iChat) on their computer, or send instant messages to their friends. This gives them convenience and cost savings. Many telecommunications companies and other organizations have been switching their legacy phone infrastructure to a VoIP network, which reduces costs for lines, equipment, manpower, and maintenance.

However, the benefits of VoIP are not free. There are disadvantages to using VoIP. The integrated rich media makes it difficult to design the network architecture. Multiple VoIP protocols and different methods of implementation create serious interoperability issues. Integration with existing data networks creates quality of service issues. The fact that so many network elements are involved through open (or public) networks creates serious security issues, because each element and network has vulnerable factors.

The security issues especially are becoming more serious because traditional security devices (such as firewalls) and protocols (such as encryption) cannot protect VoIP services or networks from recent intelligent threats.

This book focuses on the important topic of VoIP security by analyzing current and potential threats to demonstrating the methods of prevention.

Goals and Methods

The most important goal of this book is to give you correct and practical answers for the following questions:

- What are the current and potential threats?
- What are the impacts of those threats?
- Why are current data security devices not able to protect against recent intelligent threats?
- How can you protect VoIP services and networks from those threats?
- What is lawful interception and how do you implement it?

One key methodology used in this book is to give you hands-on experience of current well-known threats by simulating them with publicly available tools. Through the simulation, you can realize the characteristics and impacts of those threats and have a better understanding of mitigation.

Another key methodology is to give you detailed examples of protection methods with protocols, products, and architecture so that you may apply them to real VoIP service environments.

This book also gives you clarification of VoIP security concepts, definitions, standards, requirements, limitations, and related terms.

Who Should Read This Book

This book is NOT designed to give you information about VoIP in general which is available almost everywhere. Instead, this book focuses on VoIP security and gives practical information to people like those in the following list:

- Managers or engineers who are planning to employ VoIP systems in their organizations
- System engineers or architects who design and implement VoIP networks
- · Network administrators who administer, upgrade, or secure networks that include VoIP elements
- Security consultants who perform security assessments for VoIP environments
- Developers who implement VoIP products or solutions
- · Researchers and analysts who are interested in VoIP security

This book assumes that the readers have some minimal knowledge of networking (such as TCP/IP), operating systems, and VoIP in general (such as IP phones).

How This Book Is Organized

Although this book could be read cover to cover, it is designed to be flexible and allow you to easily move between chapters and sections of chapters to cover just the material that you need more work with.

This book consists of three parts. Part I, "VoIP Security Fundamentals," contains Chapters 1 through 5 and covers VoIP security fundamentals that are essential to understand current threats and security practices. Part II, "VoIP Security Best Practices," contains Chapters 6 through 9 and demonstrates VoIP security best practices with the detailed analysis and simulation of current threats. Part III, "Lawful Interception (CALEA)," contains Chapters 10 through 11 and covers another aspect of VoIP security, Lawful Interception, from basic concept to real implementation.

Chapter 1, "Working with VoIP," provides an overview of VoIP and its vulnerability in general. Chapters 2 through 11 are the core chapters and can be read in any order. If you do intend to read them all, the order in the book is an excellent sequence to use.

The core chapters, Chapters 2 through 11, cover the following topics:

- **Chapter 2, "VoIP Threat Taxonomy"**—This chapter defines VoIP threat taxonomy, based on four different categories: threats against availability, confidentiality, integrity, and social context. This chapter is not intended to provide exhaustive lists of current and potential threats, but to define the taxonomy for identifying the threat in the first place, measuring the current and potential impact, and helping implementers to develop protection methods and secure service architecture. Twenty-two typical threats are introduced with examples and features.
- Chapter 3, "Security Profiles in VoIP Protocols"—This chapter introduces the security profiles of VoIP protocols: SIP, H.323, and MGCP. The content shows how each protocol defines specific security mechanisms and recommends combined solution with other security protocols, such as IPSec, TLS, and SRTP.

- **Chapter 4, "Cryptography"**—This chapter provides a high-level understanding of cryptographic algorithms with comprehensible figures, avoiding mathematical details. Well-known cryptographic algorithms are introduced, such as DES, 3DES, AES, RAS, DSA, and hash functions (MD5, SHA, and HMAC). This chapter also covers the mechanism of key management, focusing on key distribution.
- **Chapter 5, "VoIP Network Elements"**—This chapter covers what devices are involved in the VoIP network architecture, and how they work for secure services. Session Border Controller, VoIP-aware firewalls, NAT servers, lawful interception servers, customer premise equipment, call processing servers, and media gateways are introduced.
- **Chapter 6, "Analysis and Simulation of Current Threats"**—This chapter covers two main topics: detailed analysis and hands-on simulation of most common threats, and the guidelines for mitigation. For the analysis, it examines the detailed patterns, usage examples, and impacts of the threats. For the simulation, it introduces negative testing tools that are available on the Internet so that you can have hands-on experience. The threats that this chapter covers are DoS, malformed messages, sniffing (eavesdropping), spoofing (identity theft), and VoIP spam (voice, instant message, and presence spam).
- **Chapter 7, "Protection with VoIP Protocol"**—This chapter demonstrates the details of how to make VoIP service secure with SIP and other supplementary protocols. It focuses on the methodology of protection in these five categories: authentication, encryption, transport and network layer security, threat model and prevention, and limitations.
- Chapter 8, "Protection with Session Border Controller"—This chapter examines security issues on the VoIP network borders, and provides the methodology of preventing the issues with an SBC. This chapter includes the details of SBC functionality (such as network topology hiding, DoS protection, overload prevention, NAT traversal, and lawful interception), as well as the method of designing service architecture with an SBC in terms of high availability, secure network connectivity, virtualization, and optimization of traffic flow.
- Chapter 9, "Protection with Enterprise Network Devices"—This chapter demonstrates how to protect the enterprise VoIP network with Cisco devices for practical information. Cisco firewalls, Unified Communications Manager, Unified Communications Manager Express, IP phone, and multilayer switches are used. This chapter includes security features, usage examples, and configuration guidelines for those devices.
- **Chapter 10, "Lawful Interception Fundamentals"**—This chapter covers the fundamentals of lawful interception. The topics are definition, background information, requirements from law enforcement agents, the reference model from an architectural perspective, functional specifications, request/response interface, and operational considerations.
- **Chapter 11, "Lawful Interception Implementation"**—This chapter demonstrates how to implement lawful interception into the VoIP service environment. It focuses on how the interception request and response work between functional modules, based on industry specifications.



VoIP Threat Taxonomy

The VoIP vulnerabilities that were introduced in Chapter 1, "Working with VoIP," can be exploited to create many different kinds of threats. Attackers may disrupt media service by flooding traffic, collect privacy information by intercepting call signaling or call content, hijack calls by impersonating servers or impersonating users, make fraudulent calls by spoofing identities, and so on.

Spammers may utilize VoIP networks to deliver spam calls, instant messages, or presence information, which are more effective than email spams because it is very difficult to filter VoIP spam.

This chapter is not intended to provide exhaustive lists of current and potential threats, but to define the taxonomy for the following purposes:

- To identify the threat in the first place
- To measure the current impact and potential future impact of the threat
- To help develop the protection method and design a secure service architecture

For an exhaustive list of all current and potential threats, go to www.voipsa.org (Voice over IP Security Alliance).

There are many possible ways to categorize the threats. This book uses the following four categories that most VoIP threats can belong to:

Threats against availability

NOTE

- Threats against confidentiality
- Threats against integrity
- Threats against social context

Each section in this chapter covers each category with typical threat examples. To give you a better understanding, each section uses figures and protocol examples with Session Initiation Protocol (SIP).

NOTE This chapter approaches these threats at a high level, focusing on the taxonomy. If you want to see a detailed analysis with simulation, refer to Chapter 6, "Analysis and Simulation of Current Threats."

The following section introduces the most critical threats that impact service availability.

Threats Against Availability

Threats against availability are actually a group of threats against service availability that is supposed to be running 24/7 (24 hours, 7 days a week). That is, these threats aim at VoIP service interruption, typically in the form of Denial of Service (DoS).

The typical threats against availability are as follows:

- Call flooding
- Malformed messages (protocol fuzzing)
- Spoofed messages (call teardown, toll fraud)
- Call hijacking (registration or media session hijacking)
- Server impersonating
- Quality of Service (QoS) abuse

The following subsections describe the threats with examples, which show you how they impact service availability.

Call Flooding

The typical example of DoS is intentional call flooding; an attacker floods valid or invalid heavy traffic (signals or media) to a target system (for example, VoIP server, client, and underlying infrastructure), and drops the performance significantly or breaks down the system. The typical methods of flooding are as follows:

- Valid or invalid registration flooding—An attacker uses this method commonly because most registration servers accept the request from any endpoints in the public Internet as an initial step of authentication. Regardless of whether the messages are valid or invalid, the large number of request messages in a short period of time (for example, 10,000 SIP REGISTER messages per second) severely impacts the performance of the server.
- Valid or invalid call request flooding—Most VoIP servers have a security feature that blocks flooded call requests from unregistered endpoints. So, an attacker registers first after spoofing a legitimate user, and then sends flooded call requests in a short

period of time (for example, 10,000 SIP INVITE messages per second). This impacts the performance or functionality of the server regardless of whether the request message is valid or not.

- Call control flooding after call setup—An attacker may flood valid or invalid call control messages (for example, SIP INFO, NOTIFY, Re-INVITE) after call setup. Most proxy servers are vulnerable because they do not have a security feature to ignore and drop those messages.
- **Ping flooding**—Like Internet Control Message Protocol (ICMP) ping, VoIP protocols use ping messages in the application layer to check out the availability of a server or keep the pinhole open in the local Network Address Translation (NAT) server, such as SIP OPTIONS message. Most IP network devices (for example, a router or firewall) in the production network do not allow ICMP pings for security reasons. However, many VoIP servers should allow the application-layer ping for proper serviceability, which could be a critical security hole.

Figure 2-1 illustrates the example of distributed flooding with zombies; an attacker compromises other computers with malware (for example, a virus) and uses them as zombies flooding registration messages. Each zombie sends 1,000 SIP REGISTER messages per second with different credentials that are randomly generated.

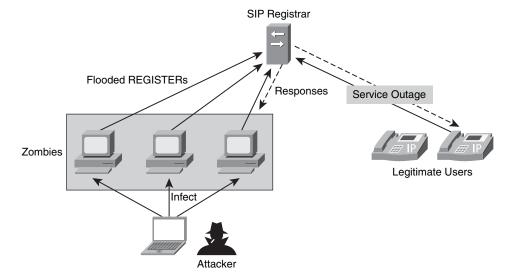


Figure 2-1Call Flooding Example

In Figure 2-1, the flooded messages will impact the registration server (SIP Registrar) severely as long as the server processes and replies with any error codes, such as "401 Unauthorized," "404 Not Found," "400 Bad Request," and so on. The impact can be high resource consumption (for example, CPU, memory, network bandwidth), system malfunction,

or service outage. Whether the server responds or not, flooding the SIP registrar with sufficient registration messages will result in the degradation of service to the legitimate endpoints.

