

# Just how vulnerable is your phone system?

by Sandro Gauci

# \$ whoami

- Sandro Gauci (from .mt)
- Security researcher and Pentester
- SIPVicious / VOIPPACK for CANVAS
- VOIPSCANNER.com
- Not just about VoIP
- EnableSecurity



# Introducing: security issues

- VoIP attack surface is huge
  - SIP RFC 3261 = 269 pages
  - Referencing a large number of other RFCs
- Most solutions come with other services
  - Web interface, tftp etc

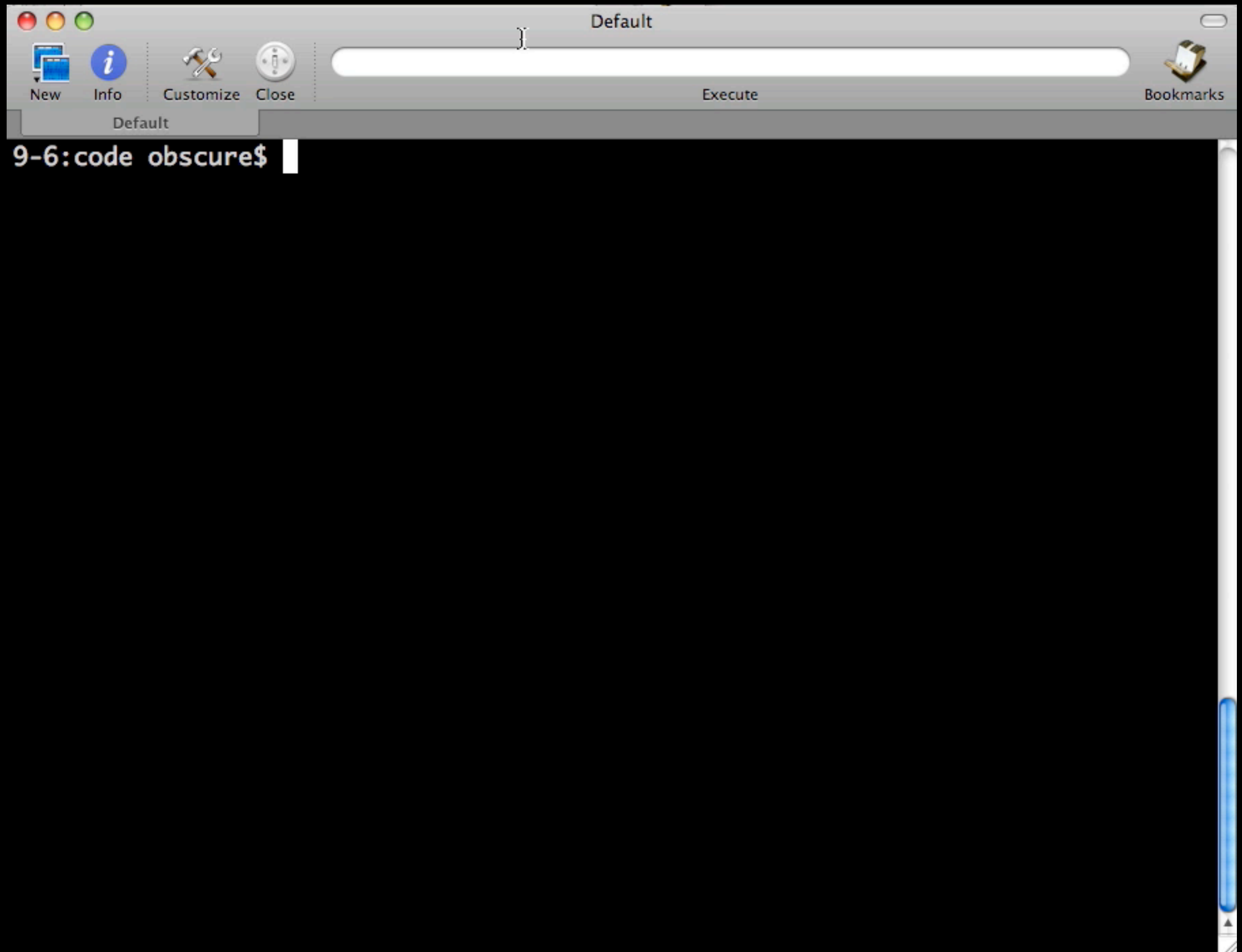
# Protocol specific issues

- SIP on UDP is most common
  - also most vulnerable to passive monitoring
  - no encryption used
- The SIP RFC enables user enumeration
- Peer to peer SIP creates some challenges

# User enumeration on SIP

- Different responses to REGISTER request
- Returns 404 when 'user' is not found
- Returns 200 or 401 when the 'user' exists
  - 200 = no password set!
  - 401 = challenge / authenticate

# enumeration demo



# Peer to Peer SIP

- When endpoints can contact each other
  - they can spoof caller ID
  - they can bypass logging
  - exploit some interesting features in SIP



# Spooing caller ID

Account

Account name:

Protocol: SIP

Use for: ☒ Call ☐ IM/Presence

**General** Voicemail Topology Presence Transport Advanced

User Details

\* User ID

\* Domain

Password

Display name

Authorization name

Domain Proxy

☐ Register with domain and receive calls

Send outbound via:

