

# SIP session helper / ALG

## Starting @ 1:30pm

# Who am I ?

- David Attias
- Installing VoIP systems for over 11 years
- Owner of Penny Tone LLC
- Mikrotik user for 6 years
- Mikrotik Trainer  
MTCNA, MTCRE & MTCWE

# Purpose of this lecture

To inform Mikrotik users on the purpose and functions of SIP ALG.

# Agenda

- 1- What is ALG & what does it do.
- 2- The problem with VoIP and NAT
- 3- When is SIP ALG necessary and unnecessary?
- 4- How SIP ALG corrects problems.
- 5- Testing with Wireshark
- 6- SIP ALG Timeout
- 7- SIP ALG direct-media

# WHAT IS ALG?

# WHAT IS ALG?

- Application Layer Gateway
- A **G**ateway (firewall) that re-writes specific **A**pplication **L**ayer data fields.
- ALG is a firewall feature that rewrites Layer 7 data for specific applications.

# Keep in mind

- Only applies to NAT translation rules.
- NAT'ed devices are unaware that ALG is changing anything.
- Also known as:
  - NAT helper (Linux)
  - NAT session helper
  - SIP Transformations
  - Service ports

# The Problem with VoIP and NAT



# The Problem with VoIP and NAT

- SIP servers need to know the IP of all registered phones.
- Phones register their locally configured IP with the SIP server.
- If the phone and server are in the same network, no problems.
- If the phone is behind NAT and reports its IP to a remote server, the server responses will NOT be able to reach the phone.

# The Result

- Phone can not receive calls
- One way audio

# What ALG Does.

## What ALG Does.

- **ALG does exactly the same thing NAT translation does, but at layer 7**
- ALG intercepts the application messages before they leave the router
- Then inspects and replaces the “private client ip:port” with the “public ip:port” of the router (nat rule)

# What ALG Does



Dear SIP Server,  
I've been thinking about you and I want to  
INVITE you to SIP and RTP with me.  
Contact 192.168.20.100

# What ALG Does



Dear SIP Server,  
I've been thinking about you and I want to  
INVITE you to SIP and RTP with me.  
Contact ~~192.168.20.100~~

75.142.151.49

ALG WAS HERE

# Basic terms

# SIP and SDP

- SIP and SDP are VoIP Layer 7 protocols
- SIP – Session Initiated Protocol are commands exchanged between sip devices (register, invite, trying, hold, xfer, bye)
- SDP – Session Description Protocol is information about the audio (RTP) stream of a call.



# RouterOS SIP ALG settings

# RouterOS SIP ALG options

/ip firewall service-port

- Ports:**
- Remote Sip Server listening port. default values are 5060,5061
  - Applies to TCP and UDP
  - Single port, no ranges
  - Up to 8 entries

- Sip-direct-media**
- Allows a redirect of the RTP media stream to go directly from sip device to sip device
  - Default value is yes.

- Timeout:**
- Sets the sip UDP timeout in connection tracker.
  - Default is 1 hour

Mikrotik CLI

/ip firewall service-port

```
set sip ports=5060,5061 sip-direct-media=yes sip-timeout=01:00:00 disabled=no
```

# How does ALG correct SIP problems?

# How does ALG correct SIP problems?

- By replacing specific private IP:port with router's wan side IP:port

ALG changes:

SIP headers: Via, Contact

SDP Body: m= o= c=

- ALG makes changes to Layer 7 data transparently as it passes through the NAT rule.

# **Layer 7 data before and after (with ALG enabled)**

# Layer 7 Data with before ALG

## SIP REGISTER message **BEFORE** ALG modification:

REGISTER sip:207.252.1.148 SIP/2.0

**Via:** SIP/2.0/UDP **192.168.20.100:5060**;branch=z9hG4bK-8fb0e171

From: "David Attias" <sip:201525@207.252.1.148>;tag=191914b06bec0

To: "David Attias" <sip:201525@207.252.1.148>

Call-ID: 6894e30c-h1c8d357@192.168.20.100

CSeq: 1373 REGISTER

Max-Forwards: 70

**Contact:** "David Attias" <sip:201525@**192.168.20.100:5060**>;expires=3600

User-Agent: Cisco/SPA504G-7.6.2b

Content-Length: 0

Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER, UPDATE

Supported: replaces



The fields ALG  
will change

# Layer 7 Data with after ALG

SIP REGISTER message **AFTER** ALG modification:

REGISTER sip:207.252.1.148 SIP/2.0

**Via:** SIP/2.0/UDP **75.142.151.49:1024**;branch=z9hG4bK-8fb0e171

From: "David Attias" <sip:525@207.252.1.148>;tag=191914b06bec0

To: "David Attias" <sip:525@207.252.1.148>

Call-ID: 6894e30c-h1c8d357@192.168.20.100

CSeq: 1373 REGISTER

Max-Forwards: 70

**Contact:** "David Attias" <sip:525@**75.142.151.49:1024**>;expires=3600

User-Agent: Cisco/SPA504G-7.6.2b

Content-Length: 0

Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER, UPDATE

Supported: replaces



After ALG  
The "respond to"  
IP and port

# Layer 7 Data with before ALG

```
INVITE sip:*98@207.252.1.148 SIP/2.0
Via: SIP/2.0/UDP 192.168.20.100:5060;branch=z9hG4bK-7badf56d
From: "David Attias" <sip:525@207.252.1.148>;tag=f95367fa52060ce5o0
To: "Voice Mail" <sip:*98@207.252.1.148>
Call-ID: 9c8a315e-419d32d1@192.168.20.100
CSeq: 101 INVITE
Max-Forwards: 70
Contact: "David Attias" <sip:525@192.168.20.100:5060>
Expires: 240
User-Agent: Cisco/SPA504G-7.6.2b
Content-Length: 397
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER, UPDATE
Supported: replaces
Content-Type: application/sdp
```

SIP Headers

Before ALG  
modifies layer 7  
data

```
v=0
o= 176664 176664 IN IP4 192.168.20.100
s=-
c=IN IP4 192.168.20.100
t=0 0
m=audio 14254 RTP/AVP 0 2 8 9 18 96 97 98 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=rtpmap:2 G726-32/8000
a=fmtp:101 0-15
a=ptime:30
a=sendrecv
```

SDP Body



# Layer 7 Data with after ALG

```
INVITE sip:*98@207.252.1.148 SIP/2.0
Via: SIP/2.0/UDP 75.142.151.49:1024;branch=z9hG4bK-7badf56d
From: "David Attias" <sip:525@207.252.1.148>;tag=f95367fa52060ce5o0
To: "Voice Mail" <sip:*98@207.252.1.148>
Call-ID: 9c8a315e-419d32d1@192.168.20.100
CSeq: 101 INVITE
Max-Forwards: 70
Contact: "David Attias" <sip:525@75.142.151.49:1024>
Expires: 240
User-Agent: Cisco/SPA504G-7.6.2b
Content-Length: 397
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER, UPDATE
Supported: replaces
Content-Type: application/sdp
```

SIP Headers

After ALG  
modifies layer 7  
data

```
v=0
o= 176664 176664 IN IP4 75.142.151.49
s=-
c=IN IP4 75.142.151.49
t=0 0
m=audio 19032 RTP/AVP 0 2 8 9 18 96 97 98 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=rtpmap:2 G726-32/8000
a=fmtp:101 0-15
a=ptime:30
a=sendrecv
```

SDP Body

# When is SIP ALG necessary?

# When is SIP ALG necessary?

When the SIP device behind NAT is NOT NAT aware

# SIP devices that are not “NAT Aware”

- Some SIP devices are not NAT aware and write their (private) device IP in layer 7 messages to the server.
- The remote SIP server receives the layer 7 message which specifies a private reply address.
- The server sends replies to the private address, which can never be reached.