Not only the intentional flooding just mentioned, but also unintentional flooding exists in VoIP networks, so-called "self-attack," because of incorrect configuration of devices, architectural service design problems, or unique circumstances. Here are some examples:

- **Regional power outage and restoration**—When the power is backed up after a regional outage, all endpoints (for example, 10,000 IP phones) will boot up and send registration messages to the server almost at the same time, which are unintentional flooded messages. Because those phones are legitimate and distributed over a wide area, it is hard to control the flooding traffic proactively.
- **Incorrect configuration of device**—The most common incorrect configuration is setting endpoint devices (for example, IP phones) to send too many unnecessary messages, such as a registration interval that is too short.
- **Misbehaving endpoints**—Problematic software (firmware) or hardware could create unexpected flooding, especially when multiple or anonymous types of endpoints are involved in the VoIP service network.
- Legitimate call flooding—There are unusual days or moments when many legitimate calls are made almost at the same time. One example is Mother's Day, when a lot of calls are placed in the United States. Another example is natural disasters (for example, earthquakes), when people within the area make a lot of calls to emergency numbers (for example, 911) and their family and friends make calls to the affected area at the same time.

Those types of intentional and unintentional call flooding are common and most critical threats to VoIP service providers, who have to maintain service availability continually.

The next type is another form of threat against service availability, by means of malformed messages.

Malformed Messages (Protocol Fuzzing)

An attacker may create and send malformed messages to the target server or client for the purpose of service interruption. A *malformed message* is a protocol message with wrong syntax. Example 2-1 shows an example with a SIP INVITE message.

NOTE Protocol fuzzing is another name for malformed messages. A small difference is that protocol fuzzing includes malicious messages that have correct syntax but break the sequence of messages, which may cause system error by making the state machine confused.

Example 2-1 Malformed SIP INVITE Message

```
Request-URI: aaaaaaaaa sip:1001@192.168.10.10 SIP/2.0
Message Header
    Via: SIP/2.0/UDP CAL-D600-5814.cc-
ntd1.example.com:5060;branch=z9hG4bK00002000005
    To: Receiver <sip:1001@192.168.10.10>
   555555556@CAL-D600-5814.cc-ntd1.example.com
   CSeq: 1 INVITE
    Contact: 2 <sip:user@CAL-D600-5814.cc-ntd1.example.com>
    Expires: 1200
    Max-Forwards: 70
    Content-Type: application/sdp
    Content-Length: 143
Message body
Session Description Protocol
    Session Description Protocol Version (v): = = = = = 0
    Owner/Creator, Session Id (o): 2 2 2 IN IP4 CAL-D600-5814.cc-ntd1.example.com
    Session Name (s): Session SDP
    Connection Information (c): IN IP4 192.168.10.10
    Time Description, active time (t): 0 0 % \left( \left( {{{\bf{0}}} \right) } \right)
    Media Description, name and address (m): audio 9876 RTP/AVP 0
    Media Attribute (a): rtpmap:0 PCMU/8000
```

Note that the comments (bold letters) in Example 2-1 are not shown in the actual SIP INVITE message. You can find something wrong in the example of an INVITE message. Three SIP headers (Request-URI, From, and Call-Id) and one version in Session Description Protocol (SDP) have the wrong format.

The server receiving this kind of unexpected message could be confused (fuzzed) and react in many different ways depending on the implementation. The typical impacts are as follows:

- Infinite loop of parsing
- Buffer overflow, which may permit execution of arbitrary code
- Break state machine
- Unable to process other normal messages
- System crash

This vulnerability comes from the following sources in general:

1 Weakness of protocol specification

Most VoIP protocols are open to the public and don't strictly define every single line. Attackers could find where the weakness of syntax is. Additionally, there are many customizable fields or tags.

2 Ease of creating the malformed message

Creating a message like that in Example 2-1 is easy for regular programmers. Even for nonprogrammers, many tools are available to make customized messages.

3 Lack of exception handling in the implementation

Because of time restrictions, most implementers are apt to focus on product features and interfaces, rather than create exception handling for massive negative cases.

4 Difficulty of testing all malformed cases

It is very difficult to test all the negative cases, even though sophisticated testing tools covering more cases are coming out these days.

The threat of malformed messages should be preventable as long as the parsing algorithm handles them properly.

The next threat is spoofed messages that are not malformed but still impact service availability.

Spoofed Messages

An attacker may insert fake (spoofed) messages into a certain VoIP session to interrupt the service, or insert them to steal the session. The typical examples are "call teardown" and "toll fraud."

Call Teardown

The method of malicious call teardown is that an attacker monitors a SIP dialog and obtains session information (Call-ID, From tag, and To tag), and sends a call termination message (for example, SIP BYE) to the communication device while the users are talking. The device receiving the termination message will close the call session immediately. Figure 2-2 illustrates the example with SIP messages.

Figure 2-2 Malicious Call Teardown

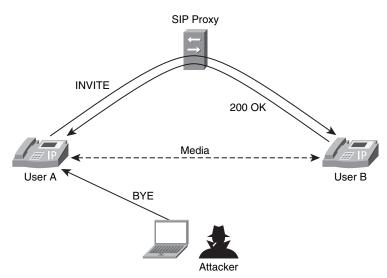


Figure 2-2 assumes that the attacker already monitored call signals between User A and B, and knew the session information (SIP dialog). The attacker injects the session information to the BYE message. The IP phone of user A receives the BYE and disconnects the media channel.

Another method of attack is that an attacker sends the termination messages to random devices (especially, proxy server) without knowing session information, which may affect current call sessions.

Compared to previous threats in this section, the malicious call teardown is not a common attack because the attacker should monitor the target call session before sending a termination message (BYE).

The next type of attack, toll fraud, also requires preliminary information like credentials before making fraud calls, but it happens commonly because of monetary benefit.

Toll Fraud

A fraudulent toll call is one of the common threats these days, especially for long distance or international calls. Because most mediation devices (for example, public switched telephone network [PSTN] media gateway, proxy server) require valid credentials (for example, ID and password) before setting up the toll call, an attacker collects the credentials first in many different ways. Typically, an attacker creates spoofed messages for brute-force password assault on the server until he receives authorization. If the clients use default passwords or easy-to-guess passwords, it is much easier to find them, especially when an attacker uses a password dictionary (see Note).

NOTE

A *password dictionary* is a file that contains millions of frequently used passwords. Most passwords are manually created by humans (rather than by computers), so it's highly likely that they will be simple and easy to remember. No one really wants to have to remember random passwords that are longer than 10 digits, except perhaps system administrators. For example, a user named John Kim is apt to have passwords such as "jkim," "iamjohn," "johnkim," "john2kim," "john4me," and so on. Therefore, an attacker using a password dictionary containing millions of commonly used passwords would not need much time to crack most user-created passwords.

In some cases, the server does not require the credentials, but checks out the source IP address or subnet of the client to control the access. Especially when call trunking (for example, SIP trunking) is set up between a VoIP service provider and an enterprise customer, access control based on the source IP or subnet is commonly used. An attacker may be able to access the server by spoofing the source IP address.

Call Hijacking

Hijacking occurs when some transactions between a VoIP endpoint and the network are taken over by an attacker.

The transactions can be registration, call setup, media flow, and so on. This hijacking can make serious service interruption by disabling legitimate users to use the VoIP service. It is similar to call teardown in terms of stealing session information as a preliminary, but the actual form of attack and impact are different.

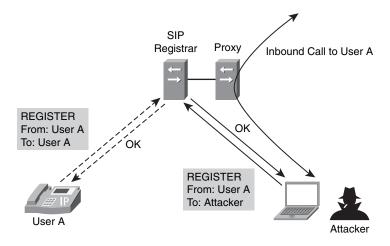
The typical cases are registration hijacking, media session hijacking, and server impersonating. The next few sections describe each of these cases.

Registration Hijacking

The registration process allows an endpoint to identify itself to the server (for example, SIP Registrar) as a device that a user is located.

An attacker monitors this transaction and sends spoofed messages to the server in order to hijack the session. When a legitimate user has been compromised, that user cannot receive inbound calls. Figure 2-3 illustrates the example with SIP messages.

Figure 2-3 Registration Hijacking



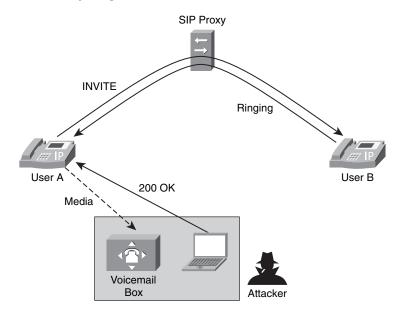
In Figure 2-3, an attacker impersonates a user agent by modifying the "From" header and adding the attacker's address to the "To" header when it sends a REGISTER message, which updates the address-of-record of the target user. All inbound calls to User A will be routed to the attacker.

This threat happens when the user agent server (Registrar) is relying on only SIP headers to identify the user agent.

Media Session Hijacking

When a media session is being negotiated between VoIP endpoints, an attacker may send spoofed messages to either one of them to redirect the media to another endpoint such as the attacker's phone or voicemail box. The victim will only be able to talk with the attacker's endpoint. Figure 2-4 illustrates the example with SIP messages.

Figure 2-4 Media Session Hijacking



In Figure 2-4, User A tries to make a call to User B and the IP phone of User B is ringing. Having monitored call requests to User B, an attacker detects the call and sends 200 OK messages to User A with the IP/port address of the attacker's voicemail server. User A leaves a voice message for User B in the attacker's voicemail box. This hijacking happens before the media session is established between User A and (the intended) user B.

Even after the media session is established between A and B, an attacker can still hijack an active session by sending a Re-Invite message to User A.

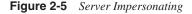
Server Impersonating

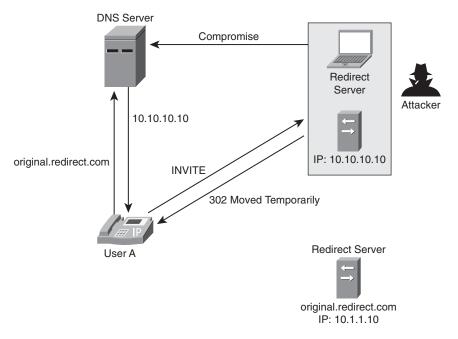
A VoIP client sends a request message to a server in the target domain for registration, call setup or routing, and so on. It is possible for an attacker to impersonate the server, receive the request message, and then manipulate it for malicious purposes.

The typical method of impersonating a server is attacking the local TFTP server or Domain Name Service (DNS) server as the initial step. An attacker may intrude into the TFTP server and replace the configuration file for IP phones with his file having an IP address of a malicious server (for example, SIP Registrar).