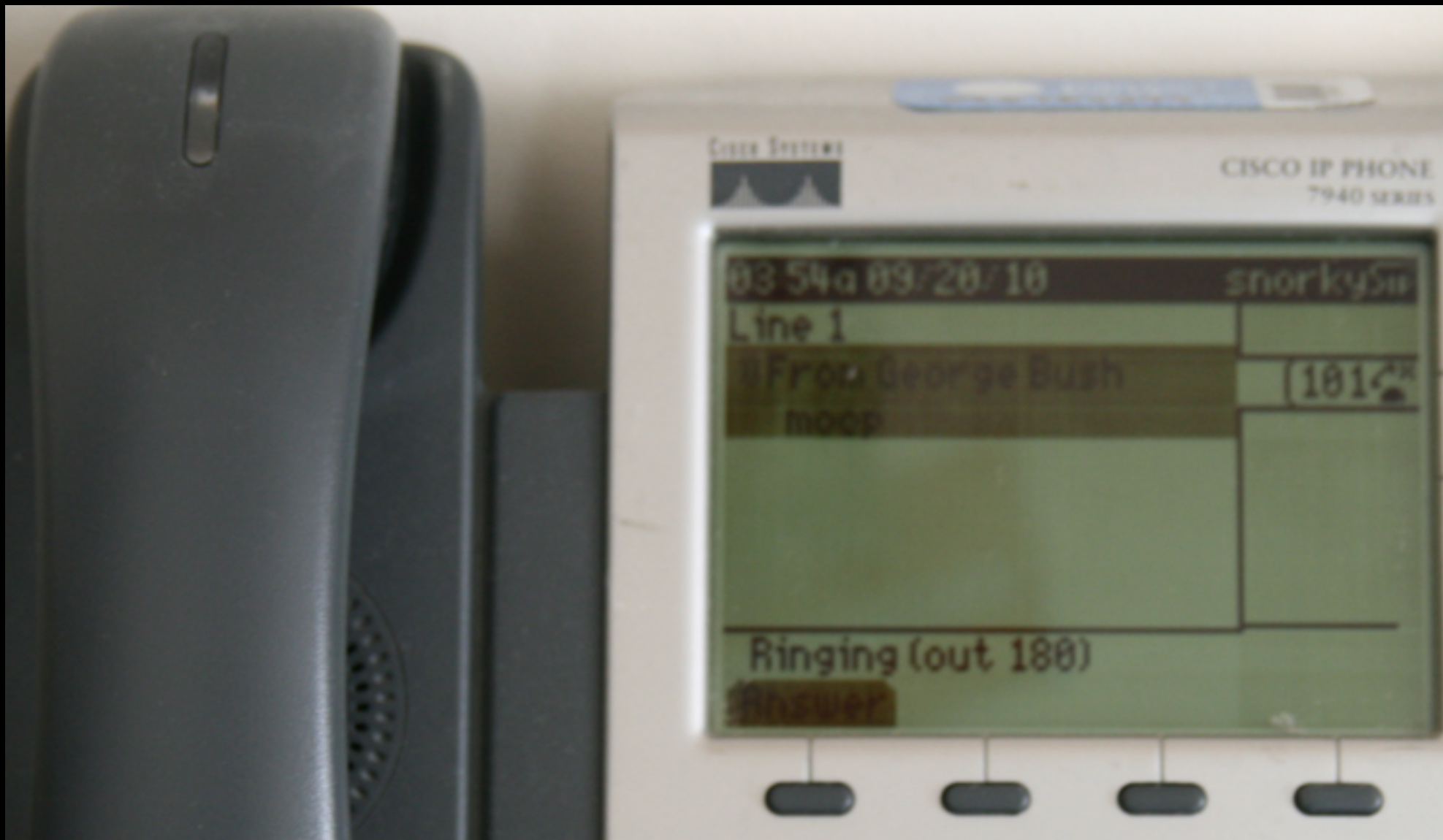
☐ Domain

☒ Proxy Address

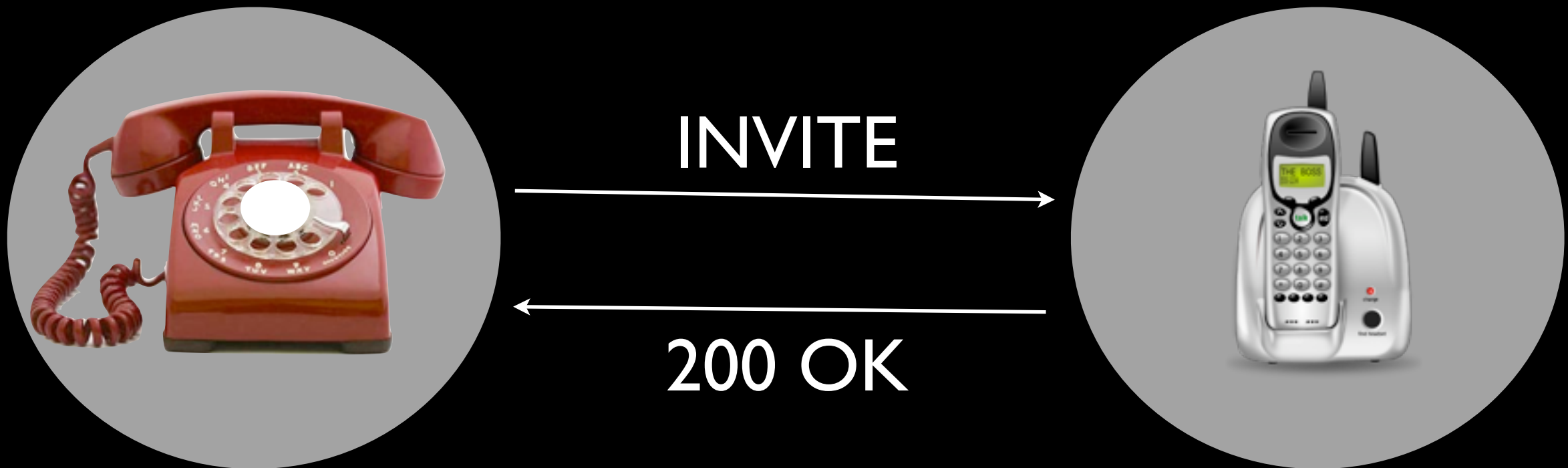
Dial plan

Cancel OK

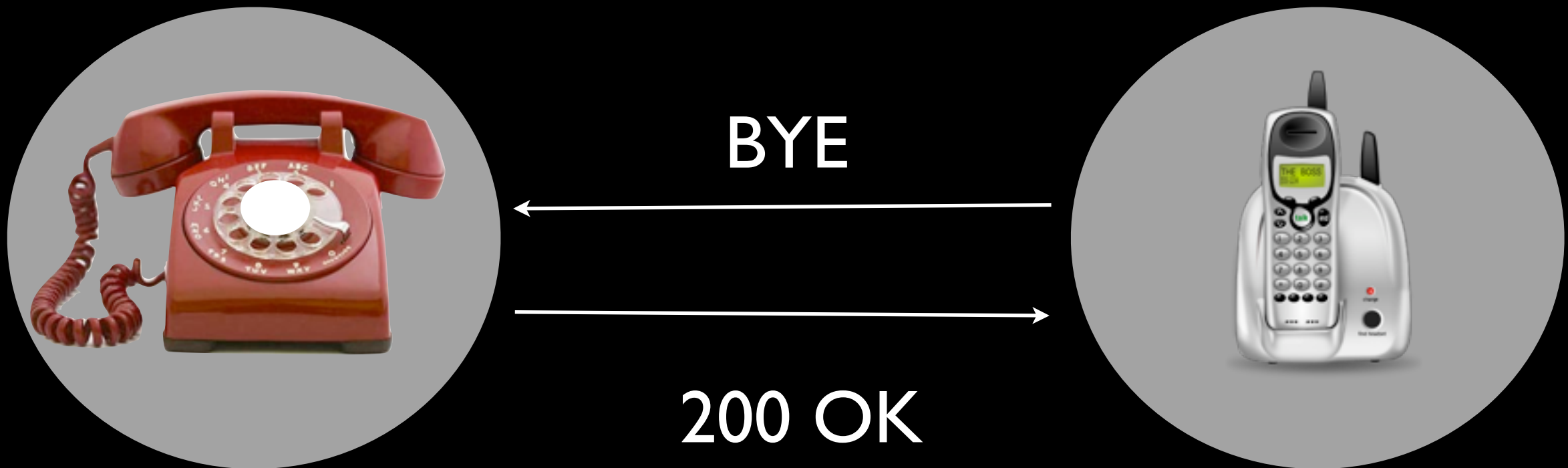
# Spooofing caller ID



# SIP Digest Leak



# SIP Digest Leak

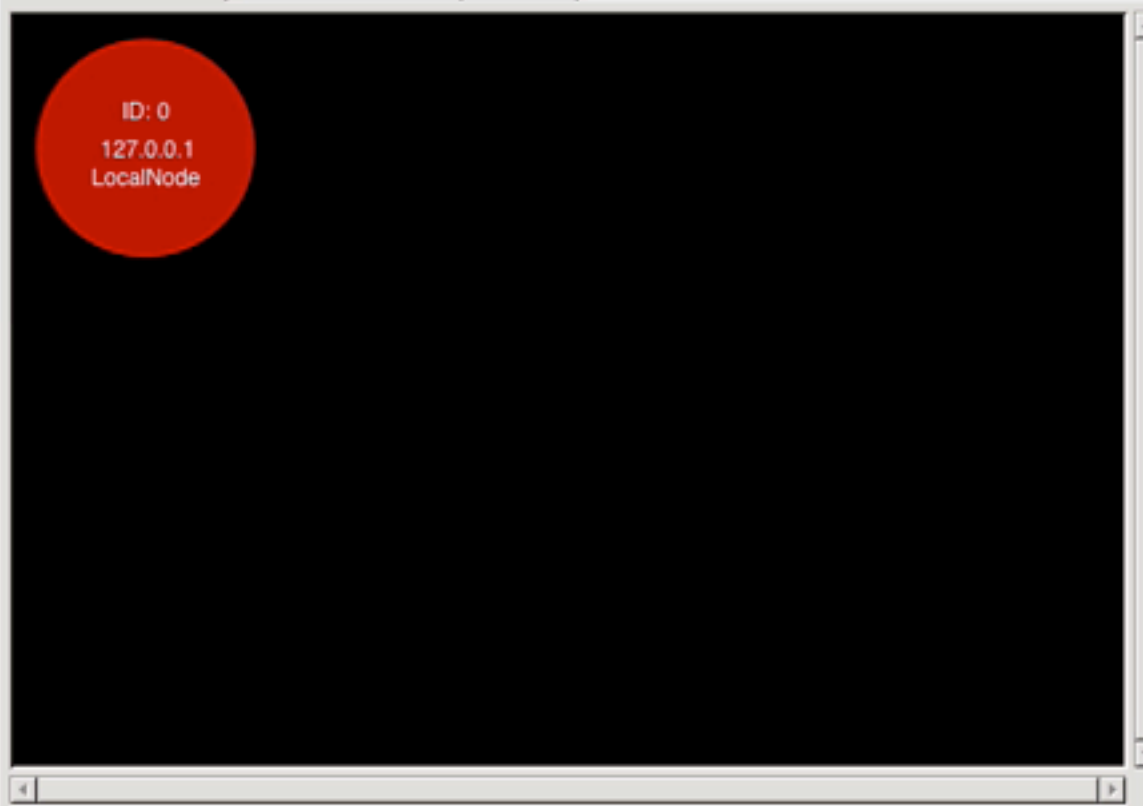


# SIP Digest Leak



# Demo of SIP Digest leak

Name	Description
digestcracker	Digest Offline Cracker
sipdigestleak	SIP Digest leak
-- 2 results for that query --	



# Distro and PBX specific

- Default passwords
- PHP-based web applications
- FreePBX etc: emulation of other systems
- Various other services



# Trixbox defaults

Service	Username	Password
FOP	NA	passw0rd
AMP	admin	amp111
Admin (freepbx)	maint	password
FTP	ftpuser	asteriskftp


# Elastix defaults

Service	Username	Password
web interface	admin	palosanto
freePBX	admin	admin
FOP	admin	eLaStIx.2007
a2billing	admin	mypassword
sugar crm	admin	admin
vTiger	admin	password

\* reference

Elastix - Login page

https://192.168.2.102/vtigercrm/

 **elastix**  
CUSTOM MADE TELEPHONY

» Welcome to Elastix

Please enter your username and password

Username:

Password:

Copyright © 2006 by [PaloSanto Solutions](#)

Waiting for 192.168.2.102...