# SIP servers that are not “NAT Aware”

\*\*\*The public server can receive data, but reply packets are dropped.\*\*\*

Example of a SIP device that is not NAT aware



Private / NAT

X100  
192.168.20.100





## Private / NAT

X100  
192.168.20.100



SIP headers  
Via: 192.168.20.100  
Contact: 192.168.20.100

SDP Body  
o = IN IP4 192.168.20.100  
c = IN IP4 192.168.20.100  
m = audio 19032





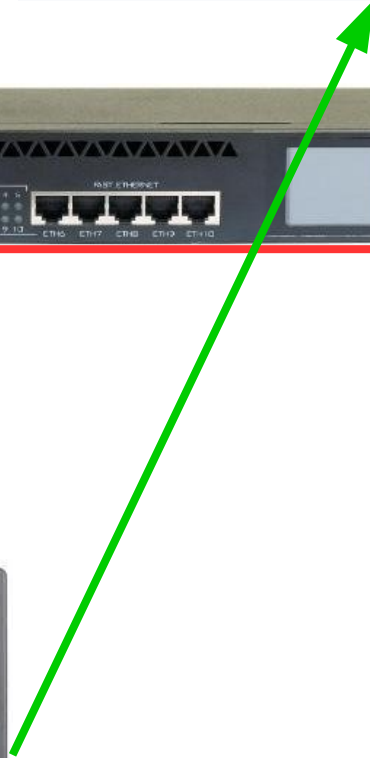
SIP headers  
Via: 192.168.20.100  
Contact: 192.168.20.100

SDP Body  
o = IN IP4 192.168.20.100  
c = IN IP4 192.168.20.100  
m = audio 19032



Private / NAT

X100  
192.168.20.100





SIP headers  
Via: 192.168.20.100  
Contact: 192.168.20.100

SDP Body  
o = IN IP4 192.168.20.100  
c = IN IP4 192.168.20.100  
m = audio 19032

ALG IS required here



Private / NAT

X100  
192.168.20.100



With RouterOS SIP ALG enabled



WAN 75.142.151.49



Private / NAT

X100  
192.168.20.100



SIP headers  
Via: 192.168.20.100  
Contact: 192.168.20.100

SDP Body  
o = IN IP4 192.168.20.100  
c = IN IP4 192.168.20.100  
m = audio 19032



SIP headers  
Via: 75.142.151.49  
Contact: 75.142.151.49

SDP Body  
o = IN IP4 75.142.151.49  
c = IN IP4 75.142.151.49  
m = audio 19032

ALG Enabled

WAN 75.142.151.49



Private / NAT

X100  
192.168.20.100

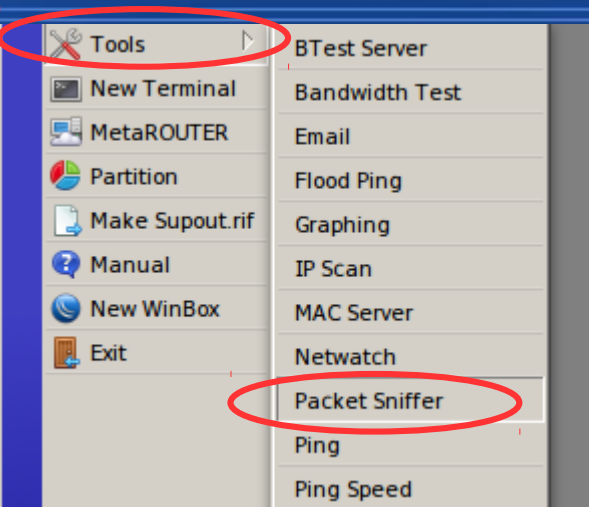


# Is your SIP device NAT Aware?

- Packet capture in routerOS
  - Capture packets before and after they get modified by ALG
- Decode the capture files in Wireshark