The IP phones downloading the malicious file will send a request message to the wrong server.

An attacker may also compromise the DNS server and replace the entry of current VoIP server with an IP address of a malicious server. The IP phones looking up the server IP will receive a wrong one. Figure 2-5 illustrates an example based on SIP transactions with a Redirect server.





In Figure 2-5, the attacker compromised the local DNS server first by replacing the IP address (10.1.1.10) of original.redirect.com with 10.10.10.10, which is the attacker's redirect server.

When User A tries to make a call to User B, the IP phone looks up the IP address of the redirect server (original.redirect.com) and receives the IP (10.10.10.10) of the impersonated server. The INVITE message is sent to the impersonated server, and it replies "302 Moved Temporarily" with wrong contact information that could be a dummy address or attacker's proxy server for further threat. The original redirect server (10.1.1.10) cannot receive any call request in this situation.

QoS Abuse

The elements of a media session are negotiated between VoIP endpoints during call setup time, such as media type, coder-decoder (codec) bit rate, and payload type. For example, it may be necessary or desirable to use G.729 when leaving a network (to conserve bandwidth)

but to use G.711 when calls are staying inside a network (to keep call quality higher). An attacker may intervene in this negotiation and abuse the Quality of Service (QoS), by replacing, deleting, or modifying codecs or payload type.

Another method of QoS abuse is exhausting the limited bandwidth with a malicious tool so that legitimate users cannot use bandwidth for their service. Some VoIP service providers or hosting companies limit the bandwidth for certain groups of hosts to protect the network. An attacker may know the rate limit and generate excessive media traffic through the channel, so voice quality between users may be degraded.

In this section so far, you have learned about threats against availability, such as call flooding, malformed messages, spoofed messages (call teardown, toll fraud), call hijacking (registration and media session hijacking, server impersonating), and QoS abuse. The next section covers another type of threat: attacks against call data and media confidentiality.

Threats Against Confidentiality

Another category of VoIP threat is the threat against confidentiality.

Unlike the service interruptions in the previous section, threats against confidentiality do not impact current communications generally, but provide an unauthorized means of capturing media, identities, patterns, and credentials that are used for subsequent unauthorized connections or other deceptive practices.

VoIP transactions are mostly exposed to the confidentiality threat because most VoIP service does not provide full confidentiality (both signal and media) end-to-end. In fact, full encryption of message headers is not possible because intermediary servers (for example, SIP proxy server) have to look at the headers to route the call. In some cases, the servers have to insert some information into the header (for example, Via header in SIP) as the protocol is designed.

This section introduces the most popular types of confidentiality threats: eavesdropping media, call pattern tracking, data mining, and reconstruction.

Eavesdropping Media

Eavesdropping on someone's conversation has been a popular threat since telecommunication service started a long time ago, even though the methods of eavesdropping are different between legacy phone systems and VoIP systems.

In VoIP, an attacker uses two methods typically. One is sniffing media packets in the same broadcasting domain as a target user's, or on the same path as the media. The other is compromising an access device (for example, Layer 2 switch) and forwarding (duplicating) the target media to an attacker's device.

The media can be voice-only or integrated with video, text, fax, or image. Figure 2-6 illustrates these cases.

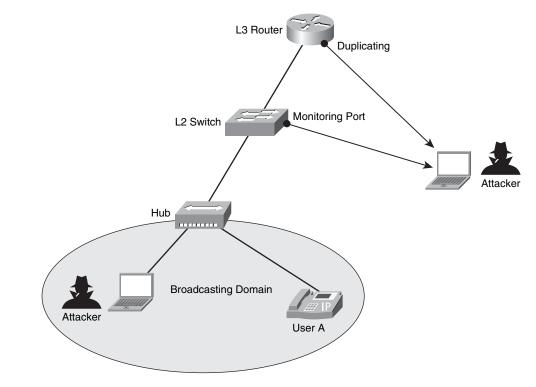


Figure 2-6 Eavesdropping Media

In Figure 2-6, the attacker's device that is in the same broadcasting domain as the IP phone of User A can capture all signals and media through the hub. This figure also shows the possibility that the attacker intrudes in a switch or router, and configures a monitoring port for voice VLAN, and forwards (duplicates) the media to the attacker's capturing device.

Another possible way of eavesdropping media is that an attacker taps the same path as the media itself, which is similar to legacy tapping technique on PSTN. For example, the attacker has access to the T1 itself and physically splits the T1 into two signals.

Although this technique is targeting media, the next method (call pattern tracking) is targeting signal information.

Call Pattern Tracking

Call pattern tracking is the unauthorized analysis of VoIP traffic from or to any specific nodes or network so that an attacker may find a potential target device, access information (IP/port), protocol, or vulnerability of network. It could also be useful for traffic analysis—knowing who called who, and when. For example, knowing that a company's CEO and CFO have been calling the CEO and CFO of another company could indicate that an acquisition is under way. For another example, knowing that a CEO called her stockbroker immediately after meeting with someone with insider stock knowledge is useful. That is, this is useful for learning about people and information.

To show an example of unauthorized analysis, sample messages that an attacker may capture in the middle of a network are illustrated in Example 2-2. It shows simple SIP request (INVITE) and response (200 OK) messages, but an attacker can extract a great deal of information from them by analyzing the protocol (key fields are highlighted).

Example 2-2 Exposed Information from SIP Messages

```
INVITE sip:9252226543@192.168.10.10:5060 SIP/2.0
Via: SIP/2.0/UDP 10.10.10.10:5060;branch=z9hG4bK00002000005
From: Alice <sip:4085251111@10.10.10.10:5060>;tag=2345
To: Bob <sip:9252226543@192.168.10.10>
Call-Id: 9252226543-0001
CSeq: 1 INVITE
Contact: <sip:4085251111@10.10.10.10>
Expires: 1200
Max-Forwards: 70
Content-Type: application/sdp
Content-Length: 143
Session Description Protocol Version (v): = 0
Owner/Creator, Session Id (o): 2 2 2 IN IP4 10.10.10.10
Session Name (s): Session SDP
Connection Information (c): IN IP4 10.10.10.10
Media Description, name and address (m): audio 9876 RTP/AVP 0 8 18
Media Attribute (a): rtpmap:0 PCMU/8000
Media Attribute (a): rtpmap:8 PCMA/8000
Media Attribute (a): rtpmap:18 G729a/8000
------
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.10.10.10:5060;branch=z9hG4bK00002000005
From: Alice <sip:4085251111@10.10.10.10:5060>;tag=2345
To: Bob <sip:9252226543@192.168.10.10>;tag=4567
Call-Id: 9252226543-0001
CSeq: 1 INVITE
Contact: <sip:9252226543@172.26.10.10>
Content-Type: application/sdp
Content-Length: 131
```

Example 2-2 Exposed Information from SIP Messages (Continued)

```
Session Description Protocol Version (v): = 0
Owner/Creator, Session Id (o): 2 2 2 IN IP4 172.26.10.10
Session Name (s): Session SDP
Connection Information (c): IN IP4 172.26.10.10
Media Description, name and address (m): audio 20000 RTP/AVP 18
Media Attribute (a): rtpmap:18 G729a/8000
```

The following list shows sample information that the attacker may extract from Example 2-2:

- The IP address of the SIP proxy server is 192.168.10.10, and the listening port is 5060.
- They use User Datagram Protocol (UDP) packets for signaling without any encryption, such as Transport Layer Security (TLS) or Secure Multipurpose Internet Mail Extension (S/MIME).
- The proxy server does not require authentication for a call request.
- The caller (Alice), who has a phone number 4085251111, makes a call to Bob at 9252226543.
- The IP address of Alice's phone is 10.10.10.10 and a media gateway is 172.26.10.10 (supposing that the call goes to PSTN).
- The media gateway opens a UDP port, 20000, to receive Real-time Transport Protocol (RTP) stream from Alice's phone.
- The media gateway accepts only G.729a codec (Alice's phone offered G.711a, G.711u, and G.729a initially).

The information just presented can be used for future attacks, such as DoS attack on the proxy server or the media gateway.

Data Mining

Like email spammers who collect email addresses from various sources like web pages or address books, VoIP spammers also collect user information like phone numbers from intercepted messages, which is one example of data mining.

The general meaning of data mining in VoIP is the unauthorized collection of identifiers that could be user name, phone number, password, URL, email address, strings or any other identifiers that represent phones, server nodes, parties, or organizations on the network. In Example 2-2, you can see that kind of information from the messages.

An attacker utilizes the information for subsequent unauthorized connections such as:

- Toll fraud calls
- Spam calls (for example, voice, Instant Messaging [IM], presence spam)

- Service interruptions (for example, call flooding, call hijacking, and call teardown)
- Phishing (identity fraud; see the section "Threats Against Social Context" for more information)

With valid identities, attackers could have a better chance to interrupt service by sending many different types of malicious messages. Many servers reject all messages, except registration, unless the endpoint is registered.

Reconstruction

Reconstruction means any unauthorized reconstruction of voice, video, fax, text, or presence information after capturing the signals or media between parties. The reconstruction includes monitoring, recording, interpretation, recognition, and extraction of any type of communications without the consent of all parties. A few examples are as follows:

- Decode credentials encrypted by a particular protocol.
- Extract dual-tone multifrequency (DTMF) tones from recorded conversations.
- Extract fax images from converged communications (voice and fax).
- Interpret the mechanism of assigning session keys between parties.

These reconstructions do not affect current communications, but they are utilized for future attacks or other deceptive practices.

In this section so far, you have learned about threats against confidentiality such as eavesdropping media, call pattern tracking, data mining, and reconstruction. The next section covers another type of threats: breaking message and media integrity.

Threats Against Integrity

Another category of VoIP threat is the threat against integrity, which impacts current service severely in most cases.

The basic method of the integrity threat is altering messages (signals) or media after intercepting them in the middle of the network. That is, an attacker can see the entire signaling and media stream between endpoints as an intermediary. The alteration can consist of deleting, injecting, or replacing certain information in the VoIP message or media.

This section is divided into two types of threat at a high level:

- Threats against message integrity (message alteration)
- Threats against media integrity (media alteration)

The next section describes and gives examples of each type of threat.

Message Alteration

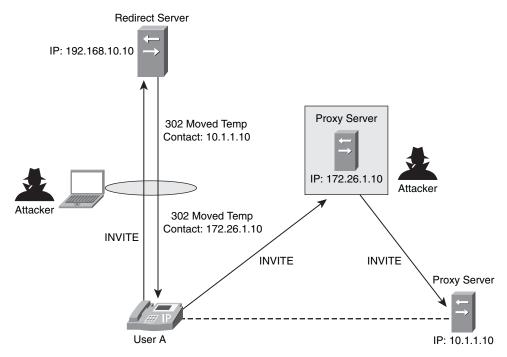
Message alteration is the threat that an attacker intercepts messages in the middle of communication entities and alters certain information to reroute the call, change information, interrupt the service, and so on. The typical examples are call rerouting and black holing.

