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About Me

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About this Presentation

All attacks discussed are either recently developed, or extremely significant

Making the case that attack tools are both available and mature

\boxtimes Divided into three sections:

🕾 Briefly, VoIP Basics

Attacks (Vulns, Attacks, Impact, Tools, Mitigation)

Problems with suggested mitigation actions

I'll be discussing only technical attacks; not social attacks like SPIT, Phishing, etc.

Tim Burton is AWESOME!

Notes on Mitigation

Many times there are no clear-cut "solutions" to any vulnerability or attack

- I will refrain from using the "so just isolate your VoIP network" cop-out "solution"
- Some mitigation techniques suggested work; In part three, I'll only be discussing:
 Those that don't work well
 - Those that have significant drawbacks
 - Those that have significant barriers to implementation

C.M.A.

All Mars Attacks! Audio and Video is Copyright Warner Brothers Pictures (Time Warner Entertainment)



VoIP Basics

VoIP for the uninitiated...

Terminology

VoIP - Voice over Internet Protocol **Call - the session aggregate of signaling and** media between endpoints Endpoint - Point where a call terminates Soft-phone - VoIP phone implemented entirely in software Hard-phone - VoIP phone with a physical presence, also sometimes referred to as a

"handset"

PSTN - Public Switched Telephone Network, or your traditional telephony networks.

Signaling vs. Media

Separate channels for signaling information vs. media (bearer) data due to abuse

Adopted from traditional telephony systems

Some protocols like IAX/IAX2 combine these into a single channel

Protocols & Ports

Signaling

Session Initiation Protocol (SIP) : TCP/UDP 5060,5061
 Session Description Protocol (SDP) : Encapsulated in SIP
 Media Gateway Control Protocol (MGCP) : UDP 2427,2727
 Skinny Client Control Protocol (SCCP/Skinny) : TCP 2000,2001
 Real-time Transfer Control Protocol (RTCP) : (S)RTP+1

Media

Real-time Transfer Protocol (RTP) : Dynamic Secure Real-time Transfer Protocol (SRTP) : Dynamic

Hybrid

☑ Inter-Asterisk eXchange v.1 (IAX): UDP 5036 (obsolete)
 ☑ Inter-Asterisk eXchange v.2 (IAX2) : UDP 4569

H.323 Protocol Suite & Ports

Signaling H.245 - Call Parameters - Dynamic TCP ₩ H.225.0 ☑ Q.931 - Call Setup - TCP 1720 **RAS - UDP 1719** Audio Call Control - TCP 1731 **RTCP - RTP Control - Dynamic UDP** Media 🖾 RTP - Audio - Dynamic UDP RTP - Video - Dynamic UDP

Audio Codecs

DoD CELP - 4.8 Kbps GIPS Family - 13.3 Kbps and up WiLBC - 15 Kbps, 20ms frames / 13.3 Kbps, 30ms frames ITU G.711 - 64Kbps (a.k.a. alaw / ulaw) XITU G.722 - 48 / 56 / 64 Kbps ITU G.723.1 - 5.3 / 6.3 Kbps, 30ms frames X ITU G.726 - 16 / 24 / 32 / 40 Kbps **ITU G.728 - 16 Kbps** XITU G.729 - 8 Kbps, 10ms frames Speex - 2.15 to 44.2 Kbps, Free Open-Source codec The http://www.voip-info.org/wiki-Codecs



Attacks Against Availability



Flooding



Flooding

- Most hard-phones are limited or underpowered hardware
- Protocols provide unauthenticated and unauthorized functions

☆Attack:

SIP INVITE, OPTIONS

Bogus RTP media packets

Flood the device with network protocol packets:
 TCP SYN
 UDP

Effect:

Degraded call quality
 Device crash, halt, freeze, or respond poorly

Flooding

⊠Tools:

Scapy - General purpose packet tool
 Mttp://www.secdev.org/projects/scapy/

 InviteFlood - SIP Invite flooder
 Mttp://www.hackingexposedvoip.com/tools/inviteflood.tar.gz