# Setting up the packet capture

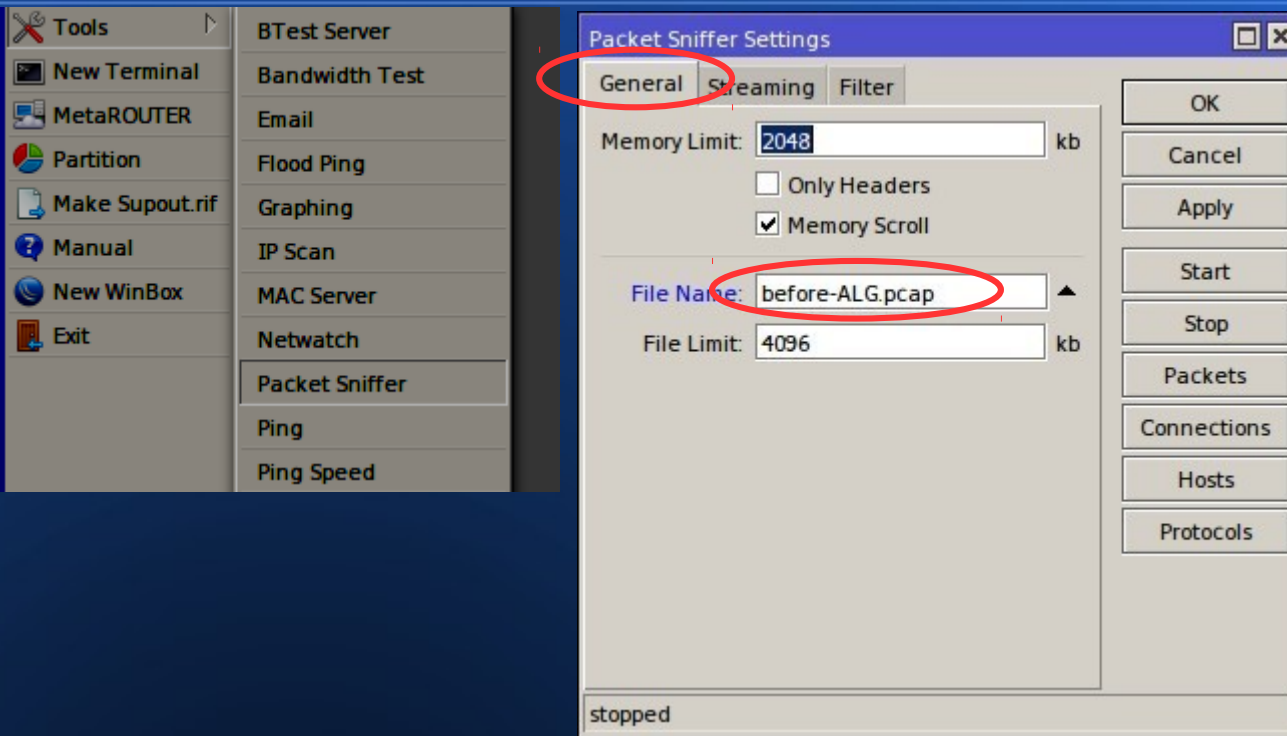
# Before ALG modifications pcap



/tool sniffer



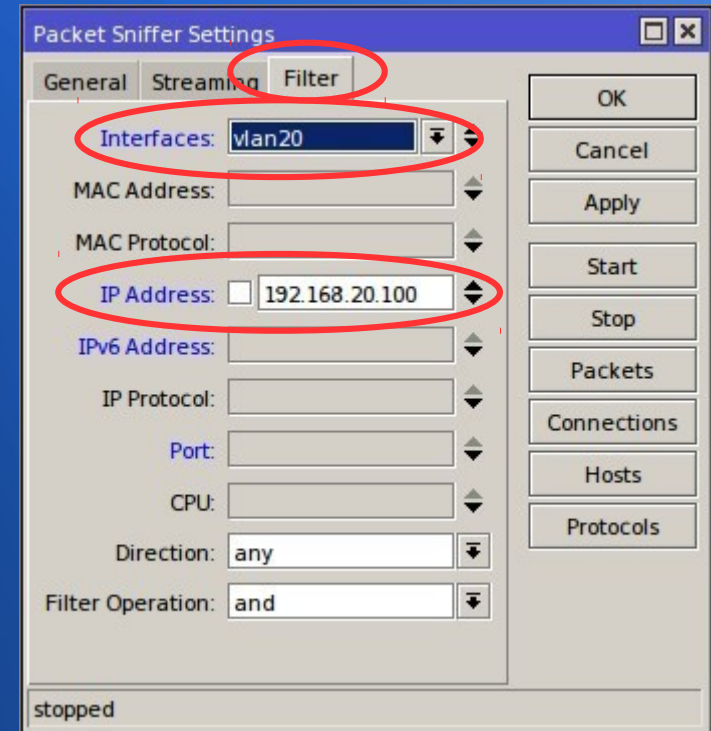
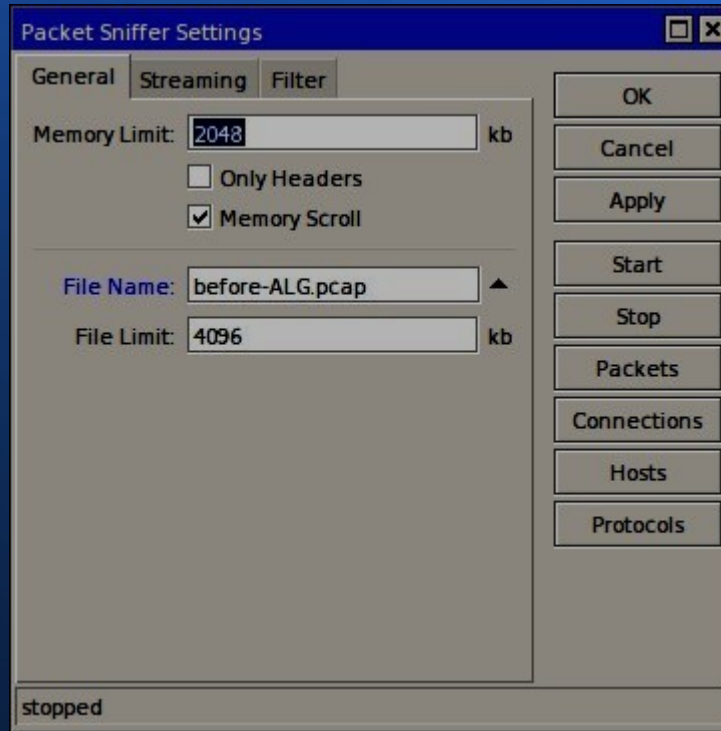
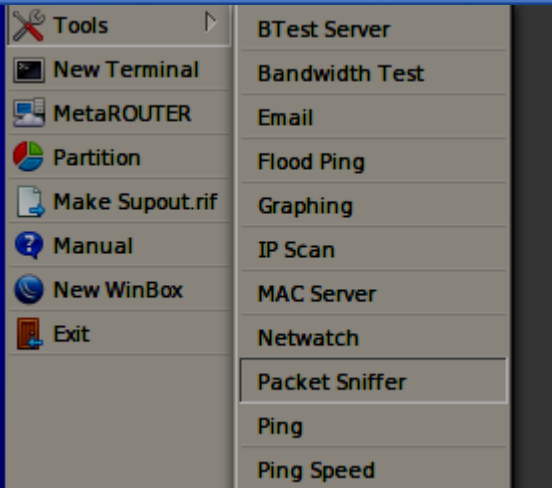
# Before ALG modifications pcap



```
/tool sniffer
```

```
set only-headers=no file-name=before-ALG.pcap file-limit=4096
```

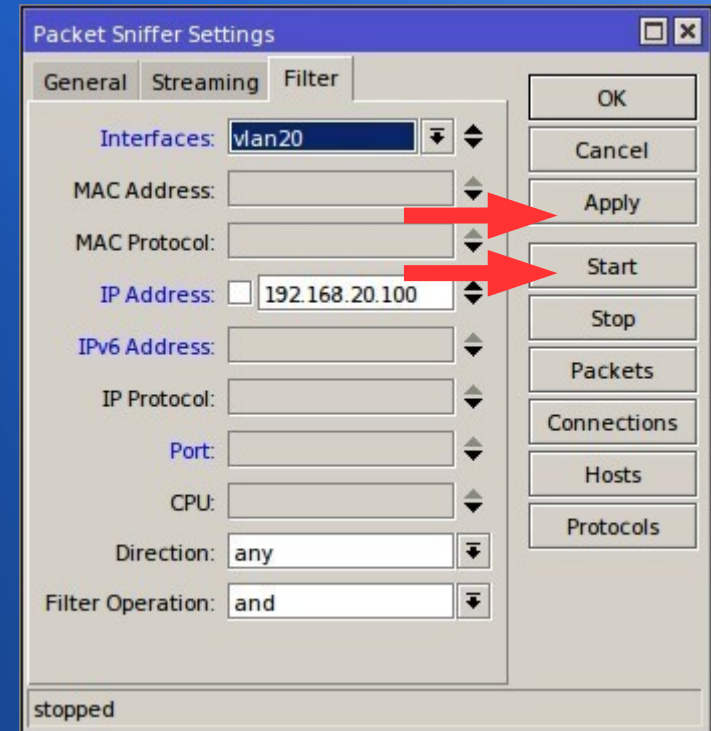
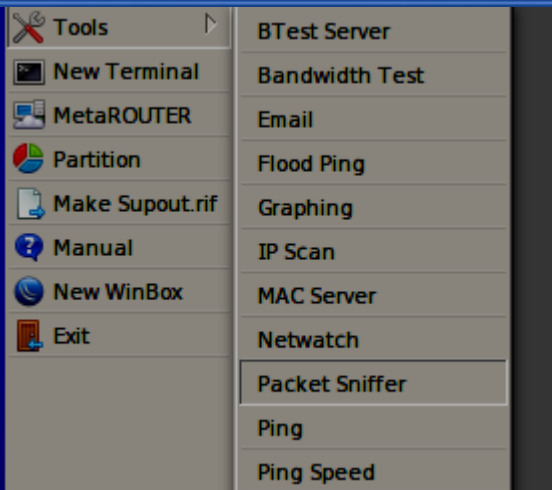
# Before ALG modifications pcap



```
/tool sniffer
```

```
set only-headers=no file-name=before-ALG.pcap file-limit=4096 filter-interface=vlan20 filter-  
ip-address=192.168.20.100/32 filter-direction=any
```