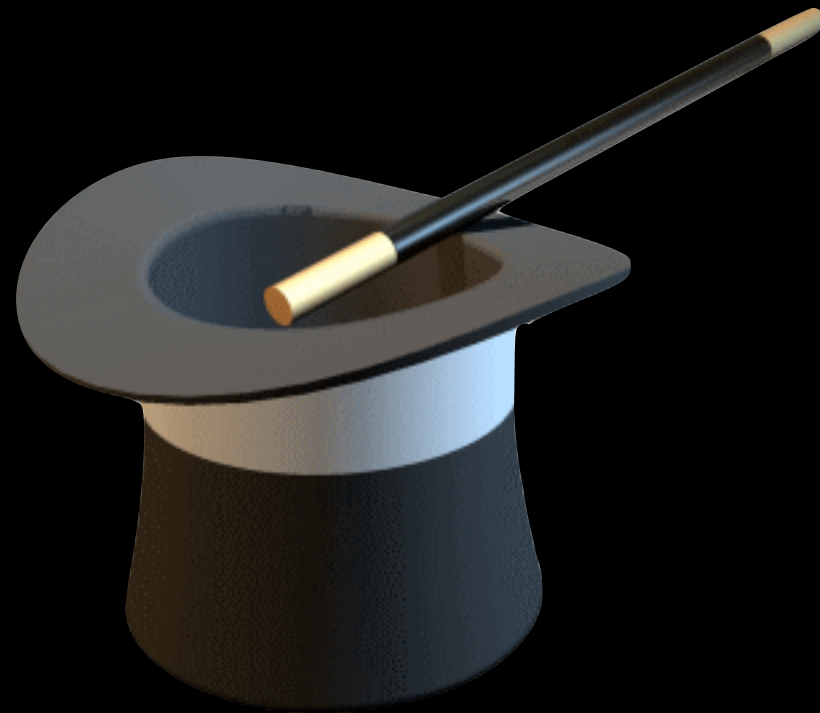
# What's happening?

- Are these vulnerabilities really an issue?
- Which ones are being abused?
- What are their motivations?
- Who are they?

# Introducing voiphun

- Short for “voip honeypot” :-)
- A very simple fake SIP registration server
- And fake proxy too (i.e. takes calls)
- Which can be used as a honeypot
- Still limited in functionality

# What's in voiphun's hat?



# What we're seeing

- Compromised hosts looking for SIP devices
- Attackers trying to make phone calls
- Attackers scanning for extensions with weak passwords

# What we're seeing

- SIPVicious scans
- Custom / unknown scanners
- INVITE scans



# INVITE scan example

```
INVITE sip:00423662701946@xx.xx.xx.xx;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 188.27.208.189:62399;branch=z9hG4bK-d8754z-ffab3c4b5a504640-1---d8754z-
Max-Forwards: 70
Contact: <sip:100@188.27.208.189:62399;transport=UDP>
To: <sip:00423662701946@xx.xx.xx.xx;transport=UDP>
From: "UNKNOWN"<sip:100@xx.xx.xx.xx;transport=UDP>;tag=9a46293c
Call-ID: OGVmNmI1NmU3MTVmYTBmMTliMWZjMzd1YjI2N2U3ZTk.
CSeq: 1 INVITE
Allow: INVITE, ACK, CANCEL, BYE, NOTIFY, REFER, MESSAGE, OPTIONS, INFO, SUBSCRIBE
Content-Type: application/sdp
User-Agent: Zoiper rev.5324
Content-Length: 332
```

```
v=0
o=Zoiper_user 0 0 IN IP4 188.27.208.189
s=Zoiper_session
c=IN IP4 188.27.208.189
t=0 0
m=audio 65287 RTP/AVP 3 0 8 110 98 101
a=rtpmap:3 GSM/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:110 speex/8000
a=rtpmap:98 iLBC/8000
a=fmtp:98 mode=30
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

# I <3 Patterns

- INVITE scans bruteforces phone numbers
- Why not extract those numbers?
- Group them by source IP / country

# INVITE scan I

Came from Romania Data Systems network

00000447799584555  
0000441372456539  
00011442086702315  
0001440129870903  
000441622620388  
0011447876617548  
001442075828187  
00441189780316  
011442076339733  
01447850294946  
0442078375450  
1442072425376  
900441767677666  
9011442082163104  
90447973642015  
9442074998161

# INVITE scan 2

Also from China Telecom (Guangdong) network

#442076501050  
00#442076501050  
011#442076501050  
011441616606065  
0442076501050  
442076501050  
900442076501050  
9011442076501050  
9442076501050

fax number

# INVITE scan 3

Came from China Telecom (Shanxi) network