 IAXFlood - IAX protocol flooder
 Mttp://www.hackingexposedvoip.com/tools/iaxflood.tar.gz

 UDPFlood - General UDP flooder
 Mttp://www.hackingexposedvoip.com/tools/udpflood.tar.gz

 RTPFlood - RTP protocol flooder
 Mttp://www.hackingexposedvoip.com/tools/udpflood.tar.gz

Mitigation:

Protect your core network devices from external access
 Rate-limit VoIP traffic at points of control

Fuzzing

Protocol stack implementations suck

Attack:

Send malformed messages to a device's input vectors

Most endpoint devices will crash, halt, freeze, or otherwise respond poorly
 Some core devices may behave similarly
 You may find bugs that do more than just provide a Denial of Service

Fuzzing

⊠Tools: **PROTOS Suite - SIP, HTTP, SNMP** Mattheway and the second of 🕾 ohrwurm - RTP http://mazzoo.de/blog/2006/08/25#ohrwurm Tuzzy Packet - RTP, built-in ARP poisoner ²² http://libresource.inria.fr/projects/VoIP_Security/fuzzypacket 🖾 Other tools ™ http://www.threatmind.net/secwiki/FuzzingTools ☑ Mitigation:

Use open-source soft-phones and hard-phone firmware
 Demand resilient devices from your device vendor
 Ask about and review your vendor's QA processes

Forced Call Teardown



Forced Call Teardown

[™]Vulnerabilities:

Most protocols are unencrypted and do not authenticate all packets

The signaling channel can be monitored

Attack:

Inject spoofed call tear-down messages into the signaling channel such as:

SIP: BYE

SCCP: Reset (Message type 159 (0x9f))

☑ IAX: HANGUP (Frame type 0x06, Subclass 0x05)

⊠Effect:

☑ DoS: A call in progress is forcibly closed.

Forced Call Teardown

⊠Tools:

Teardown - SIP BYE injector
 http://www.hackingexposedvoip.com/tools/teardown.tar.gz
 sip-kill - Injects valid SIP messages such as BYE into an existing session
 http://skora.net/uploads/media/sip-kill
 sip-proxykill - Similar technique against SIP proxies
 http://skora.net/uploads/media/sip-proxykill

Mitigation:

Encrypt the signaling channel
 Authenticate every signaling message

Registration/Call Hijacking

⊠Vulnerability:

Signaling protocols are unencrypted

☑Attack:

Sniff a legitimate endpoint registration
 Use sniffed information and credentials to replace the legitimate registration

Sniff a call-setup message

Effect

New calls for the endpoint are routed to the malicious device rather than the legitimate device

Registration Hijacking

Tools

Registration Hijacker

Mackingexposedvoip.com/tools/reghijacker.tar.gz

Registration Remover

http://www.hackingexposedvoip.com/tools/eraseregistrations. tar.gz

Registration Adder

http://www.hackingexposedvoip.com/tools/add_registrations.t ar.gz

RedirectPoison

Mitigation

Encrypt signaling traffic

Attacks Against Integrity

Media Hijacking

- Signaling protocols are unencrypted and unauthenticated
- Signaling extends to endpoint device

☑Attack:

- Inject malicious signaling messages into a signaling channel
- Send new signaling messages to endpoints or services
- ⊠Effect:

Media redirection, duplication, or termination

Media Hijacking Example



Media Hijacking Example



Media Hijacking Example



Media Hijacking

⊠Tools:

Sip-redirectrtp + rtpproxy
 http://skora.net/voip/attacks/
 Mitigation:
 Encrypt the signaling channel
 Fix protocols to authenticate ALL signaling messages related to a call

Media Injection



Media Injection

WVulnerability

Media channel packets are unauthenticated and unencrypted

☑Attack:

Inject new media into an active media channel
 Replace media in an active media channel
 Effect:
 Modification of media
 Replacement of media
 Deletion of media

Media Injection Example: RTP Real-Time Transfer Protocol **Normally UDP Transport** Requisites: Able to observe a legitimate RTP session Adjust sequence numbers of packets to be injected so that they will arrive "before" legitimate packet Send away!