# Before ALG modifications pcap



```
/tool sniffer
```

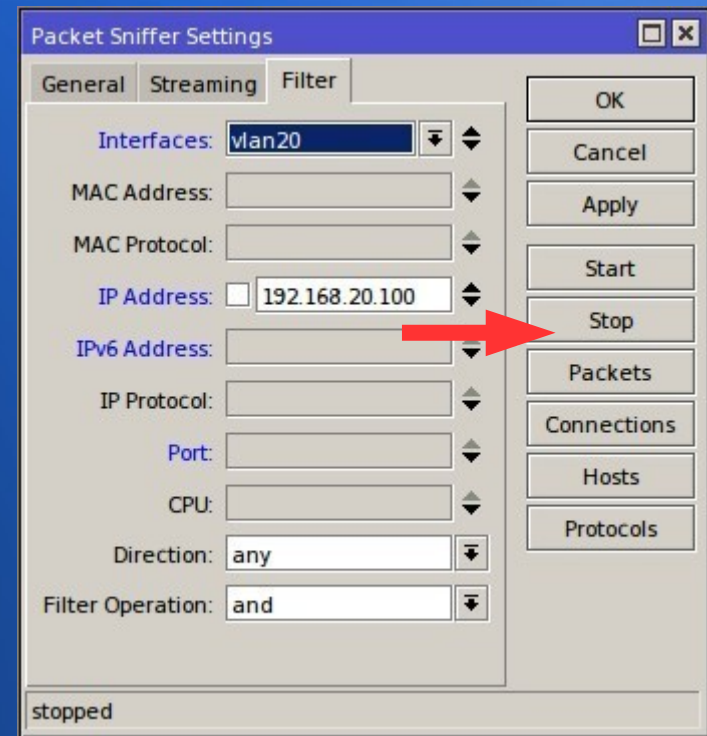
```
set only-headers=no file-name=before-ALG.pcap file-limit=4096 filter-interface=vlan20 filter-  
ip-address=192.168.20.100/32 filter-direction=any
```

```
start
```

Generate some traffic while the sniffer is capturing packets.

# Before ALG modifications pcap

Make sure to stop the sniffer



```
/tool sniffer
```

```
stop
```

# Download pcap files

The screenshot shows the Mikrotik WinBox interface. The top navigation bar includes 'Session', 'Settings', and 'Dashboard'. Below this, there are navigation buttons and a 'Safe Mode' indicator. The 'Session' field displays the IP address '192.168.88.1'. On the left sidebar, the 'Files' menu item is circled in red. The main window displays a 'File List' dialog with a table of files. A red arrow points to the file 'before-ALG.pcap' in the table.

File Name	Type	Size	Creation Time
before-ALG.pcap	.pcap file	6.5 KiB	May/11/2017 15:13:01

1 item | 19.7 MiB of 128.0 MiB used | 84% free

# Decode in wireshark

before-ALG.pcap

File Edit View Go Capture Analyze Statistics Telephony Wireless Tools Help

Apply a display filter ... <Ctrl-/> Expression... +

No.	Time	Source	Destination	Protocol	Length	Info
1	0.000000	192.168.20.1	192.168.20.100	DHCP	357	DHCP Offer - Transaction ID 0xbed323ae
2	1.003062	192.168.20.1	192.168.20.100	DHCP	357	DHCP ACK - Transaction ID 0xbed323ae
3	1.688359	192.168.20.100	192.168.20.1	DNS	72	Standard query 0x0001 A pool.ntp.org
4	1.702269	192.168.20.1	192.168.20.100	DNS	136	Standard query response 0x0001 A pool.ntp.org A...
5	1.703640	192.168.20.100	208.75.89.4	NTP	90	NTP Version 3, client
6	1.706975	192.168.20.100	192.168.20.1	TFTP	93	Read Request, File: SEPECE1A9CDAA7D.cnf.xml, Tr...
7	1.707186	192.168.20.1	192.168.20.100	ICMP	121	Destination unreachable (Port unreachable)
8	1.708256	192.168.20.100	192.168.1.254	TCP	74	1024 → 80 [SYN] Seq=0 Win=8192 Len=0 MSS=1456 W...
9	1.726550	208.75.89.4	192.168.20.100	NTP	90	NTP Version 3, server
10	2.061140	192.168.20.100	224.168.168.168	IGMPv2	56	Membership Report group 224.168.168.168
11	2.087226	192.168.20.100	207.252.1.148	SIP	524	Request: REGISTER sip:207.252.1.148 (1 binding...
12	2.088547	207.252.1.148	192.168.20.100	SIP	580	Status: 401 Unauthorized
13	2.293830	192.168.20.100	207.252.1.148	SIP	678	Request: REGISTER sip:207.252.1.148 (1 binding...
14	2.296420	207.252.1.148	192.168.20.100	SIP	601	Request: OPTIONS sip:100@192.168.20.100:5060
15	2.296603	207.252.1.148	192.168.20.100	SIP	599	Status: 200 OK (1 binding)
16	2.297538	207.252.1.148	192.168.20.100	SIP	605	Request: NOTIFY sip:100@192.168.20.100:5060
17	2.341624	192.168.20.100	207.252.1.148	SIP	460	Status: 200 OK
18	2.352654	192.168.20.100	207.252.1.148	SIP	367	Status: 200 OK
19	2.648746	192.168.20.100	224.168.168.168	IGMPv2	56	Membership Report group 224.168.168.168
20	5.308720	192.168.20.100	255.255.255.255	UDP	70	55656 → 55656 Len=28
21	6.708057	192.168.20.100	192.168.20.1	TFTP	93	Read Request, File: SEPECE1A9CDAA7D.cnf.xml, Tr...
22	6.708229	192.168.20.1	192.168.20.100	ICMP	121	Destination unreachable (Port unreachable)
23	7.348782	192.168.20.100	192.168.1.254	TCP	74	[TCP Retransmission] 1024 → 80 [SYN] Seq=0 Win=...