Call Rerouting

Call rerouting is any unauthorized change of call direction by altering the routing information in the protocol message. The result of call rerouting is either to exclude legitimate entities or to include illegitimate entities in the path of call signal or media.

Figure 2-7 illustrates the example of including a malicious entity during call setup.

Figure 2-7 Call Rerouting



In Figure 2-7, an attacker keeps monitoring the call request message (for example, SIP INVITE) from User A to a redirect server. When User A initiates a call, the IP phone sends an INVITE message to the redirect server, as shown in Example 2-3.

Example 2-3 IP Phone Sends an INVITE Message to the Redirect Server

```
INVITE sip:Bob@192.168.10.10:5060 SIP/2.0
Via: SIP/2.0/UDP 10.10.10.10:5060;branch=z9hG4bK00002000005
From: UserA <sip:UserA@10.10.10.10:5060>;tag=2345
To: Bob <sip:Bob@192.168.10.10>
Call-Id: 9252226543-0001
CSeq: 1 INVITE
Contact: <sip:UserA@10.10.10.10>
Max-Forwards: 70
Content-Length: 0
```

The attacker detects the INVITE and intercepts the response message (that is, "302 Moved Temporarily") from the redirect server, as shown in the continuation of Example 2-3.

```
SIP/2.0 302 Moved Temporarily
From: UserA <sip:UserA@10.10.10.10:5060>;tag=2345
To: Bob <sip:Bob@192.168.10.10>;tag=6789
Call-Id: 9252226543-0001
CSeq: 1 INVITE
Contact: <sip:Bob@10.1.1.10>
Content-Length: 0
```

The attacker replaces the IP address of the proxy server (10.1.1.10) in the Contact header with his proxy server (172.26.1.10), and sends to the IP phone, as shown in the continuation of Example 2-3.

```
SIP/2.0 302 Moved Temporarily
From: UserA <sip:UserA@10.10.10.10:5060>;tag=2345
To: Bob <sip:Bob@192.168.10.10>;tag=6789
Call-Id: 9252226543-0001
CSeq: 1 INVITE
Contact: <sip:Bob@172.26.1.10>
Content-Length: 0
```

The IP phone sends a new INVITE to attacker's proxy server rather than the legitimate server, and his server relays the message as shown in the picture. From now on, the attacker in the middle can see all signals between the endpoints and modify for any malicious purpose.

Call Black Holing

Call black holing is any unauthorized method of deleting or refusing to pass any essential elements of protocol messages, in the middle of communication entities. The consequence of call black holing is to delay call setup, refuse subsequent messages, make errors on applications, drop call connections, and so on. Here are a few examples with SIP:

1 An attacker as an intermediary drops only ACK messages between call entities so that the SIP dialog cannot be completed, even though there could be early media between them.

- **2** An attacker as an intermediary deletes media session information (SDP) in the INVITE message, which could result in one-way audio or call disconnection.
- **3** An attacker as an intermediary refuses to pass all messages to a specific user (victim) so that the user cannot receive any inbound calls.

The call rerouting and black holing belong to message alteration as previously described. The next section covers media alteration as part of the threat against integrity.

Media Alteration

Media alteration is the threat that an attacker intercepts media in the middle of communication entities and alters media information to inject unauthorized media, degrade the QoS, delete certain information, and so on. The media can be voice-only or integrated with video, text, fax, or image. The typical examples are media injection and degrading.

Media Injection

Media injection is an unauthorized method in which an attacker injects new media into an active media channel or replaces media in an active media channel. The consequence of media injection is that the end user (victim) may hear advertisement, noise, or silence in the middle of conversation. Figure 2-8 illustrates the example with voice stream.

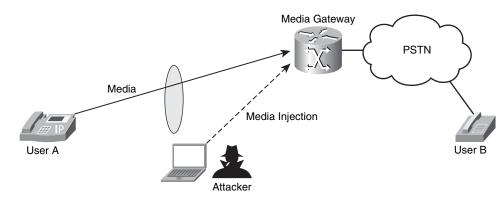


Figure 2-8 Media Injection

In Figure 2-8, User A with an IP phone makes a call to User B who has a PSTN phone through a media gateway. After the call setup, the IP phone sends voice (RTP) packets to the media gateway. An attacker in the middle monitors the RTP sequence number of the voice packets, and adjusts the sequence number of illegitimate packets (for example, advertisements), and injects them into the voice channel so that they will arrive before the legitimate packets. User B in PSTN hears the injected voice.

Media Degrading

Media degrading is an unauthorized method in which an attacker manipulates media or media control (for example, Real-Time Control Protocol [RTCP]) packets and reduces the QoS of any communication. Here are a couple of examples:

- 1 An attacker intercepts RTCP packets in the middle, and changes (or erases) the statistic values of media traffic (packet loss, delay, and jitter) so that the endpoint devices may not control the media properly.
- 2 An attacker intercepts RTCP packets in the middle, and changes the sequence number of the packets so that the endpoint device may play the media with wrong sequence, which degrades the quality.

In this section so far, you have learned about VoIP threats against integrity such as message alteration (call rerouting, call black holing) and media alteration (media injection, media degrading). The next section covers another type of threats: social threats.

Threats Against Social Context

A threat against social context (as known as "social threat") is somewhat different from other technical threats against availability, confidentiality, or integrity, as previously discussed, in terms of the intention and methodology. It focuses on how to manipulate the social context between communication parties so that an attacker can misrepresent himself as a trusted entity and convey false information to the target user (victim).

The typical threats against social context are as follows:

- Misrepresentation of identity, authority, rights, and content
- Spam of call (voice), IM, and presence
- Phishing

NOTE A call with misrepresentation is initiated by an attacker who is a communication entity, which is different from the threats in the "Threats Against Integrity" section, which are based on interception and then modification.

The general meaning of spam is unsolicited bulk email that you may see every day. It wastes network bandwidth and system resources, as well as annoying email users. The spam exists in VoIP space as well, so-called VoIP spam, in the form of voice, IM, and presence spam. This section looks into each type of VoIP spam with SIP protocol. The content refers to RFC 5039.¹

Phishing is becoming popular in the VoIP world these days as a method of getting somebody's personal information by deceiving the identity of an attacker.

The following sections give more details about these social threats.

NOTE These same types of attacks are equally available in today's PSTN environment.

Misrepresentation

Misrepresentation is the intentional presentation of a false identity, authority, rights, or content as if it were true so that the target user (victim) or system may be deceived by the false information. These misrepresentations are common elements of a multistage attack, such as phishing.

Identity misrepresentation is the typical threat that an attacker presents his identity with false information, such as false caller name, number, domain, organization, email address, or presence information.

Authority or rights misrepresentation is the method of presenting false information to an authentication system to obtain the access permit, or bypassing an authentication system by inserting the appearance of authentication when there was none. It includes presentation of password, key, certificate, and so on. The consequence of this threat could be improper access to toll calls, toll calling features, call logs, configuration files, presence information of others, and so on.

Content misrepresentation is the method of presenting false content as if it came from a trusted source of origin. It includes false impersonation of voice, video, text, or image of a caller.

Call Spam (SPIT)

Call (or voice) spam is defined as a bulk unsolicited set of session initiation attempts (for example, INVITE requests), attempting to establish a voice or video communications session. If the user should answer, the spammer proceeds to relay their message over real-time media. This is the classic telemarketer spam, applied to VoIP, such as SIP. This is often called SPam over IP Telephony, or SPIT.

The main reason SPIT is becoming popular is that it is cost-effective for spammers. As you know, legacy PSTN-call spam already exists in the form of telemarketer calls. Although these calls are annoying, they do not arrive in the same kind of volume as email spam. The difference is cost; it costs more for the spammer to make a phone call than it does to send email. This cost manifests itself in terms of the cost for systems that can perform telemarketer calls, and in cost per call. However, the cost is dramatically dropped when switching to

SPIT for many reasons: low hardware cost, low line cost, ease of writing a spam application, no boundary for international calls, and so on. Additionally, in some countries, such telemarketing calls over the PSTN are regulated.

In some cases, spammers utilize computational and bandwidth resources provided by others, by infecting their machines with viruses that turn them into "zombies" that can be used to generate call spam.

Another reason SPIT is getting popular is its effectiveness, compared to email spams. For email spams, you may already realize that there is a big difference between turning on and off a spam filter for your email account. In fact, most spam filters for email today work very well (filter more than 90 percent of spams) because of the nature of email; store and forward. All emails can be stored and examined in one place before forwarding to users. Even though users may still receive a small percentage of email spams, they usually look at profiles (for example, sender name and subject) and delete most of them without seeing the contents. However, the method of filtering emails does not work for SPIT because voice is real-time media. Only after listening to some information initially can users recognize whether it is a spam or not. So, spammers try to put main information in the initial announcement so that users may listen to it before hanging up the phone. There is a way to block those call attempts based on a blacklist (spammers' IP address or caller ID), but it is useless if spammers spoof the source information.

You can find more information on SPIT and mitigation methods in Chapter 6, "Analysis and Simulation of Current Threats."

The next topic is a different type of VoIP spam, IM spam.

IM Spam (SPIM)

IM spam is similar to email. It is defined as a bulk unsolicited set of instant messages, whose content contains the message that the spammer is seeking to convey. This is often called Spam over Instant Messaging, or SPIM.

SPIM is usually sent in the form of request messages that cause content to automatically appear on the user's display. The typical request messages in SIP are as follows:

- SIP MESSAGE request (most common)
- INVITE request with large Subject headers (since the Subject is sometimes rendered to the user)
- INVITE request with text or HTML bodies

Example 2-4 shows examples with SIP INVITE and MESSAGE.

Example 2-4 IM Spam

```
INVITE sip:Bob1@192.168.10.10:5060 SIP/2.0
Via: SIP/2.0/UDP 10.10.10.10:5060;branch=z9hG4bK00002000005
From: Spammer <sip:spammer1@10.10.10.10:5060>;tag=2345
To: Bob <sip:Bob1@192.168.10.10>
Call-Id: 9252226543-0001
CSeq: 1 INVITE
Subject: Hi there, buy a cool stuff in our website www.spam-example.com
Contact: <sip:spammer1@10.10.10.10>
Expires: 1200
Max-Forwards: 70
Content-Type: application/sdp
Content-Length: 143
MESSAGE sip:Bob1@192.168.10.10:5060 SIP/2.0
Via: SIP/2.0/UDP 10.10.10.10:5060;branch=z9hG4bK00002000005
From: Spammer <sip:spammer1@10.10.10.10:5060>;tag=2345
To: Bob <sip:Bob1@192.168.10.10>
Call-Id: 9252226543-0001
CSeq: 1 MESSAGE
Max-Forwards: 70
Content-Type: test/plain
Content-Length: 25
Hi there, buy a cool stuff in our website www.spam-example.com
```

SPIM is very much like email, but much more intrusive than email. In today's systems, IMs automatically pop up and present themselves to the user. Email, of course, must be deliberately selected and displayed.