```
00000447799584555
0000441372456539
0000442076297347
00011442086702315
0001440129870903
0001441844208220
000441622620388
000442073878081
0011442076381111
0011447876617548
001442075828187
001447775742174
00441189780316
011442076339733
012441535610840
01447024074657
-- clipped --
```

# INVITE scan 4

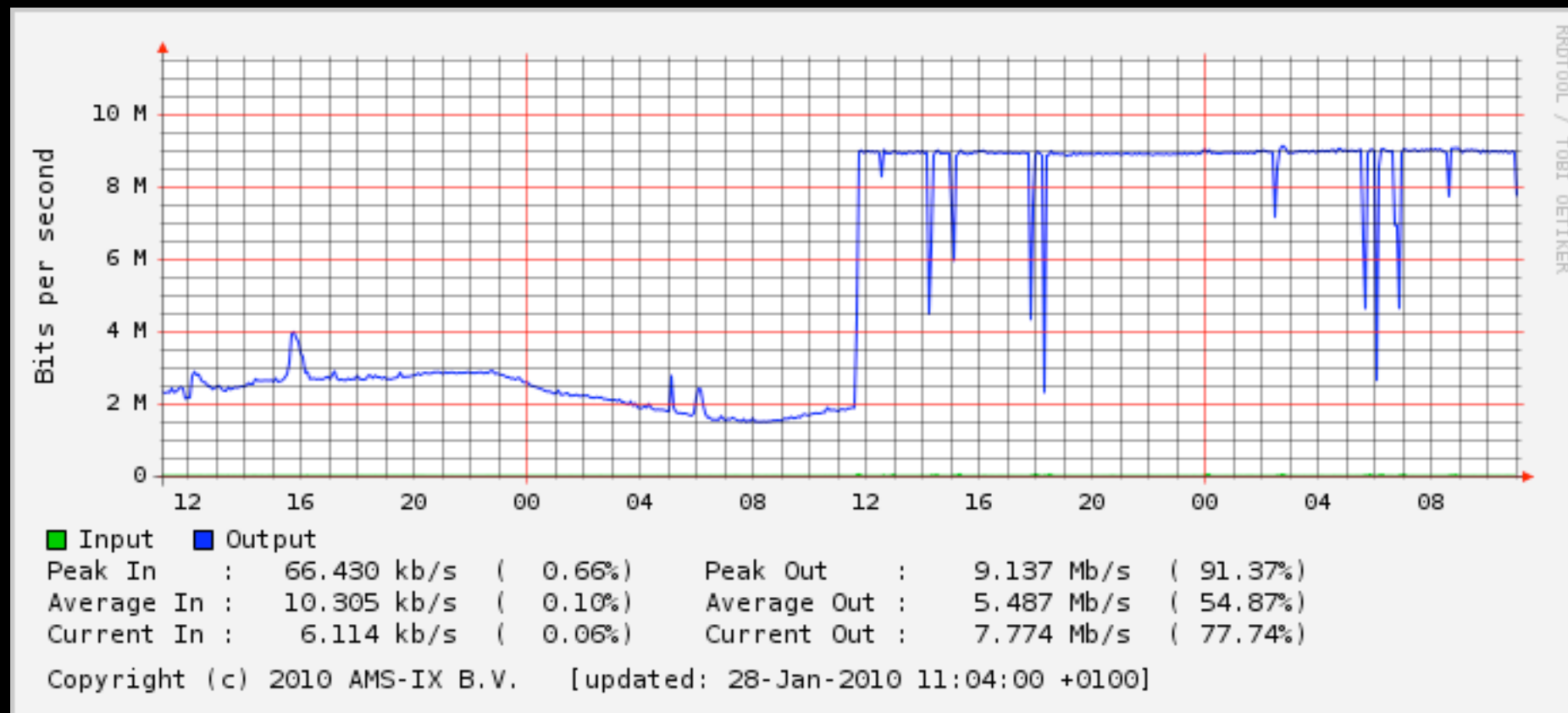
Came from ProXad network

33681368319  
37322719718  
33681368319  
37322719718  
33681368319  
37322719718  
33681368319  
33681368319

# The RIPE experiment

- 2010-01-27 they started announcing 1.1.1.0/24
- Only 10 MBit port
- It was maxed out immediately

# The RIPE experiment

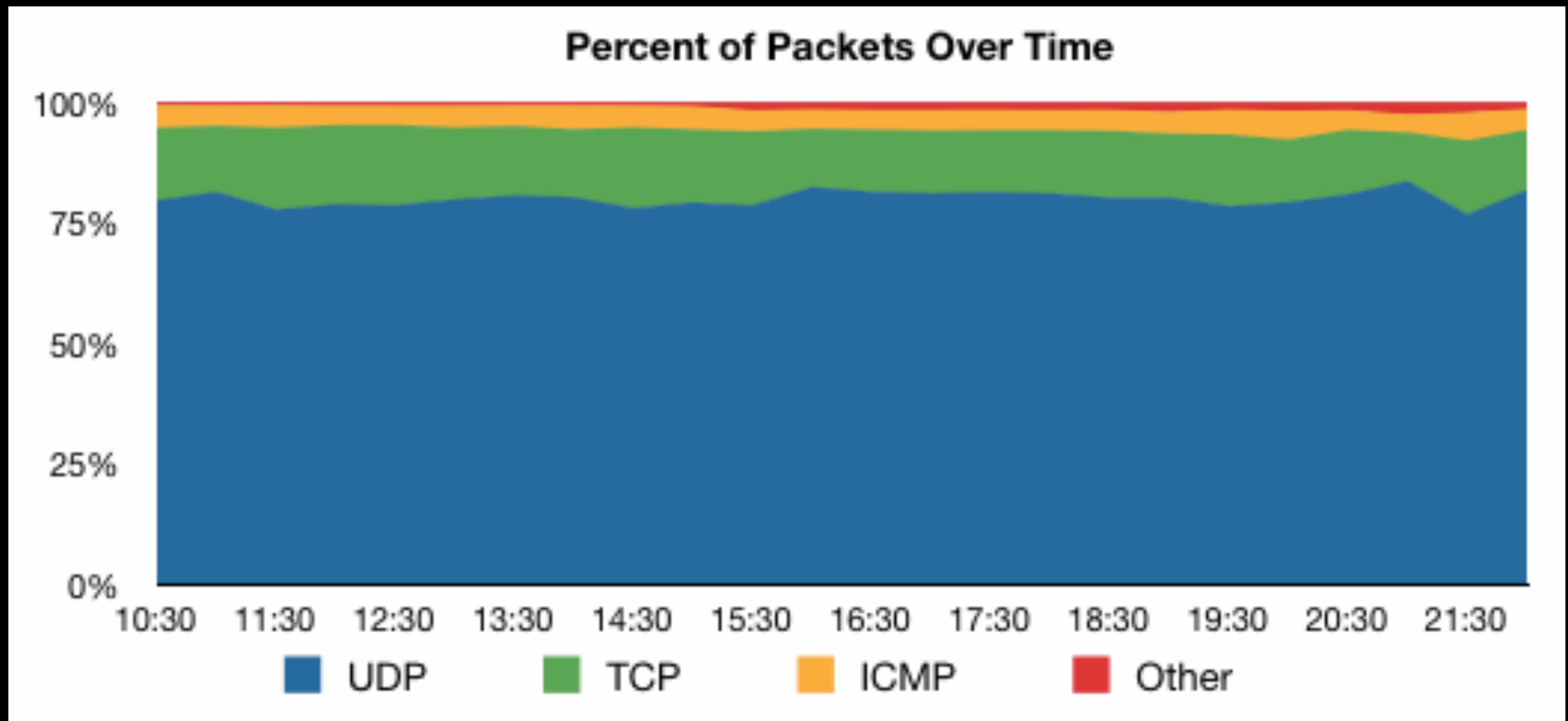


graph from RIPE blog