▶ Frame 19: 56 bytes on wire (448 bits), 56 bytes captured (448 bits)

▶ Ethernet II, Src: CiscoInc\_cd:aa:7d (ec:e1:a9:cd:aa:7d), Dst: IPv4mcast\_28:a8:a8 (01:00:5e:28:a8:a8)

▶ Internet Protocol Version 4, Src: 192.168.20.100, Dst: 224.168.168.168

▶ Internet Group Management Protocol

before-ALG Packets: 23 · Displayed: 23 (100.0%) · Load time: 0:0.0 Profile: Default

# Decode in wireshark

The screenshot shows the Wireshark interface with a packet capture named 'before-ALG.pcap'. The display filter is set to 'sip'. The packet list pane shows several SIP packets, with packet 18 selected. The packet details pane shows the structure of the selected packet:

No.	Time	Source	Destination	Protocol	Length	Info
11	2.087226	192.168.20.100	207.252.1.148	SIP	524	Request: REGISTER sip:207.252.1.148 (1 binding...
12	2.088547	207.252.1.148	192.168.20.100	SIP	580	Status: 401 Unauthorized
13	2.293830	192.168.20.100	207.252.1.148	SIP	678	Request: REGISTER sip:207.252.1.148 (1 binding...
14	2.296420	207.252.1.148	192.168.20.100	SIP	601	Request: OPTIONS sip:100@192.168.20.100:5060
15	2.296603	207.252.1.148	192.168.20.100	SIP	599	Status: 200 OK (1 binding)
16	2.297538	207.252.1.148	192.168.20.100	SIP	605	Request: NOTIFY sip:100@192.168.20.100:5060
17	2.341624	192.168.20.100	207.252.1.148	SIP	460	Status: 200 OK
18	2.352654	192.168.20.100	207.252.1.148	SIP	367	Status: 200 OK

Packet 18 details:

- Frame 18: 367 bytes on wire (2936 bits), 367 bytes captured (2936 bits)
- Ethernet II, Src: CiscoInc\_cd:aa:7d (ec:e1:a9:cd:aa:7d), Dst: Routerbo\_c3:62:47 (4c:5e:0c:c3:62:47)
- Internet Protocol Version 4, Src: 192.168.20.100, Dst: 207.252.1.148
- User Datagram Protocol, Src Port: 5060 (5060), Dst Port: 5060 (5060)
- Session Initiation Protocol (200)

Session Initiation Protocol: Protocol | Packets: 23 · Displayed: 8 (34.8%) · Load time: 0:0.3 · Profile: Default



# Decode in wireshark

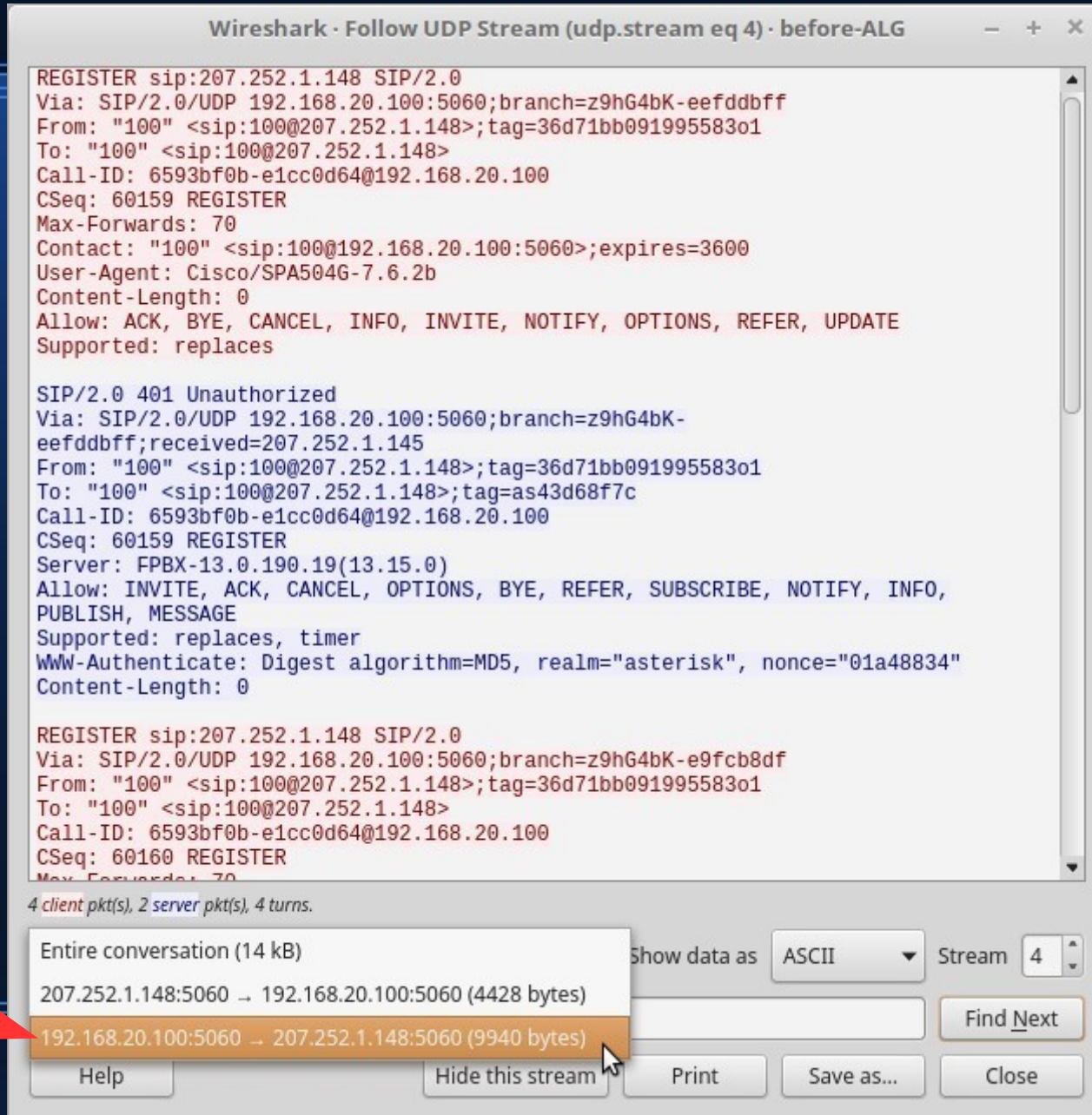
The image shows the Wireshark interface with a packet capture named 'before-ALG.pcap'. The filter bar is set to 'sip'. Packet 11 is selected, and a context menu is open over it. A red arrow points to the 'Follow' option in the menu, and another red arrow points to the 'UDP Stream' option in the sub-menu.