Presence Spam (SPPP)

Presence spam is similar to SPIM. It is defined as a bulk unsolicited set of presence requests (for example, SIP SUBSCRIBE requests) in an attempt to get on the "buddy list" or "white list" of a user to subsequently send them IM or INVITEs. This is occasionally called SPam over Presence Protocol, or SPPP.

The cost of SPPP is within a small constant factor of IM spam, so the same cost estimates can be used here. What would be the effect of such spam? Most presence systems provide some kind of consent framework. A watcher that has not been granted permission to see the user's presence will not gain access to their presence. However, the presence request is usually noted and conveyed to the user, allowing them to approve or deny the request. This request itself can be spam, as shown in Example 2-5.

In SIP, this is done using the watcherinfo event package. This package allows a user to learn the identity of the watcher, in order to make an authorization decision. This could provide a vehicle for conveying information to a user; Example 2-5 shows the example with SIP SUBSCRIBE.

Example 2-5 Presence Spam

```
SUBSCRIBE sip:bob@example.com SIP/2.0
Event: presence
To: sip:bob@example.com
From: sip:buy-cool-dvds-and-games@spam-example.com
Contact: sip:buy-cool-dvds-and-games@spam-example.com
Call-ID: knsd08alas9dy@3.4.5.6
CSeq: 1 SUBSCRIBE
Expires: 3600
Content-Length: 0
```

A spammer in Example 2-5 generates the SUBSCRIBE request from the identity (sip:buycool-dvds-and-games@spam-example.com), and this brief message can be conveyed to the user, even though the spammer does not have permission to access presence. As such, presence spam can be viewed as a form of IM spam, where the amount of content to be conveyed is limited. The limit is equal to the amount of information generated by the watcher that gets conveyed to the user through the permission system.

Phishing

The general meaning of *phishing* is an illegal attempt to obtain somebody's personal information (for example, ID, password, bank account number, credit card information) by posing as a trust entity in the communication. In VoIP, phishing is typically happening through voice or IM communication, and voice phishing is sometimes called "vishing."

The typical sequence is that a phisher picks target users and creates request messages (for example, SIP INVITE) with spoofed identities, pretending to be a trusted party. When the target user accepts the call request, either voice or IM, the phisher provides fake information (for example, bank policy announcement) and asks for personal information. Some information like user name and password may not be directly valuable to the phisher, but it may be used to access more information useful in identity theft.

Here are a couple of phishing examples:

1 A phisher makes a call to a target user and leaves a voice message like: "This is an important message from ABC Bank. Because our system has changed, you need to change your password. Please call back at this number: 1-800-123-4567." When the target user calls the number back, the phisher's Interactive Voice Response (IVR) system picks up the call and acquires the user's password by asking "Please enter your current password for validation purposes"

2 A phisher sends an instant text message to a smart phone (for example, PDA phone) or softphone (for example, Skype client) users, saying "This message is from ABC Bank. Your credit card rate has been increased. Please check it out on our website: http://www.abcbank.example.com." When the users click the URL, it goes to a phisher's website (example.com) that appears to have exactly the same web page that ABC Bank has. The fake website collects IDs and passwords that the users type in.

In this section, you have learned about VoIP threats in a social context, such as misrepresentation, call spamming, IM spamming, presence spamming and phishing. For more detailed information about VoIP spamming, refer to Chapter 6, "Analysis and Simulation of Current Threats."

Summary

VoIP vulnerabilities can be exploited to create many different kinds of threats. The threats can be categorized as four different types: threats against availability, confidentiality, integrity, and social context.

A threat against availability is a threat against service availability that is supposed to be running 24/7. That is, the threat is aiming at VoIP service interruption, typically, in the form of DoS. The examples are call flooding, malformed messages (protocol fuzzing), spoofed messages (call teardown, toll fraud), call hijacking (registration or media session hijacking), server impersonating, and QoS abuse.

A threat against confidentiality does not impact current communications generally, but provides an unauthorized means of capturing conversations, identities, patterns, and credentials that are used for the subsequent unauthorized connections or other deceptive practices. VoIP transactions are mostly exposed to the confidentiality threat because most VoIP service does not provide full confidentiality (both signal and media) end-to-end. The threat examples are eavesdropping media, call pattern tracking, data mining, and reconstruction.

A threat against integrity is altering messages (signals) or media after intercepting them in the middle of the network. That is, an attacker can see the entire signaling and media stream between endpoints as an intermediary. The alteration can consist of deleting, injecting, or replacing certain information in the VoIP message or media. The typical examples are call rerouting, call black holing, media injection, and media degrading.

A threat against social context focuses on how to manipulate the social context between communication parties so that an attacker can misrepresent himself as a trusted entity and convey false information to the target user. The typical examples are misrepresentation (identity, authority, rights, and content), voice spam, instant message spam, presence spam, and phishing.

End Notes

1 RFC 5039, "SIP and Spam," J. Rosenberg, C. Jennings, http://www.ietf.org/ rfc/rfc5039.txt, January 2008.

References

"Phishing," Wikipedia, http://en.wikipedia.org/wiki/Phishing.

RFC 3261, "SIP (Session Initiation Protocol)," J. Rosenberg, H. Schulzrinne, G. Camarillo, A. Johnston, J. Peterson, R. Sparks, M. Handley, and E. Schooler, June 2002.

RFC 3428, "Session Initiation Protocol (SIP) Extension for Instant Messaging," B. Campbell, J. Rosenberg, H. Schulzrinne, C. Huitema, D. Gurle, December 2002.

Trammell, Dustin D. "VoIP Attacks," http://www.dustintrammell.com/presentations/.

"VoIP Security Threat Taxonomy," VOIPSA, http://www.voipsa.org/Activities/ taxonomy-wiki.php.

NDEX

Numerics

3DES (Triple Data Encryption Standard), 87

A

access control, Unified CME, 259-261 devices, 13, 277 deployment, 284-286 IP phones, 278 Switch, 278-282 VLAN ACLs, 282-284 policies, 213 ports, preventing, 279 SBCs, 208 Access Control Engines. See ACEs Access Control Lists. See ACLs Access Function (AF), 116, 295 access gateways, 74 **ACEs (Access Control Engines)**, 216 ACF (admission confirm), 50 ACLs (Access Control Lists), 108, 215 DoS protection, 216 VLANs, 282-284 Active-Active mode, 231 Active-Standby mode, 230-231, 250 Address Resolution Protocol. See ARP addresses alias address modification, 50 call content interception, 301-302 limited-use, 171 NAT, 21, 109-113 obfuscation, 170 translation, 49 traversal, 222-224 address-of-record (AoR), 69 AddRoundKey() function, 92 admission confirm (ACF), 50 admissions, control, 49 AES (Advanced DES), 89-92 AF (Access Function), 116, 295 aggregation routers, 327–328 ALG (Application Layer Gateway), 253 algorithms DES, 87. See also DES DSA, 95-96 hashing, 96 MAC, 99-100 MD5. 97-98 SHS, 98-99 **RSA**. 95 SHA. 84 alias address modification, 50 Alliance for Telecommunications Industry Solutions (ATIS), 292 alteration media, 37-38 messages, 35-37 amplification, DoS and, 197 Analog Telephone Adapter (ATA), 117 analysis, 160 flooding attacks, 135-137 malformed messages, 150-153 service policies, 234-237 sniffing/eavesdropping, 158-161 spoofing, 164-165 unintentional flooding, 139 anchoring media, 240 **ANMPv3** (Simple Network Management Protocol version 3), 316 Annex D (H.235) baseline security, 54 Annex E (H.235) signature security, 55-56 Annex F (H.235) hybrid security, 56-57 Answer messages, 334 AoR (address-of-record), 69 Application Layer Gateway (ALG), 253 applications pkcs7-mime types, 183 VoIP, 12 architecture Cisco SII architecture, 313, 329 connectivity, 232-234 hardware, DoS protection, 215-216 LI, 294-297 networks, 8 SBC locations, 224

services, 5 Active-Active, 231 Active-Standby, 230-231 high availability, 229-230 network connectivity, 232-234 policy analysis, 234–237 SBCs, 228-229 traffic flow optimization, 239-244 virtualization, 237-238 area codes, 8 **ARP (Address Resolution Protocol)** firewalls, 252 misbehaving endpoints, 140-141 responses, limiting, 282 ASA firewalls, 251–256 asymmetric (public) key cryptography, 92-93 DSA, 95-96 RSA, 93-95 ATA (Analog Telephone Adapter), 117 **ATIS (Alliance for Telecommunications Industry** Solutions), 292 attacks DoS. 128 intentional flooding, 129-138 unintentional flooding, 138-143 **MITM. 198** replay, 326 TCP SYN, 128 attributes, service policies, 234 audio, saving, 161 AuditConnection (AUCX), 75 AUEP (AuditEndpoint), 75 authentication, 14, 175 CHAP, 119 devices, 270 digest, 68-69, 198, 271-272 files, 270 identity, 69-70 images, 270 integrity, 53 MAC, 99-100 PAP, 119 signaling, 271 Unified CM, 269-273 user-to-proxy, 176-179 user-to-user, 179-182 white lists, 214

authentication-only security services, 53 authority, 39 authorization, 272 calls, 49 white lists, 214 availability, threats against, 20–30

В

B2BUA (Back-to-Back User Agent), 59, 118 bandwidth control, 49, 109 management, 49 transcoding, 226-227 baseline security (H.235), 54 benefits of VoIP, 6-8 binding information, 282 black holing, calls, 36 black traffic, 213 black/white lists, consent-based, 171 blind transfers, 336 blocking after-hours calls, 266-267 bodies, formatting S/MIME, 186-188 borders SBCs between access and core networks, 206 - 207between core and peer networks, 207 troubleshooting, 204-206 sessions, 203 BTS (Cisco Broadband Telephony Softswitch), 302

С

CA (Certificate Authority), 95 CALEA (Communications Assistance for Law Enforcement Act), 292 call content (CC), 292 call content channel (CCC), 296 call content connection identifier (CCCID), 316 call data (CD), 292 call data channel (CDC), 296 calls after-hours call blocking, 266–267 authorization, 49

black holing, 36 content connection interfaces, 329-339 interception, 301-302 control flooding, 21, 129 signaling, 49 flow, 20-22 digest authentication, 176 H.323, 50-52 through SBCs, 210 SIP. 60-61 forwarding, 303 hijacking, 26 pattern tracking, 32-33 processing servers, 117-120 units, 216 requests, flooding, 129 rerouting, 35 spam, 39 teardown, 25 toll fraud, 26 transfers, 336 CAM (content-addressable memory), 278-279 capacity, LI, 304 **CAPF certificates**, 268 CC (call content), 292 CCC (call content channel), 296 CCChange message, 334 CCCID (call content connection identifier), 316 CCClose message, 334 CCOpen message, 334 CD (call data), 292 CDC (call data channel), 296 Certificate Authority (CA), 95 certificates, 269-277 S/MIME, 184 Unified CM, 267-269 CFs (Collection Functions), 116, 296, 315 **CHAP** (Challenge Handshake Authentication Protocol), 119 **Cisco Broadband Telephony Softswitch (BTS)** 10200, 328 Cisco Lawful Intercept Control MIB (CISCO-**TAP-MIB**), 327 Cisco PSTN Gateway 2200 (PGW 2200), 328