<http://labs.ripe.net/content/pollution-18>



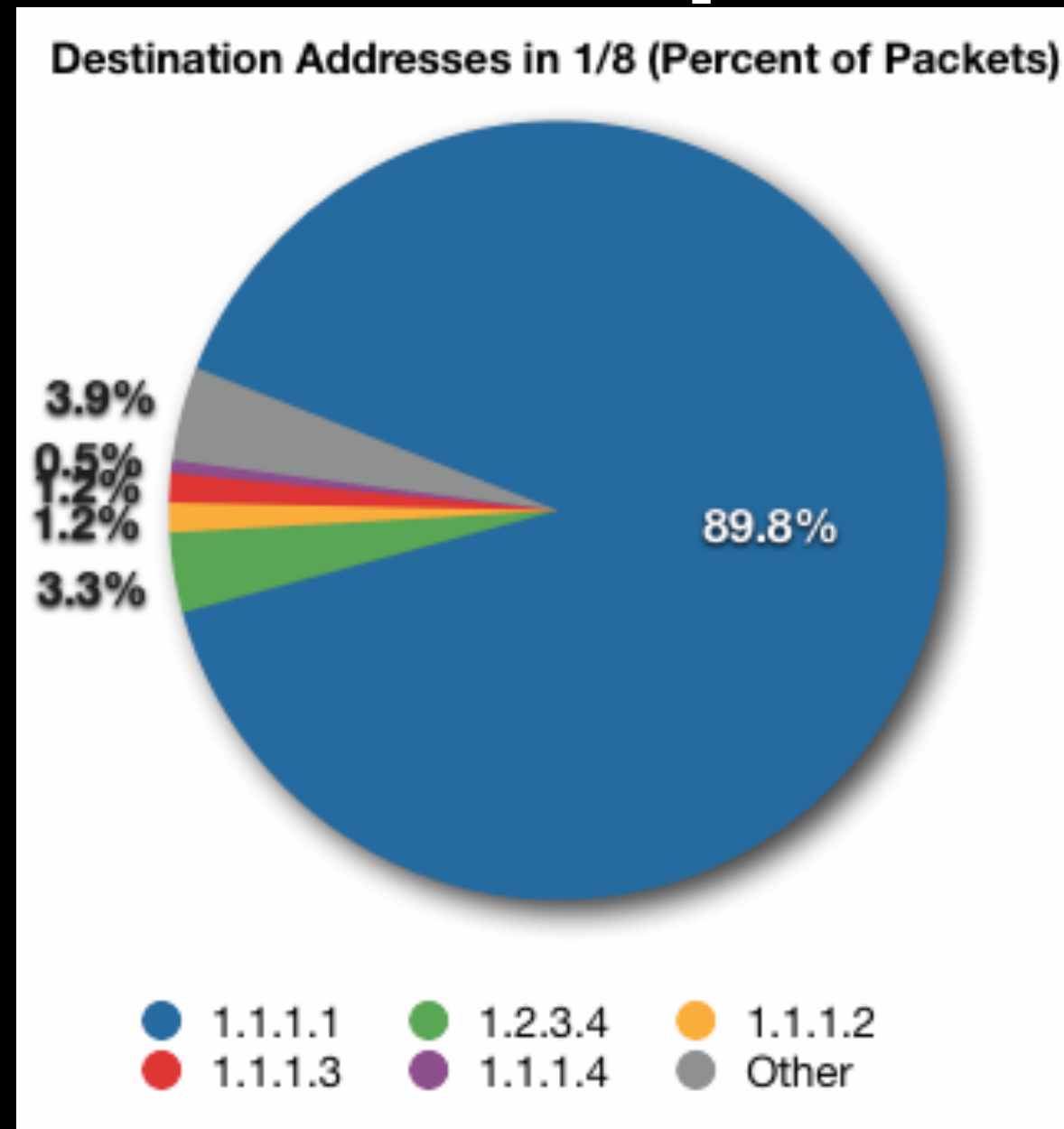
# The RIPE experiment



graph from RIPE blog

<http://labs.ripe.net/content/pollution-18>

# The RIPE experiment

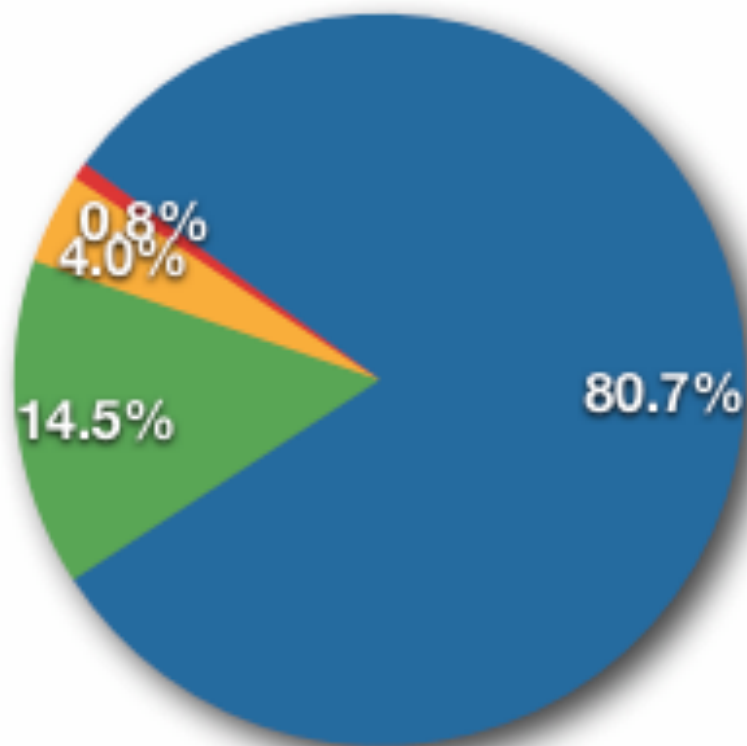


graph from RIPE blog

<http://labs.ripe.net/content/pollution-l8>

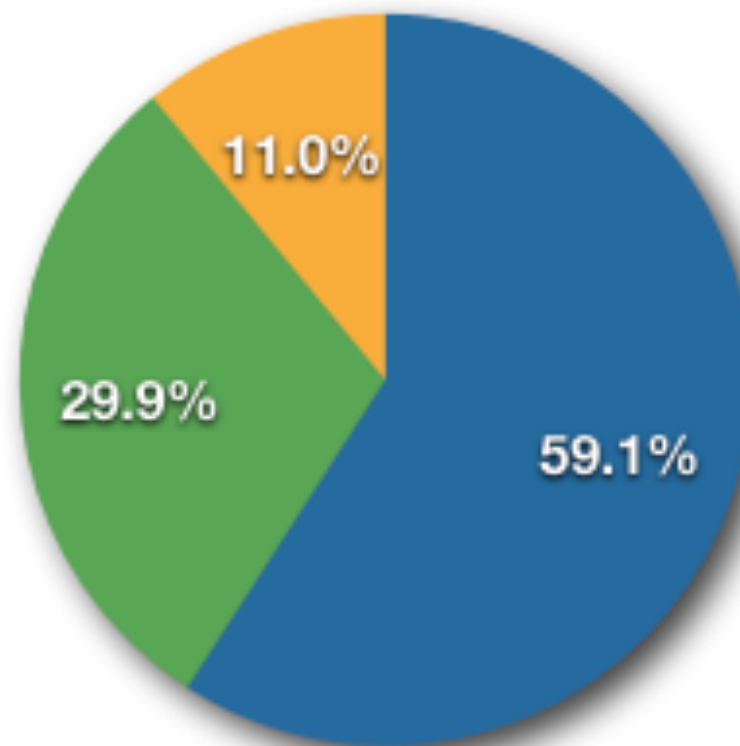
# The RIPE experiment

Traffic in 1/8



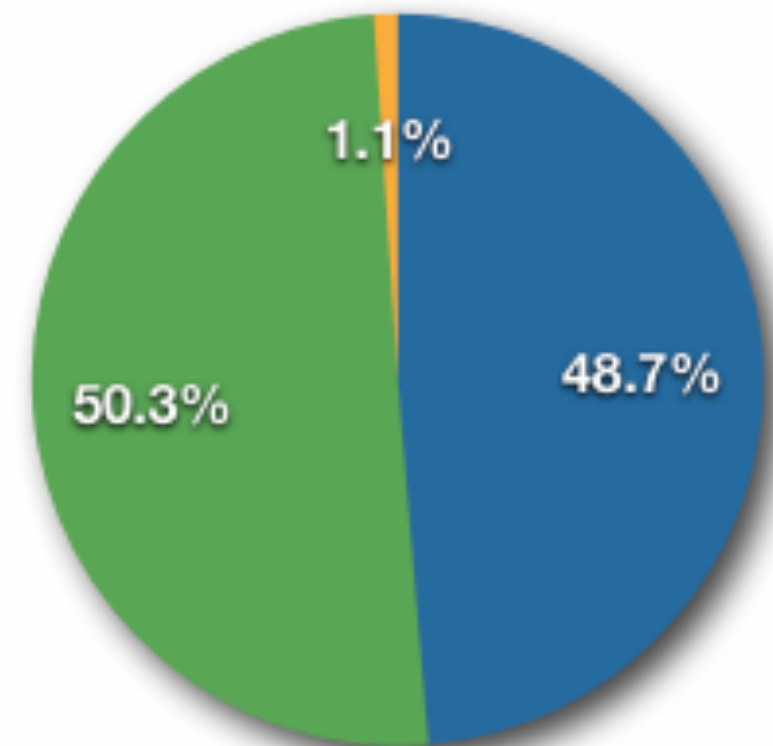
● UDP  
● ICMP Traffic  
● TCP  
● Other

UDP Traffic in 1/8



● Port 15206  
● Media Gateway Control Protocol  
● Other

TCP Traffic in 1/8



● Attempted HTTP connections  
● Other  
● "Established" HTTP connections

graph from RIPE blog

<http://labs.ripe.net/content/pollution-18>

# The RIPE experiment

*the part that i found interesting:*

“We found that almost **60%** of the **UDP packets** are sent towards the IP address **1.1.1.1** on **port 15206** which makes up the largest amount of packets seen by our RRC. Most of these packets **start their data section with 0x80**, continue with seemingly random data and are padded to 172 bytes with an (again seemingly random) 2 byte value. Some sources (<http://www.proxyblind.org/trojan.shtml>) list the port as being used by **a trojan called "KiLo"**, however information about it seem sparse.”

quoting the RIPE blog  
<http://labs.ripe.net/content/pollution-18>

# back in voiphun land