No.	Time	Source	Destination	Protocol	Length	Info
11	2.087226	192.168.20.100	207.252.1.148	SIP	524	Request: REGISTER sip:207.252.1.148 (1 binding...
12	2.088547	207.252.1.148	192.168.20.100			252.1.148 (1 binding...
13	2.293830	192.168.20.100	207.252.1.148			192.168.20.100:5060
14	2.296420	207.252.1.148	192.168.20.100			
15	2.296603	207.252.1.148	192.168.20.100			2.168.20.100:5060
16	2.297538	207.252.1.148	192.168.20.100			
17	2.341624	192.168.20.100	207.252.1.148			
18	2.352654	192.168.20.100	207.252.1.148			

Frame 11: 524 bytes on wire (4192 bits), 524 bytes captured  
Ethernet II, Src: CiscoInc\_cd:aa:7d (ec:e1:a9:cd:aa:7d), Dst:  
Internet Protocol Version 4, Src: 192.168.20.100, Dst: 207.  
User Datagram Protocol, Src Port: 5060 (5060), Dst Port: 50  
Session Initiation Protocol (REGISTER)

Session Initiation Protocol: Protocol      Packets: 23 · Displayed: 8 (34.8%) · Load time: 0:0.3 · Profile: Default

# Decode in wireshark



Wireshark - Follow UDP Stream (udp.stream eq 4) - before-ALG

```
REGISTER sip:207.252.1.148 SIP/2.0
Via: SIP/2.0/UDP 192.168.20.100:5060;branch=z9hG4bK-eefddbff
From: "100" <sip:100@207.252.1.148>;tag=36d71bb091995583o1
To: "100" <sip:100@207.252.1.148>
Call-ID: 6593bf0b-e1cc0d64@192.168.20.100
CSeq: 60159 REGISTER
Max-Forwards: 70
Contact: "100" <sip:100@192.168.20.100:5060>;expires=3600
User-Agent: Cisco/SPA504G-7.6.2b
Content-Length: 0
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER, UPDATE
Supported: replaces

SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP 192.168.20.100:5060;branch=z9hG4bK-
eefddbff;received=207.252.1.145
From: "100" <sip:100@207.252.1.148>;tag=36d71bb091995583o1
To: "100" <sip:100@207.252.1.148>;tag=as43d68f7c
Call-ID: 6593bf0b-e1cc0d64@192.168.20.100
CSeq: 60159 REGISTER
Server: FPBX-13.0.190.19(13.15.0)
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO,
PUBLISH, MESSAGE
Supported: replaces, timer
WWW-Authenticate: Digest algorithm=MD5, realm="asterisk", nonce="01a48834"
Content-Length: 0

REGISTER sip:207.252.1.148 SIP/2.0
Via: SIP/2.0/UDP 192.168.20.100:5060;branch=z9hG4bK-e9fcb8df
From: "100" <sip:100@207.252.1.148>;tag=36d71bb091995583o1
To: "100" <sip:100@207.252.1.148>
Call-ID: 6593bf0b-e1cc0d64@192.168.20.100
CSeq: 60160 REGISTER
Max-Forwards: 70
```

4 client pkt(s), 2 server pkt(s), 4 turns.

Entire conversation (14 kB)  
207.252.1.148:5060 → 192.168.20.100:5060 (4428 bytes)  
192.168.20.100:5060 → 207.252.1.148:5060 (9940 bytes)

Show data as ASCII Stream 4 Find Next

Help Hide this stream Print Save as... Close

# Decode in wireshark

Wireshark - Follow UDP Stream (udp.stream eq 4) - before-ALG

```
REGISTER sip:207.252.1.148 SIP/2.0
Via: SIP/2.0/UDP 192.168.20.100:5060;branch=z9hG4bK-eefddbff
From: "100" <sip:100@207.252.1.148>;tag=36d71bb091995583o1
To: "100" <sip:100@207.252.1.148>
Call-ID: 6593bf0b-e1cc0d64@192.168.20.100
CSeq: 60159 REGISTER
Max-Forwards: 70
Contact: "100" <sip:100@192.168.20.100:5060>;expires=3600
User-Agent: Cisco/SPA504G-7.6.2b
Content-Length: 0
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER, UPDATE
Supported: replaces

REGISTER sip:207.252.1.148 SIP/2.0
Via: SIP/2.0/UDP 192.168.20.100:5060;branch=z9hG4bK-e9fcb8df
From: "100" <sip:100@207.252.1.148>;tag=36d71bb091995583o1
To: "100" <sip:100@207.252.1.148>
Call-ID: 6593bf0b-e1cc0d64@192.168.20.100
CSeq: 60160 REGISTER
Max-Forwards: 70
Authorization: Digest username="100",realm="asterisk",nonce="01a48834",uri="sip:
207.252.1.148",algorithm=MD5,response="36cd61d13d1c58ed830cf64e3b47b82a"
Contact: "100" <sip:100@192.168.20.100:5060>;expires=3600
User-Agent: Cisco/SPA504G-7.6.2b
Content-Length: 0
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER, UPDATE
Supported: replaces
```

4 client pkt(s), 0 server pkt(s), 0 turns.

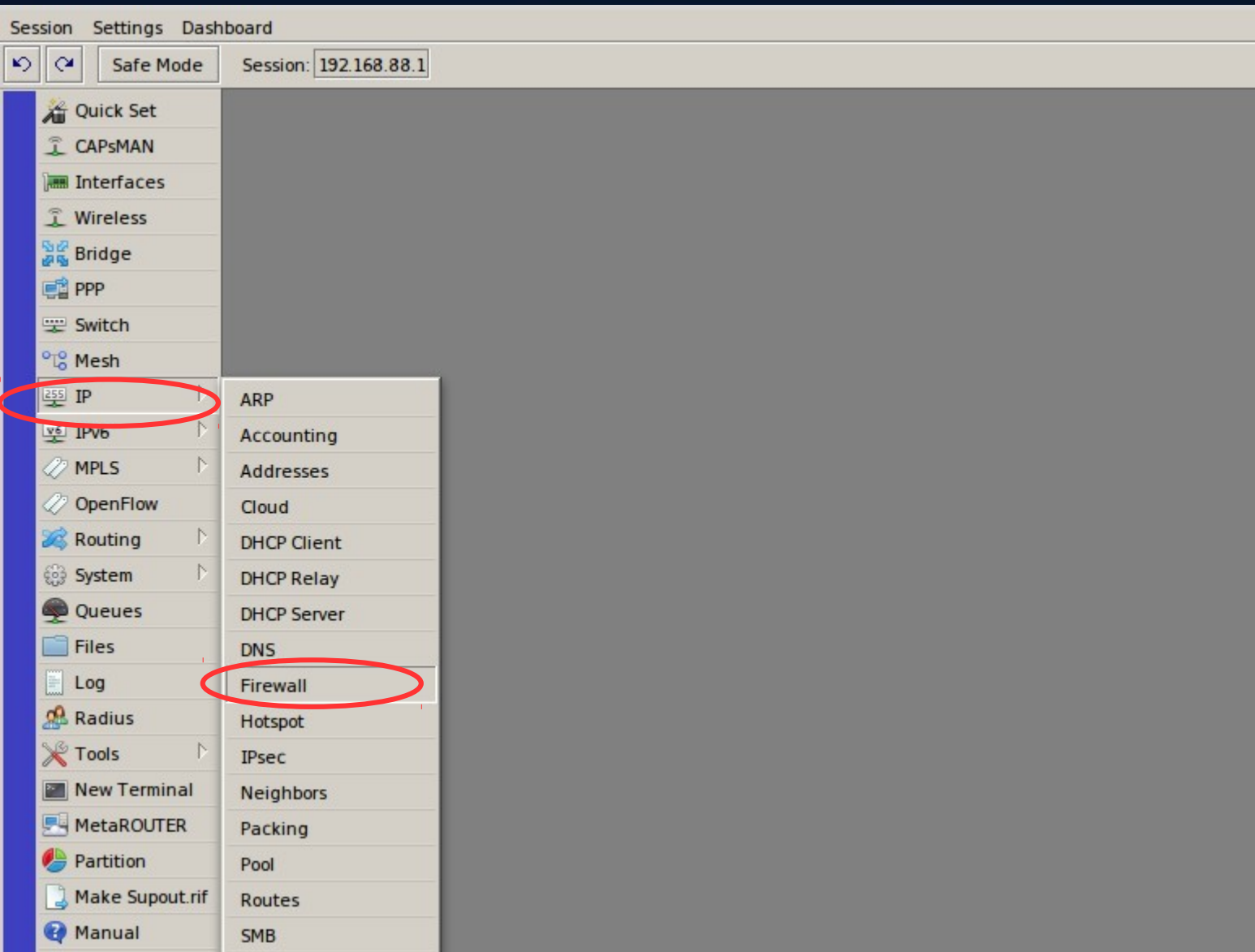
192.168.20.100:5060 → 207.252.1.148:5060 (9940 by) Show data as ASCII Stream 4