Cisco Unified Communications Manager Express. See Unified CME **Cisco Unified Communications Manager.** See Unified CM **Cisco Unity SCCP device certificates, 269 Cisco Unity server certificates**, 269 Class of Restriction (COR), 264-266 click-to-call feature, 8 clients UAC. 58, 175 web. 13 CMS (cryptographic message syntax), 183 Collection Functions (CFs), 116, 296, 315 commands enable password, 259 enable secret, 259 ip http access-class, 263 ip http authentication, 264 ip http secure-server, 263 make, 162 make no-opensssl, 162 rtpw, 192 SIPcrack, 162 communications, real-time, 11 components H.323, 49-50 SIP, 58-59 vulnerabilities, 12-13 ConferencePartyChange message, 334 confidentiality breaches, 326 threats against, 30-34 configuration access devices, 284-286 firewalls ASA and PIX. 254-256 FWSM, 257-258 LI, 325-327 SBCs Active-Active, 231 Active-Standby, 230-231 high availability, 229-230 network connectivity, 232-234 service architecture design, 228–229 service policy analysis, 234–237 traffic flow optimization, 239-244 virtualization, 237-238

security VoIP-aware firewalls, 108–109 SIP sessions, 61-67 troubleshooting, 22 Unified CM, 275-277 wrong configuration of devices, 139 consent-based black/white lists, 171 consultative transfers, 336 content connection interfaces, 329-339 encryption, LI, 302 filtering, 168 misrepresentation, 39 control access, Unified CME, 259-261 admissions, 49 bandwidth, 49, 109 media, 240 MGCP, 74-75 phone registrations, Unified CME, 261-262 ping, 220 planes, 314 registration timers, 217-220 security profiles, 75-77 transcoding, 226-227 conversion, protocols, 226 COR (Class of Restriction), 264–266 cost savings, 6-7 CPE (Customer Premise Equipment), 116–117 SBCs, 209 cracking passwords, 163 **CRCX** (CreateConnection), 75 CreateConnection (CRCX), 75 cryptanalysis, 83 cryptographic message syntax (CMS), 183 cryptography, 83 asymmetric (public) key, 92-93 DSA, 95-96 RSA, 93-95 symmetric (private) key, 84-85 3DES, 87 AES, 89-92 DES. 85-87 Customer Premise Equipment (CPE), 116-117

D

DAI (Dynamic ARP Inspection), 282 Data Encryption Standard. See DES data integration, 12 data mining, 33-34 data security infrastructure, 15 DDoS (Distributed Denial-of-Service), 197 debugging firewalls, 252 decryption, 189 degrading media, 38 **DeleteConnection (DLCX)**, 75 Delivery Function (DF), 116, 295-296 demarcation points, 339 Denial-of-Service. See DoS attacks deployment access devices, 284-286 firewalls, 250 location, 239-240 derivation, keys, 188-190 DES (Data Encryption Standard), 54, 85-87 design configuration. See configuration service architecture, 228-229 Active-Active, 231 Active-Standby, 230-231 high availability, 229-230 network connectivity, 232-234 service policy analysis, 234-237 traffic flow optimization, 239-244 virtualization, 237-238 detection by target subscribers, LI, 300-301 unauthorized creation of intercepts, 303 devices access, 13, 277 deployment, 284-286 IP phones, 278 Switch, 278-282 VLAN ACLs, 282-284 authentication, 270 configuration, 22 endpoint, 140 interfaces, Cisco SII, 314

lack of, 11 MD, 298, 339-341 security, 108 LI, 114-116 NAT, 109–113 SBC, 113–114 VoIP-aware firewalls, 108-109 services, 116 call processing servers, 117–120 CPE, 116-117 wrong configuration of, 139 DF (Delivery Function), 116, 295-296 DHCP (Dynamic Host Configuration Protocol) DoS attacks, mitigating, 281 Servers, preventing fraudulent, 280 dialed digital translation, 50 DialedDigitExtraction message, 334 dictionaries, passwords, 26 **Differentiated Services Code Point (DSCP), 228** Diffie-Hellman key-exchange procedures, 54 digest authentication, 68-69, 271-272 limitations of, 198 Digital Signature Algorithm. See DSA digital signatures, 95 disadvantages of VoIP, 8-10 Distributed Denial-of-Service (DDoS), 197 distribution, keys, 101–103 **DLCX (DeleteConnection)**, 75 DNS (Domain Name Service), 316 domains. 113 SBCs, 203 DoS (Denial-of-Service) attacks, 128 and amplification, 197 DHCP, mitigating, 281 flooding, 114 intentional flooding, 129-138 protection, 109 SBCs, 206, 213-216 unintentional flooding, 138-143 DSA (Digital Signature Algorithm), 95–96 **DSCP** (Differentiated Services Code Point), 228 DTMF (dual-tone multifrequency), 34, 273 **Dynamic ARP Inspection (DAI), 282 Dynamic Host Configuration Protocol.** See DHCP dynamic triggering, 308

Ε

eavesdropping, 154 analysis, 158-161 media, 30-31 mitigation, 161 simulation, 154-158 Edge Services Router (ESR), 327 **EIGRP** (Enhanced Interior Gateway Routing), 252 email, spam, 165 mitigation, 168-172 SPPP, 167-168 voice. 165-166 emergency calls, 9 **Enable Application Level Authorization check** box, 273, 277 Enable Digest Authentication check box, 276 enable password command, 259 enable secret command, 259 encryption, 14, 182 content, LI, 302 media, 188-193, 274 S/MIME, 183-188 signaling, 273 Unified CM, 273-275 encryption key (KE), 53 EndpointConfiguration (EPCF), 75 endpoints devices, 140 misbehaving, 140-141 troubleshooting, 22 Enhanced Interior Gateway Routing (EIGRP), 252 EOFB (Enhanced OFB), 53 **EPCF** (EndpointConfiguration), 75 ESR (Edge Services Router), 327 **ETSI** (European Telecommunications Standards Institute), 292 **European Council Resolution**, 292 exchanges, keys, 185-186 expiration time, registration timer control, 217-220 exposed interfaces, 11 extensions, preventing network, 280

F

fake (spoofed) messages, 24-30 features, rich, 8 file authentication, 270 filtering content, 168 SIP messages, 158 Find-Me-Follow-Me (FMFM), 8 firewalls, 249-251 ASA and PIX, 251-256 deployment, 250 FWSM, 256-258 limitations, 258-259 SBCs, 207 VoIP-aware, 108–109 First Available, 221 flooding calls, 20-22 control. 129 request, 129 DoS attacks, 114, 128 intentional, 129–138 unintentional, 138-143 MAC CAM, mitigating, 278-279 messages, 132, 137 ping, 129 registration, 129 SBCs, 206, 213-216 traffic, 132 flow calls digest authentication, 176 through SBCs, 210 SIP. 60-61 process. See process flow traffic, 239-244. See also traffic FMFM (Find-Me-Follow-Me), 8 formatting S/MIME bodies, 186–188 forwarding calls, 303 FQDN (fully qualified domain name), 316 fraud COR. 264-266 DHCP servers, preventing, 280 toll, 26

full cone NAT, 109 fully qualified domain name (FQDN), 316 functionality, SBCs, 208, 226-228 DoS protection, 213-216 LI, 224-225 NAT traversal, 222-224 network topology hiding, 208-212 overload protection, 216–222 functions. See also commands AddRoundKey(), 92 hash, 97-98 MixColumns(), 91 ShiftRows(), 90 SubBytes(), 89 fuzzing, protocol, 22-24 FWSM firewalls, 256–258

G

gatekeepers, 50 gateways, 50 ALG, 253 **EIGRP. 252** media, 119 MGCP, 74-77 trunking, 328 Unified CM. 267 authentication, 269-273 configuration, 275–277 encryption, 273–275 integrity, 269-273 security, 267-269 Unified CME, 259 access control. 259-261 after-hours call blocking, 266-267 COR, 264-266 phone registration control, 261–262 secure GUI management, 263-264 geographical limitations, 7 Gigabit Switch Router (GSR), 327 global power outages, 139 government law, LI. See LI gray traffic, 214 GSR (Gigabit Switch Router), 327

Н

H.225 (Q.931), 48 H.235.48 Annex D (baseline security), 54 Annex E (signature security), 55-56 Annex F (hybrid security), 56-57 H.245.48 H.323, 48 call flow, 50-52 components, 49-50 overview of, 48-52 security profiles, 52-57 hacking, 83. See also cryptography hairpinning, 328 handshakes, TLS, 71-73 hardware, DoS protection, 215-216 Hash-based Message Authentication Code (HMAC), 303 hashing algorithms, 96 MAC, 99-100 MD5, 97-98 SHS, 98-99 headers Cisco SII architecture, 313-329 SIP P-DCS, 309-313 heavy traffic, comparing to malicious flooding, 136 hiding topologies, 208-212 high availability, 229-230 network connectivity with, 233 hijacking calls, 26 media sessions, 27 registration, SIP, 195 HMAC (Hash-based Message Authentication Code), 303 **HTTPS certificates**, 268 hybrid security (H.235), 56-57

IAD (integrated access device), 117, 139 IAP (intercept access point), 295, 308 identity authentication, 69–70

theft, 162 analysis, 164-165 mitigation, 165 simulation, 162–164 **IETF (Internet Engineering Task Force)**, 183 IM (Instant Messaging), 5 spam, 167 SPIM, 40-41 images, authentication, 270 impersonation, 326 servers, 28-29, 196 implementation of LI intercept request interfaces, 308-329 inbound calls, 112. See also calls infrastructure data security, 15 IP, 10 sources of vulnerability, 10-13 injection, media, 37 inspection, protocol messages, 108 Instant Messaging. See IM integrated access device (IAD), 117, 139 integration data, 12 voice, 12 integrity authentication, 53 threats against, 34-38 Unified CM, 269-273 intentional flooding, 129-138 SBCs. 206 Interactive Voice Response (IVR), 42 intercept access point (IAP), 295, 308 intercept request interfaces, 308 Cisco SII, 313-329 SIP P-DCS headers, 309-313 interception call content, 301-302 capacity, 304 unauthorized creation and detection of intercepts, 303 interfaces content connection, 329-339 devices, Cisco SII, 314 exposed, 11

intercept request, 308 Cisco SII, 313-329 SIP P-DCS headers, 309-313 link-layer, 340 management, 12 Unified CME, 263-264 network layer, 340 physical, 340 request/response, LI, 297-300 user control, 7 International Telecommunication Union. See ITU Internet Engineering Task Force (IETF), 183 Internet service providers (ISPs), 204 interoperability, 8 interruption threats against availability, 20-30 against confidentiality, 30-34 against integrity, 34-38 against social context, 38-43 spoofed messages, 24-30 invalid call request flooding, 129 invalid registration flooding, 129 **INVITE message**, 112 **IP** (Internet Protocol) network infrastructure, 10 ip http access-class command, 263 ip http authentication command, 264 ip http secure-server command, 263 IP phones, 117, 278 IP-based PBX, 118 IPSec, 73, 195, 268 ISPs (Internet service providers), 204 ITU (International Telecommunication Union), 48 IVR (Interactive Voice Response), 42