```
INVITE sip:011442083327467@re.pl.ac.ed SIP/2.0
Via: SIP/2.0/UDP 83.142.202.195:3058;branch=ca4b60ae7ba821fREPLACEDjrgrg;rport
From: <sip:sip@83.142.202.195>;tag=Za4b60aeREPLACED
To: <sip:011442083327467@re.pl.ac.ed>
Contact: <sip:sip@83.142.202.195>
Call-ID: 213948958-00227506489-384748@83.142.202.195
CSeq: 102 INVITE
User-Agent: Asterisk PBX
Max-Forwards: 70
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: 503
```

```
v=0
o=sip 2147483647 1 IN IP4 1.1.1.1
s=sip
c=IN IP4 1.1.1.1
t=0 0
m=audio 15206 RTP/AVP 10 4 3 0 8 112 5 7 18 111 101
a=rtpmap:10 L16/8000
a=rtpmap:4 G723/8000
a=fmtp:4 annexa=no
a=rtpmap:3 GSM/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:112 AAL2-G726-32/8000
a=rtpmap:5 DVI4/8000
a=rtpmap:7 LPC/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:111 G726-32/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - -
a=ptime:20
a=sendrecv
```

RTP Stream goes to IP 1.1.1.1

on port 15206

# RTP & SDP

- RTP (almost) always starts with an 0x80
- If an INVITE is accepted the RTP stream is sent to the IP in the SDP

# Conclusions and solutions?

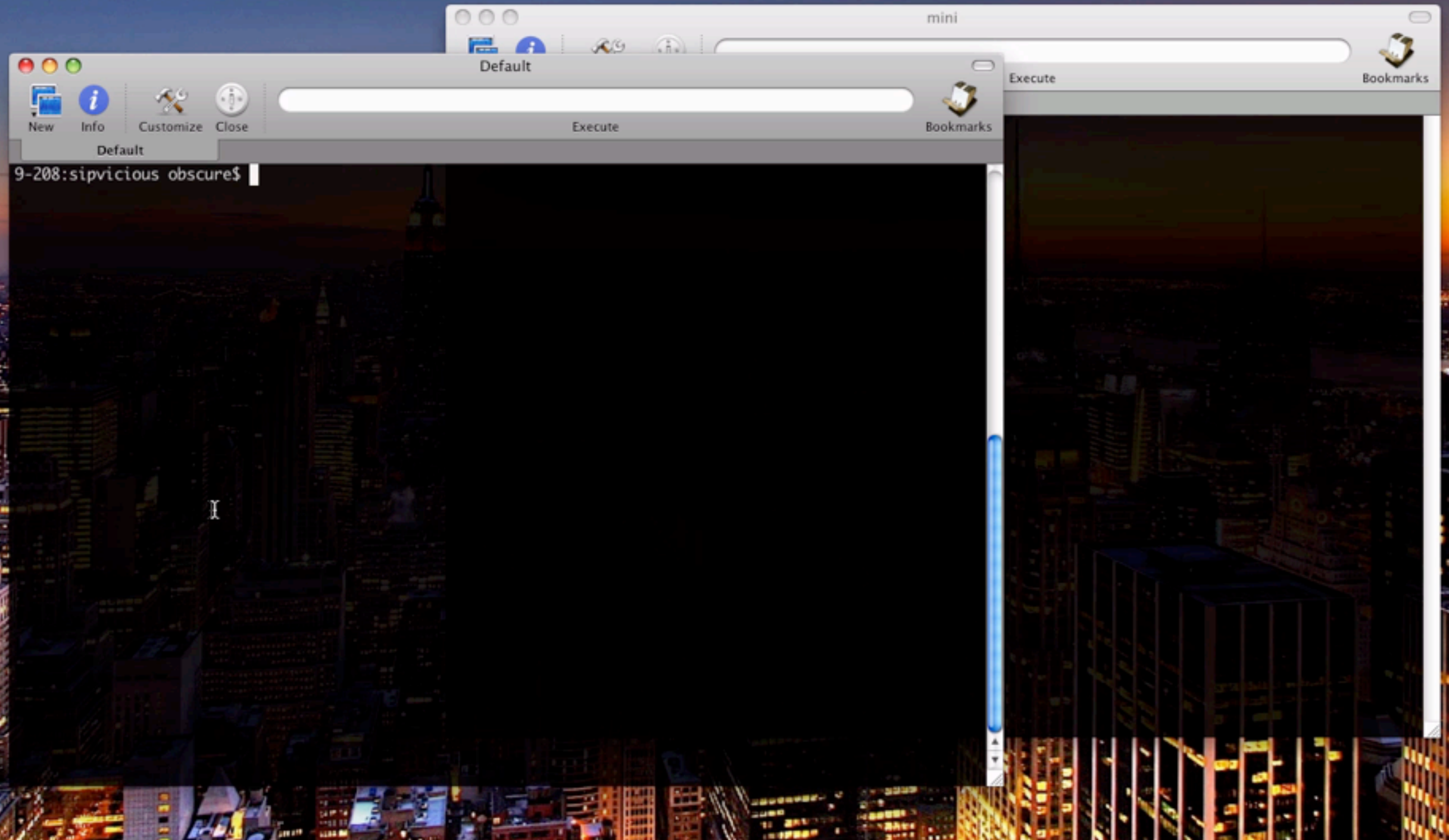
- These attacks are all internet borne
- Therefore put your Asterisk on the inside
- But ..

# .. this is not always possible

- ✓ Fail2ban + svcrash.py (comes with SIPVicious)
  - create exclusions for your providers!
- ✓ Responsive upstream provider
- ✓ Report to abuse
- ✓ Asterisk specific: enable alwaysauthreject



# svcrash demo



# Thanks

- John Todd and the Astricon team
- Sn0rky, Sjur & others who helped
- SIPVicious contributors and users

# More at..

- [EnableSecurity.com/research](http://EnableSecurity.com/research)
- [Sipvicious.org](http://Sipvicious.org)
- [VOIPSA.org](http://VOIPSA.org)

# Q.A

alternatively contact me

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