Find:  Find Next

Help Hide this stream Print Save as... Close

If your SIP device is not NAT Aware  
Enable SIP ALG !

# Enable SIP ALG



# Enable SIP ALG

Session Settings Dashboard  
Safe Mode Session: 192.168.88.1

Quick Set  
CAPsMAN  
Interfaces  
Wireless  
Bridge  
PPP  
Switch  
Mesh  
IP  
IPv6  
MPLS  
OpenFlow  
Routing  
System  
Queues  
Files  
Log  
Radius  
Tools  
New Terminal  
MetaROUTER  
Partition  
Make Supout.rif  
Manual  
New WinBox

ARP  
Accounting  
Addresses  
Cloud  
DHCP Client  
DHCP Relay  
DHCP Server  
DNS  
Firewall  
Hotspot  
IPsec  
Neighbors  
Packing  
Pool  
Routes  
SMB  
SNMP

Firewall  
Filter Rules NAT Mangle Raw Service Ports Connections Add

Name	Ports	SIP Direct Media	SIP Timeout
● dccp			
● ftp	21		
● h323			
● irc	6667		
● pptp			
● sctp			
● sip	5060	yes	01-00-10
● tftp	69		
● udplite			

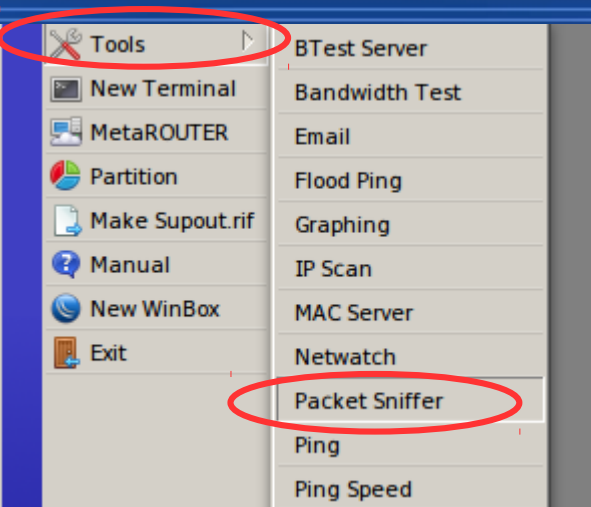
9 items (1 selected)

- Show Categories
- Detail Mode
- Show Columns
- Find Ctrl+F
- Find Next Ctrl+G
- Select All Ctrl+A
- Enable Ctrl+E**
- Disable Ctrl+D