RTP Injection



RTP Injection



☑ IPID = IPID + spoof-factor

Sequence = sequence + spoof-factor

Image: Timestamp + (payload-len * spooffactor)

Demo!

RTP Audio Injection

Media Injection

RTPInsertSound Mattheway and the second state of the secon **RTPMixSound** ²² http://www.hackingvoip.com/tools/rtpmixsound v3.0.tar.gz Mitigation Authenticate or verify media packets Encrypt the media channel

Caller-ID Spoofing



Caller-ID Spoofing

⊠Vulnerability:

Protocols are un-authorized and un-verified end-to-end
 End-point supplied data is not challenged

Many automated systems use Caller-ID information to authenticate users

☑Attack:

☑ Initiate a call with falsified Caller-ID information

Effect:

An attacker may appear to the called party as someone they are not

An attacker may be erroneously authenticated

Caller-ID Spoofing

⊠Tools: Most soft-phones Asterisk IPBX Service providers that honor usersupplied Caller-ID information ™ http://www.iax.cc/ - IAX VoIP provider ☆ http://www.telespoof.com/ - For "business" use Multip://www.fakecaller.com/ - Text to Voice "prank" messages! [™]Mitigation: Don't honor user-supplied Caller-ID information Don't trust Caller-ID information for user authentication

Attacks Against Confidentiality

Eavesdropping the Media



Eavesdropping the Media

- RTP un-encrypted on the wire
 Media traffic can be sniffed and recorded
 Attack:
 - Record the media packets
 - Reconstruct the payload into an easily playable media file

Effect: Calls are not private!

SIP_CA	LL_RTP_G7	11.pcap - Wiresharl	(
<u>Eile E</u> dit	<u>⊻</u> iew <u>G</u> o		Statistics Help	
84 64	. 🕺 🌘	i 💓 🖻 I	Summary Protocol Hierarchy	
Eilter: sip	rtp		Conversations Endpoints	Expression ⊆lear
No. +	Time	Source	IO Graphs	Protocol Info
1 2 3 152	0.000000 0.007889 0.047524 4.056633	200.57.7.19 200.57.7.20 200.57.7.20 200.57.7.20	Conversation List Endpoint List Service Response Time	SIP/SD Request SIP Status: SIP Status: SIP Request
153 498 499 500	4.072335 8.477925 8.479371 8.479599	200.57.7.19 200.57.7.20 200.57.7.20 200.57.7.20	ANSI ANSI	SIP Status: SIP/SD Status: RTP Payload RTP Payload
515 517 522	8.517413 8.524137 8.529324	200.57.7.20 200.57.7.19 200.57.7.19	 ➡ H.225 MTP3 ■ DTP 	RTP Payload SIP Request
524 528 530	8.537392 8.549261 8.565236	200.57.7.20 200.57.7.19 200.57.7.20	SCTP	Stream Analysis ad RTP Payload

🕂 Wireshark: RTP Streams

Detected 3 RTP streams. Choose one for forward and reverse direction for analysis

Src IP addr 🕞	5rc port	Dest IP addr)est port	SSRC	Payload		Packets	Lost	Max Delta (ms)	Ν
200.57.7.204	8000	200.57.7.196	40376	3535621694	ITU-T G.711	PCMA	548	0 (0.0%)	5843.74	
200.57.7.196	40376	200.57.7.204	8000	1492336106	ITU-T G.711	PCMA	891	0 (0.0%)	379.91	
200.57.7.202	30000	200.57.7.196	40362	11837	ITU-T G.711	PCMA	6	0 (0.0%)	30.04	
•										
		Sele	ect a forwa	ard stream with	left mouse b	utton				
		, Select a	reverse s	tream with SHI	FT + left mou	ise butto	n ,			L.,
Unselect	Find Rever	se Save <u>A</u> s	Mark	Packets Pre	epare Filter	⊆o	ру	Analyze	⊆lose	