Κ

KE (encryption key), 53 keys asymmetric (public) cryptography, 92–93 DSA, 95–96 RSA, 93–95 distribution, 101–103 exchanges, S/MIME, 185–186 management, 100–101 SRTP, 188–190 symmetric (private) cryptography, 84–85 3DES, 87 AES, 89–92 DES, 85–87 KS (salting key), 53

L

LAES (Lawfully Authorized Electronic Surveillance), 309 Law Enforcement Administration Function (LEAF), 116, 297 law enforcement agencies. See LEAs lawful interception. See LI Lawfully Authorized Electronic Surveillance (LAES), 309 layers, security, 193-195 **LEAF** (Law Enforcement Administration Function), 116, 297 LEAs (law enforcement agencies), 292 impersonation, 326 MDs, 339-341 requirements for, 293-294 Least Busy, 222 legacy system security, 14 legal issues, 9 legitimate call flooding, 22 LI (lawful interception), 9, 114-116 Cisco SII, 313-329 configuration, 327 content connection interfaces, 329-339 intercept request interfaces, 308 MDs, 339-341 operational issues, 300-304 overview of, 292-293 reference models, 294-297 request/response interfaces, 297-300 requirements for LEAs, 293-294 SBCs, 207, 224-225 SIP P-DCS headers, 309–313 libSRTP tool, 191–193 limitations ARP responses, 282 digest authentication, 198 firewalls, 258-259 geographical, 7 S/MIME, 198

SIP, 198–199 TLS, 199 limited-use addresses, 171 link-layer interfaces, 340 Linux resource monitoring, 134 load balancing, 220–222 local hairpinning, 328 locations, deployment, 240

Μ

MAC (Media Access Control), 278-279 MAC (Message Authentication Code), 99-100 make command, 162 make no-openssl command, 162 malformed messages, 22-24, 143-144 analysis, 150-153 mitigation, 154 simulation, 144-150 malicious call teardown, 25 malicious flooding, comparing to heavy traffic, 136 management bandwidth. 49 firewalls, 252 interfaces, 12 Unified CME, 263-264 keys, 100-103 zones, 49 man-in-the-middle (MITM) attack, 198 marking QoS, 228 master keys, 188-190 MC (Multipoint Controller), 50 MCU (Multipoint Control Unit), 50 MD (mediation device), 308 MD5 (Message Digest 5), 97-98, 303 MDCX (ModifyConnection), 75 mdpasswd, 328 MDs (Mediation Devices), 339–341 mduserid, 328 media alteration, 37-38 anchoring, 240 control, 240 degrading, 38 eavesdropping, 30-31 encryption, 188-193, 274

gateways, 74, 119 injection, 37 rich servers, 119 rich service, 7 traversal, 241 Media Access Control. See MAC Media Gateway Control Protocol. See MGCP media session hijacking, 27 MediaReport message, 334 mediation device (MD), 298, 308, 339-341 Message Authentication Code (MAC), 99-100 Message Digest 5 (MD5), 97-98, 303 messages alteration, 35-37 CDC, 333-334 encryption, S/MIME, 183-188 flooding, 132, 137 forgery, 326 INVITE, 112 malformed, 22-24, 143-144 analysis. 150-153 mitigation, 154 simulation, 144–150 ping, control, 220 privacy, 94 protocol inspection, 108 SIP filtering, 158 INVITE. 23 NAT, 111 spoofed, 24-30 MGCP (Media Gateway Control Protocol), 74-75 security profiles, 75-77 mining data, 33-34 misbehaving endpoints, 140-141 misrepresentation, 39 mitigation DHCP DoS attacks, 281 flooding attacks, 137-138 MAC CAM flooding, 278-279 malformed messages, 154 sniffing/eavesdropping, 161 spam, 168-172 spoofing, 165 unintentional flooding, 141, 143 MITM (man-in-the-middle) attack, 198

MixColumns() function, 91 mobility, 11 services, 7 models reference, LI, 294-297 threats, 195-198 modes Active-Standby, 250 packets, 296 routed ASA and PIX firewalls, 251-252 FWSM firewalls, 256 SIP, 130 transparent ASA and PIX firewalls, 252 FWSM firewalls, 256 ModifyConnection (MDCX), 75 monitoring resources, 134 multihoming, 301 multimedia, SIP, 57-58 overview of, 58-67 security profiles, 67-73 multipart/signed content type, 183 Multipoint Control Unit (MCU), 50 Multipoint Controller (MC), 50 myths versus reality, 14-15

Ν

NAPT (Network Address and Port Translation), 110 NAT (Network Address Translation), 21, 109-113 SBCs, 207 traversal, 222-224 National Institute of Standards and Technology (NIST), 5 negative testing tools, 129 Network Address and Port Translation. See NAPT Network Address Translation. See NAT network interface card (NIC), 140, 216 network layer interfaces. 340 security, 193-195 networks architecture, 8

borders between access and core networks, 206 - 207between core and peer networks, 207 troubleshooting, 204-206 connectivity, 232-234 extensions, preventing, 280 IP, 10 PSTN. 5 topologies, hiding, 208-212 vulnerabilities, 13 NetworkSignal message, 334 NIC (network interface card), 140, 216 NIST (National Institute of Standards and Technology), 5 nonrepudiation, 55 NotificationRequest (RQNT), 75 NTFY (Notify), 75 numbers, translation, 227-228

0

obfuscation, addresses, 170 OFB (Output Feedback Mode), 53 open networks, 11 Open Shortest Path First (OSPF), 251 open VoIP protocols, 11 operating systems, 12 Origination message, 334 OSPF (Open Shortest Path First), 251 Output Feedback Mode (OFB), 53 overload protection, 216–222

Ρ

PacketCable, 292, 316, 328 packets modes, 296 RTCP, 38 SRTP, processing, 190–191 PAP (Password Authentication Protocol), 119 passwords cracking, 163 dictionaries, 26 patterns, tracking, 32–33 payload of CCC diagrams, 331 peer SBCs, 207-208 performance, call flooding, 20-22 personalized ring tones, 8 PGW 2200 (Cisco PSTN Gateway 2200), 328 phishing, 42-43 phone portability, 7 phone registration control, Unified CME, 261-262 physical interfaces, 340 ping control, 220 flooding, 21, 129 PIX firewalls, 251-256 policies access, 213 services, 234-237 Policy Engine, DoS protection, 216 port restricted cone NAT, 109 portability of phones, 7 port access, preventing, 279 power outages, 9 global, 139 regional, 22 predesign considerations for LI, 325 pre-provisioning, 308 presence spam (SPPM), 41-42, 167-168 prespoofing scans, 162 prevention fraudulent DHCP servers, 280 network extensions, 280 port access, 279 threats, 195-198 privacy breaches, 326 messages, 94 private (symmetric) key cryptography, 84-85 3DES, 87 AES, 89-92 DES, 85-87 process flow for conference calls, 322-324 for forwarding calls, 319-321 for inbound calls, 311 for outbound calls, 310 for standard calls, 316-319 processing packets, SRTP, 190–191

profiles, security, 52-57 MGCP, 75-77 SIP, 67-73 properties, SHA, 99 **Proportional Distribution**, 222 protocols, 38 ARP firewalls, 252 limiting responses, 282 CHAP, 119 conversion, 226 fuzzing, 22-24 H.323 overview of, 48-52 security profiles, 52-57 IPSec, 73 messages inspection, 108 malformed, 22-24 MGCP, 74-75 security profiles, 75–77 open VoIP, 11 PAP, 119 proxy servers, 118 RIP, 251 RTCP, 48 RTP, 48 intercepted information, 332 SRTP, 188-193 SCCP, 253 SDP, 58, 226 SIP, 7, 57-58 call flow, 60-61 components, 58-59 encryption, 182 filtering, 158 limitations of, 198–199 malformed messages, 144 modes, 130 overview of, 58-67 registration hijacking, 195 S/MIME, 183-188 security profiles, 67-73 SRTP, 267 stacks, 13 UDP, 33, 274

PROTOS SIP version, 144 provisioning, 315 proxy servers, 59 user-to-proxy authentication, 176–179 PSTNs (public switched telephone networks), 5 LI, 292 public (asymmetric) key cryptography, 92–93 DSA, 95–96 RSA, 93–95 public access networks, SBCs, 206–207 public networks, 11 public service telephone networks. *See* PSTNs

Q

QoS (quality of service), 8 abuse, 29–30 firewalls, 250 marking, 228 SBCs, 207

R

RADIUS (Remote Authentication Dial-In User Service), 120, 264 Random Select, 222 realms, 113 SBCs, 203 real-time communications, 11 **Real-Time Control Protocol (RTCP), 38** reconstruction, 34 redirect servers, 59 redirection. 336 **Redirection message**, 334 reference models, LI, 294-297 regional power outages, 22 **REGISTER requests**, 59 registration flooding, 20, 129 hijacking, 27 SIP, 195 phone control, 261-262 SIP with Digest Authentication, 177 timer control, 217-220 regulatory mandates, SBCs, 207

Release message, 334 **Remote Authentication Dial-In User Service** (RADIUS), 120, 264 replay attacks, 326 reputation systems, 169 requests calls, flooding, 20, 129 interfaces, LI, 297-300 **REGISTER**, 59 requirements for LEAs, 293-294 rerouting calls, 35 residential gateways, 74 resources, monitoring, 134 responses ARP, limiting, 282 interfaces, LI, 297-300 RestartInProgress (RSIP), 75 restricted cone NAT, 109 restrictions access ports, 280 after-hours call blocking, 266-267 COR, 264-266 rich media. 8 servers, 119 services, 7 rights misrepresentation, 39 ring tones, 8 **RIP** (Routing Information Protocol), 251 Rivest, Shamir, and Adleman. See RSA Round Robin, 221 routed mode ASA and PIX firewalls, 251-252 FWSM firewalls, 256 routers, 13 aggregation, 327-328 routing, EIGRP, 252 **RQNT** (NotificationRequest), 75 RSA (Rivest, Shamir, and Adleman), 93-95 key distribution, 102 **RSIP** (RestartInProgress), 75 **RTCP** (Real-Time Transport Control Protocol), 48 **RTP** (Real-Time Transport Protocol), 48 intercepted information, 332 SRTP, 71, 188-193 rtpw command, 192

S

S/MIME (Secure/Multipurpose Internet Mail Extensions), 33, 70-71, 183-188 bodies, formatting, 186-188 certificates, 184 key exchanges, 185-186 limitations, 198 salting key (KS), 53 saving audio, 161 SBCs (Session Border Controllers), 113-114, 203 access, 208 borders between access and core networks. 206-207 between core and peer networks, 207 troubleshooting, 204-206 functionality, 208, 226-228 DoS protection, 213–216 LI. 224-225 NAT traversal, 222–224 network topology hiding, 208-212 overload protection, 216-222 load balancing, 220-222 peer, 208 service architecture design, 228-229 Active-Active, 231 Active-Standby, 230-231 high availability, 229-230 network connectivity, 232-234 policy analysis, 234-237 traffic flow optimization, 239-244 virtualization, 237-238 scans, prespoofing, 162 SCCP (Skinny Call Control Protocol), 253 SDP (Session Description Protocol), 58, 226, 302 Secure Hash Algorithm. See SHA Secure Hash Standard (SHS), 98-99 Secure Multipurpose Internet Mail Extensions. See S/ MIME Secure Real-Time Transport Protocol. See SRTP Secure Shell (SSH), 316 security, 9 authentication, 175 user-to-proxy, 176-179 user-to-user, 179-182 COR, 264-266

data security infrastructure, 15 devices. 108 lack of, 11 LI, 114–116 NAT, 113 SBC. 113-114 VoIP-aware firewalls, 109 DoS attacks, 128 intentional flooding, 129–138 unintentional flooding, 138-143 encryption, 182 media. 188–193 S/MIME, 183-188 firewalls, 249-251 ASA and PIX, 251-256 FWSM, 256-258 limitations, 258-259 GUI management, Unified CME, 263-264 IPSec, 73, 195 legacy systems, 14 LI, 326-327 malformed messages, 143-144 analysis, 150-153 mitigation, 154 simulation, 144-150 network layer, 193-195 profiles, 52-57 MGCP, 75-77 SIP, 67-73 sniffing/eavesdropping, 154 analysis, 158-161 mitigation, 161 simulation, 154–158 sources of vulnerability, 10-12 components, 12-13 spam, 165 IM. 167 mitigation, 168-172 SPPP, 167-168 voice, 165-166 spoofing, 162 analysis, 164-165 mitigation, 165 simulation. 162–164 TLS, 253 transport layer, 193-195 Unified CM, 267-269

servers call processing, 117–120 DHCP, preventing fraudulent, 280 impersonating, 196 impersonation, 28-29 LI. 114–116 NAT traversal, 222-224 proxy, 59 user-to-proxy authentication, 176-179 redirect, 59 rich media, 119 **TFTP**, 12 UAS, 58, 175 web, 13 proxy Service Independent Interception (SII), 292, 308 Service Level Agreement (SLA), 229 Service Provider Administration Function (SPAF), 116, 297 ServiceInstance message, 334 services architecture, 5 Active-Active, 231 Active-Standby, 230-231 high availability, 229-230 SBCs. 228–229 devices. 116 call processing servers, 117-120 CPE. 116-117 mobility, 7 rich media, 7 security profiles, 52-57 spoofed messages, 24-30 threats against availability, 20-30 against confidentiality, 30-34 against integrity, 34–38 against social context, 38–43 Session Border Controllers. See SBCs Session Description Protocol. See SDP Session Initiation Protocol. See SIP sessions borders, 203 keys, 188-190 media hijacking, 27 SIP configuration, 61-67 tearing down, 196

SHA (Secure Hash Algorithm), 84 ShiftRows() function, 90 SHS (Secure Hash Standard), 98-99 signaling authentication, 271 encryption, 273 H.235, 54 signatures digital, 95 security (H.235), 55-56 SII (Service Independent Interception), 292, 308 Simple Network Management Protocol version 3 (SNMPv3), 316 simulation flooding attacks, 129-135 malformed messages, 144-150 sniffing/eavesdropping, 154-158 spoofing, 162-164 simultaneous rings on multiple phones, 8 SIP (Session Initiation Protocol), 7, 57–58 call flow, 60-61 components, 58-59 encryption, 182-193 limitations of, 198-199 messages filtering, 158 malformed, 144 modes, 130 NAT. 111 overview of, 58-67 P-DCS headers, 309-313 registration with Digest Authentication, 177 hijacking, 195 security profiles, 67-73 session configuration, 61-67 **URIs. 184** SIP INVITE message, 23 SIP OPTION, 220 SIP Proxy server certificates, 269 SIP Swiss Army Knife (SIPSAK), 129 SIPcrack command, 162 SIPSAK (SIP Swiss Army Knife), 129 Skinny Call Control Protocol (SCCP), 253 SLA (Service Level Agreement), 229

sniffing, 154 analysis, 158-161 mitigation, 161 simulation, 154-158 social context, threats against, 38-43 softphones, 117 Softswitch, 118 **SPAF (Service Provider Administration** Function), 116, 297 spam, 165 calls, 39 IM, 167 mitigation, 168-172 presence, 41-42 SPPP, 167-168 voice, 165-166 Spam over Instant Messaging (SPIM), 40-41 SPam over IP Telephony (SPIT), 39 spoofing, 162 analysis, 164-165 messages, 24-30 mitigation, 165 simulation, 162-164 SPPP (presence spam), 41-42, 167–168 SRST-enabled gateway certificates, 268 SRTP (Secure RTP), 71, 188-193, 267 packet processing, 190-191 testing, 191-193 SSH (Secure Shell), 316 SSI (Service Independent Interception), 292 stacks, protocols, 13 static load balancing, 220 stream analysis, 160 SubBytes() function, 89 SubjectSignal message, 334 superusers, 264 Switch, 278–282 switches, 13 symmetric (private) key cryptography, 84-85 3DES, 87 AES, 89-92 DES, 85-87 symmetric NAT, 110

Т

TACACS+, 120 TCP (Transmission Control Protocol), SYN attacks, 128 teardown, calls, 25 tearing down sessions, 196 telecommunication service providers (TSPs), 292 **Telecommunications Industry Associations** (TIA), 292 TerminationAttempt message, 334 testing malformed messages, 144-150 analysis, 150-153 mitigation, 154 negative testing tools, 129 sniffing/eavesdropping, 154-158 analysis, 158-161 mitigation, 161 SRTP, 191-193 Turing tests, 168 **TFTP servers**, 12 threats against availability, 20-30 against confidentiality, 30-34 DoS attacks, 128 intentional flooding, 129–138 unintentional flooding, 138-143 against integrity, 34-38 malformed messages, 143-144 analysis, 150-153 mitigation, 154 simulation, 144-150 models, 195-198 sniffing/eavesdropping, 154 analysis, 158–161 mitigation, 161 simulation, 154–158 against social context, 38-43 spam, 165 IM, 167 mitigation, 168–172 SPPP, 167-168 voice, 165-166

spoofing, 162 analysis, 164-165 mitigation, 165 simulation, 162-164 **TIA (Telecommunications Industry** Associations), 292 timers, registration control, 217-220 TLS (Transport Layer Security), 33, 71–73, 253 limitations, 199 toll fraud, 26 tools libSRTP, 191-193 malformed messages, 144-150 analysis, 150-153 mitigation, 154 negative testing, 129 SIPSAK, 129 Wireshark, 156 topologies hiding, 208-212 ToS (type of service), 228 tracking call patterns, 32-33 traffic black, 213 call flooding, 20-22 firewalls, 249-251 ASA and PIX, 251-256 FWSM, 256-258 limitations, 258-259 flooding, 132 flow optimization, 239-244 gray, 214 white, 213 transactions call hijacking, 26 MGCP, 74 transcoding, 226-227, 333 transferring calls, 303 transfers, 336 translation addresses, 49 dialed digital, 50 NAT, 21, 109-113 numbers, 227-228 transparent mode ASA and PIX firewalls, 252 FWSM firewalls, 256

Transport Layer Security. See TLS traversal media, 241 NAT, 222-224 Triple DES (3DES), 87 troubleshooting borders, 204-206 between access and core networks, 206-207 between core and peer networks, 207 device configuration, 22 endpoints, 22 static load balancing, 220 trunking gateways, 74, 328 TSPs (telecommunication service providers), 292 tunneling, 195 Turing tests, 168 type of service (ToS), 228 types of firewalls, 249-251 ASA and PIX, 251-256 FWSM, 256-258 limitations, 258-259 of keys, 189 of NAT, 110

U

UAC (User Agent Client), 58, 175 UAS (User Agent Server), 58, 175 uBR (Universal Broadband Router), 327 UDP (User Datagram Protocol), 33, 274 unauthorized creation and detection of intercepts, 303 Unified CM (Cisco Unified Communications Manager), 267 authentication, 269-273 configuration, 275-277 encryption, 273-275 integrity, 269-273 node certificates, 268 security, 267-269 **Unified CME (Cisco Unified Communications** Manager Express), 259 access control, 259-261 after-hours call blocking, 266-267 COR, 264-266

phone registration control, 261-262 secure GUI management, 263-264 **Uniform Resource Indicators.** See URIs unintentional flooding, 138-143 SBCs, 206 Universal Broadband Router (uBR), 327 UNIX, resource monitoring, 134 **URIs (Uniform Resource Indicators), 184** limitations of, 199 User Agent Client (UAC), 58 User Agent Server (UAS), 58 user control interfaces, 7 User Datagram Protocol (UDP), 33, 274 User-Based Security Model (USM), 326 user-to-proxy authentication, 176–179 user-to-user authentication, 179-182 USM (User-Based Security Model), 326

V

VACM (View-based Access Control Model), 326 valid call request flooding, 129 valid registration flooding, 129 View-based Access Control Model (VACM), 326 virtual LANs. See VLANs Virtual SBC (VSBC), 237 virtualization, 237–238 VLANs (virtual LANs), 282–284 voice integration, 12 spam, 39, 165–166 VoIP (Voice over IP) applications, 12 benefits of, 6–8 disadvantages of, 8–10 firewalls, 108–109 protocol stacks, 13 VSBC (Virtual SBC), 237 vulnerability components, 12–13 sources of, 10–12

W

web clients/servers, 13 white traffic, 213 Windows, resource monitoring, 135 Wireshark, 156 wiretapping, 14, 114–116 operational issues, 300–304 overview of, 292–293 reference models, 294–297 request/response interfaces, 297–300 requirements for LEAs, 293–294 wrong configuration of devices, 139

Ζ

zone management, 49