🕂 Wireshark: RTP Stream Analysis

Forward Direction

Reversed Direction

Analysing stream from 200.57.7.204 port 8000 to 200.57.7.196 port 40376 SSRC = 3535621694

Packet +	Sequence	Delta (ms)	Jitter (ms)	BW (kbps	Marker	Status	
499	1	0.00	0.00	1.60	SET	[Ok]	
500	2	0.23	1.24	3.20		[Ok]	
515	3	37.81	2.27	4.80		[Ok]	
524	4	19.98	2.13	6.40		[Ok]	
530	5	27.84	2.49	8.00		[Ok]	
535	6	12.35	2.81	9.60		[Ok]	
577	7	1043.44	3.67	1.60		[Ok]	
580	8	19.90	3.45	3.20		[Ok]	
583	9	20.02	3.23	4.80		[Ok]	
584	10	0.18	4.27	6.40		[Ok]	
589	11	19.95	4.01	8.00		[Ok]	
593	12	20.09	3.76	9.60		[Ok]	
597	13	20.02	3.53	11.20		[Ok]	
601	14	20.07	3.31	12.80		[Ok]	
605	15	23.39	3.32	14.40		[Ok]	
609	16	16.82	3.31	16.00		[Ok]	Υ.
	Max delta Total RTP	a = 5.843742 se 9 packets = 548	c at packet no. 21 (expected 548)	95 Lost RTP pa	ockets = 0 (0.00%	6) Sequence errors = 0	

Saus pauload	Saug as CSV	Defrech	Jump to	Graph	Next pep Ok	Class
Save payload	Dave as Cov	Refresh	2ump co	Graph	Next Hon-OK	Close

New Folder	Deļete File	Rena	ame File
	C:\Prog	gram File:	s\Wireshark 🛛 🕶
Folders			<u>Files</u>
4			adns_dll.dll
<i></i>			AUTHORS-SHORT
diameter\			AUTHORS-SHORT-FORMAT
dtds\			capinfos.exe
etc\			capinfos.html
help\			cfilters
lib\			colorfilters
plugins\			comerr32.dll
	Format: 🗿 .	raw	🔘 .au
	Channels: 🔘 fi	orward	reversed • both
Selection: C:\Pro	gram Files\Wiresh	ark	
			OK Cancel

Eavesdropping the Media

^I Tools: Ethereal / Wireshark ™ http://www.wireshark.org/ 🖾 Cain & Abel ™ http://www.oxid.it/cain.html Womit - Targets Cisco devices ™ http://vomit.xtdnet.nl/ Etherpeek VX ☆ http://www.wildpackets.com/products/etherpeek/overview Mitigation: Encrypt the media channel

Directory Enumeration

⊠Vulnerabilities:

Protocols provide unauthenticated functionality
 Protocols respond differently to valid vs. invalid usernames
 Protocols are unencrypted on the wire

Attack:

Active: Send specially crafted protocol messages which elicit a telling response from the server

🕾 Passive: Watch network traffic for device registration messages

密Effect:

Valid usernames are disclosed and may be used in a more targeted attack such as pass-phrase cracking.

Directory Enumeration Example

Send this to target SIP device:

OPTIONS sip:test@172.16.3.20 SIP/2.0 Via: SIP/2.0/TCP 172.16.3.33;branch=3afGeVi3c92Lfp To: test <sip:test@172.16.3.20> Content-Length: 0

Receive:

SIP/2.0 404 Not Found

Directory Enumeration

⊠Tools:

SIPCrack - Sniffs traffic for valid usernames and then attempts to crack their passwords

☆ http://www.remote-exploit.org/index.php/Sipcrack

enumIAX - Uses IAX REGREQ messages against Asterisk http://www.tippingpoint.com/security/materials/enumiax-0.4a.tar.gz

SIPSCAN - Uses SIP OPTIONS, INVITE, and REGISTER messages against SIP servers

The http://www.hackingexposedvoip.com/tools/sipscan.msi

Mitigation:

Encrypt signaling to prevent passive enumeration
 Fix protocols that respond differently to valid vs. invalid username registrations.

Configuration Disclosure: Infrastructure

[™]Vulnerability:

☑ Most hard-phones use FTP or TFTP when booting
 ☑ TFTP is an insecure protocol
 ☑ FTP is an insecure protocol

FTP: Sniff the device's login credentials
 TFTP: Guess or sniff the filenames
 Grab the configuration file and firmware from the server
 Or just sniff the firmware and configuration file from the wire

⊠Effect:

Disclosure of sensitive information such as:
 Usernames / Passwords
 Call Server, Gateway, Registration Server, etc.
 Available VoIP services

Configuration Disclosure: Infrastructure

竖Tools:

🕾 Ethereal / Wireshark

™ http://www.wireshark.org/

I Deductive Reasoning

 \mathbb{Z} Cisco phones have MAC based filenames:

SEP<eth.addr>.cnf.xml

₩ MGC<eth.addr>.cnf

 $\overline{\mathbb{X}}$ Then there's defaults:

XMLDefault.cnf.xml

SIPDefault.cnf

🖾 dialplan.xml

TFTP-Bruteforce - Brute forces TFTP filenames

Mattp://www.hackingexposedcisco.com/tools/TFTP-bruteforce.tar.gz

Mitigation:

 \mathbb{R} Don 't use TFTP! FTP is better, but still not secure...

🕾 Use non-default filenames

Configuration Disclosure: Device

⊠Vulnerability:

Hard-phones provide management interfaces
 VXW orks remote debugging and console port open

☑ Attack:

Point a browser at the device on port 80
SNMP-walk the device
Attach a remote VXWorks debugger

恐 Effect:

Disclosure of sensitive information such as:
 Usernames / Passwords
 Call Server, Gateway, Registration Server, etc.
 Available VoIP services
 Device internals

Configuration Disclosure: Device

[™]Tools:

 Web Browser - Connect to port 80
 SNMPwalk - retrieve a subtree of management values http://net-snmp.sourceforge.net/docs/man/snmpwalk.html

 GDB configured for VXWorks support

Mitigation:

Disable device admin ports like HTTP and SNMP
 Disable remote debugging ports

Mitigation

Encrypt the Media Channel

 Not many devices support SRTP yet
 No standard way to negotiate or send keys
 Keys are generally negotiated or sent in the unencrypted signaling channel anyway
 ZRTP: DH Key Negotiation within the media channel, doesn't comply with CALEA

May use IPSec or TLS, but...

Encrypt the Signaling Channel

There is no standard way to do this
Alternatives to encrypting the signaling protocol itself include:

IPSec to encrypt at the network layer

Mot scalable

☑ Issues with call set-up times

TLS to encrypt at the transport layer
 Not end-to-end
 Issues with trust; no global PKI

Authenticate All Signaling Messages

Requires that you update/fix the protocol

The nature of VoIP requires that unknown parties be able to initiate sessions

Can potentially wrap the protocol in an authenticating transport like IPSec or TLS

Fix the Protocols

Not an immediate solution More time consuming with open / standards based protocols You have to convince a committee there is a problem **Deliberation takes time** May be faster / easier with proprietary protocols But you have to convince the vendor there is a problem

Don't Trust Caller-ID

Unfortunately, users have been trained to believe that Caller-ID is trustworthy
 Caller-ID *should* be trustworthy
 Will take time to educate users

Demand resilient devices from your VoIP device vendor Servendors aren't motivated to improve device security

Some devices in this area are getting better

Phones are limited by their hardware

Rate-limit Offensive Traffic

- Low-rate floods still effective! (just differently)
- Low-rate floods look like legitimate traffic
 Media doesn't like latency

Don't use TFTP! (or FTP)

Most vendor VoIP architectures don't provide an alternative

Conclusions



Fin.