/ip firewall service-port  
enable sip

capture packets after ALG modification

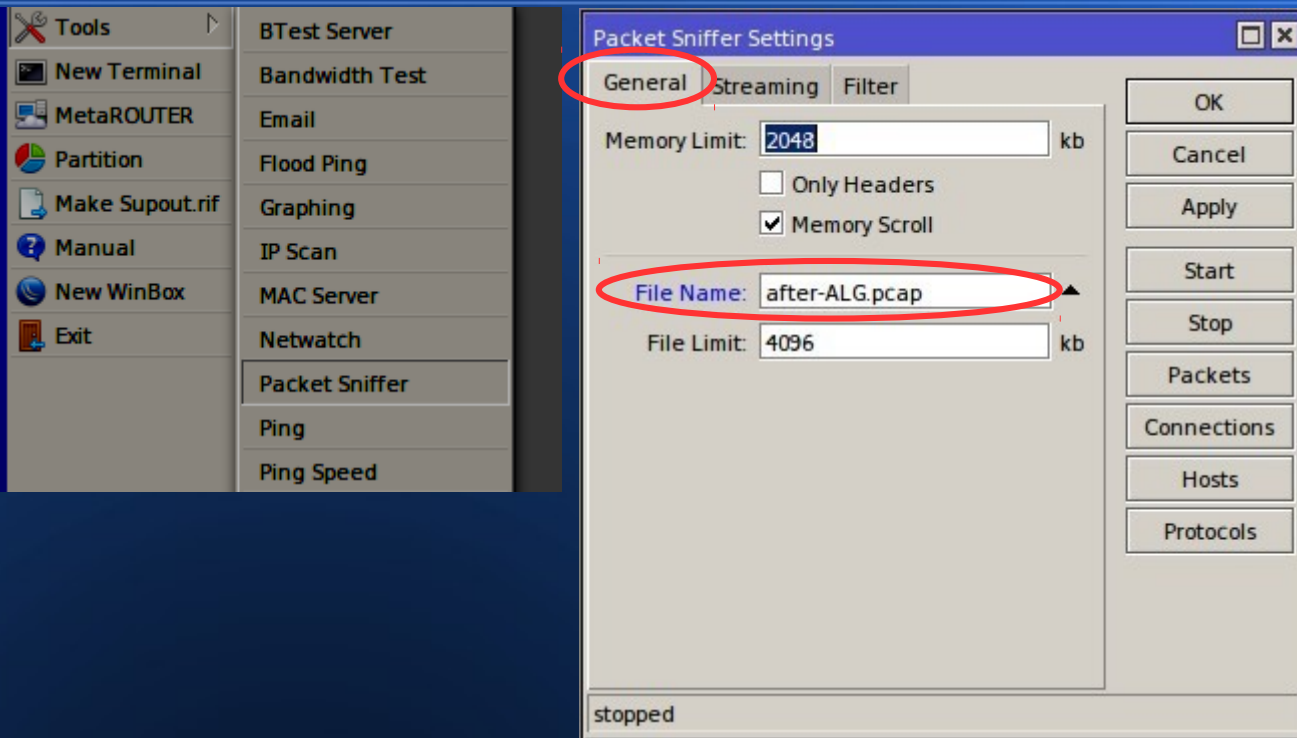
# After ALG modifications pcap



/tool sniffer



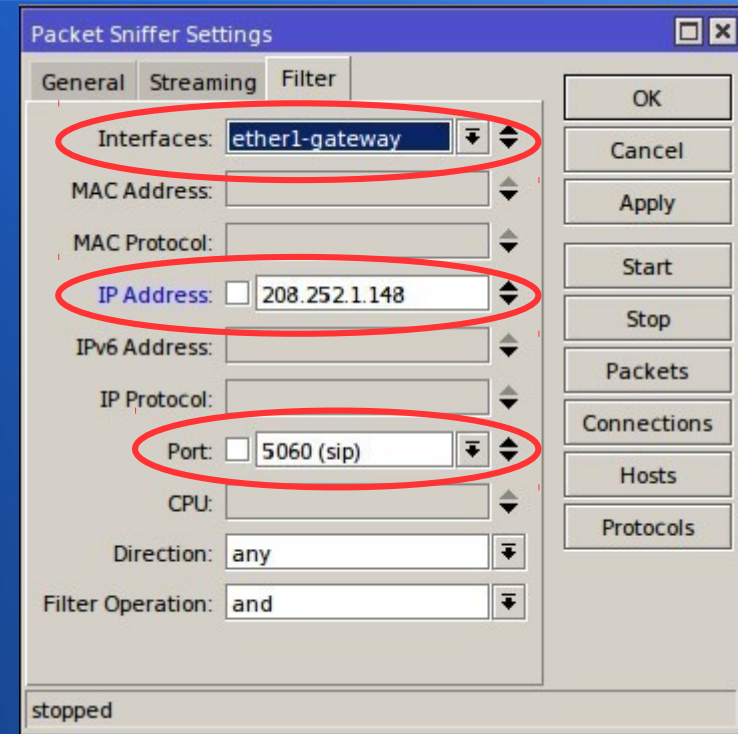
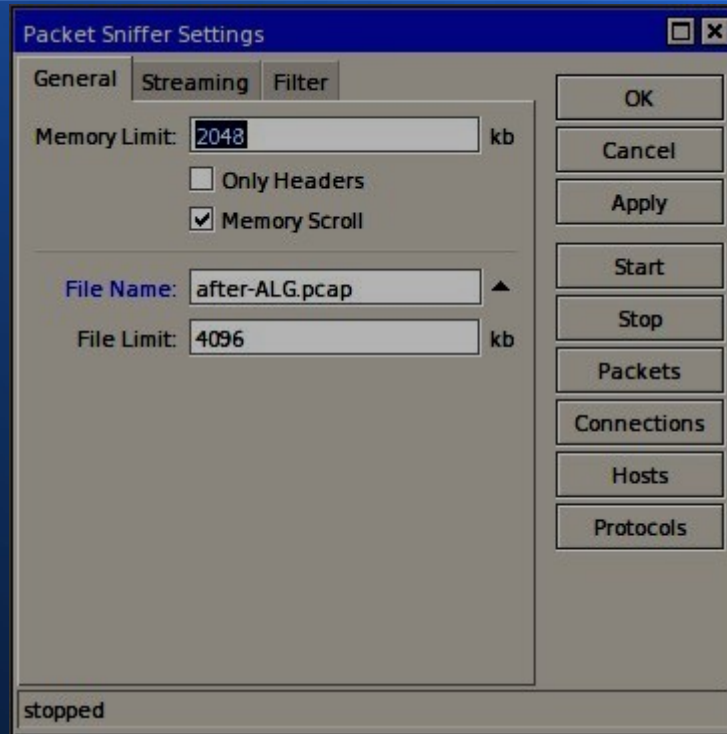
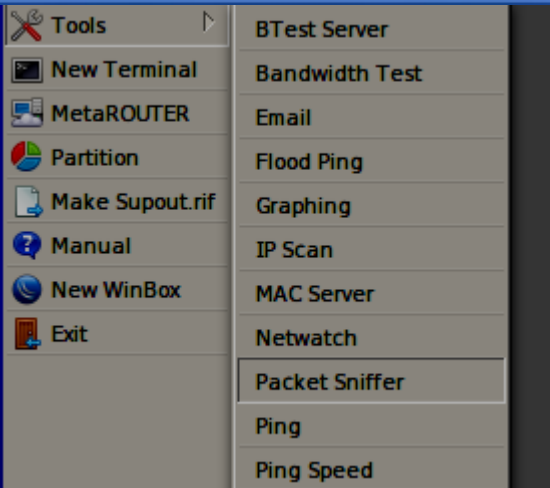
# After ALG modifications pcap



```
/tool sniffer
```

```
set only-headers=no file-name=after-ALG.pcap file-limit=4096
```

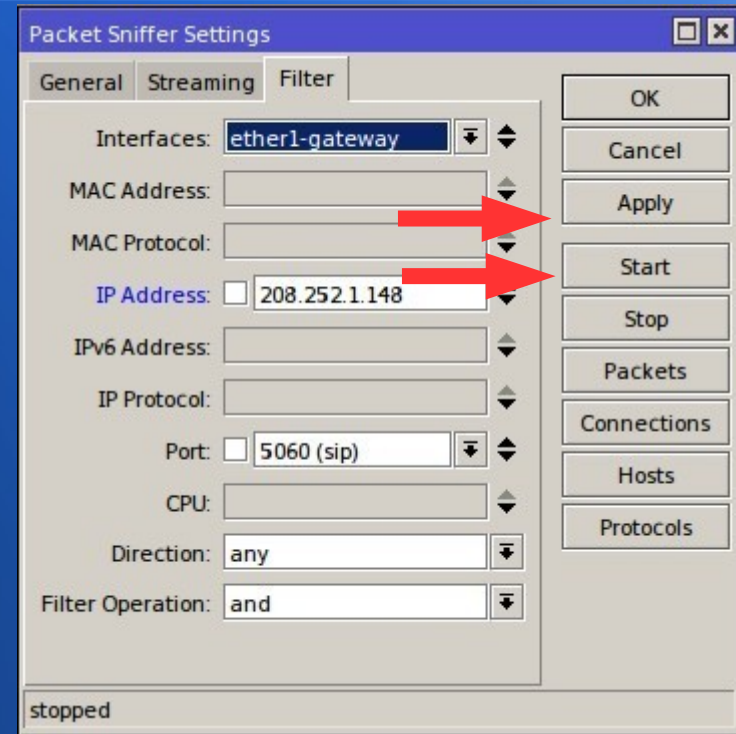
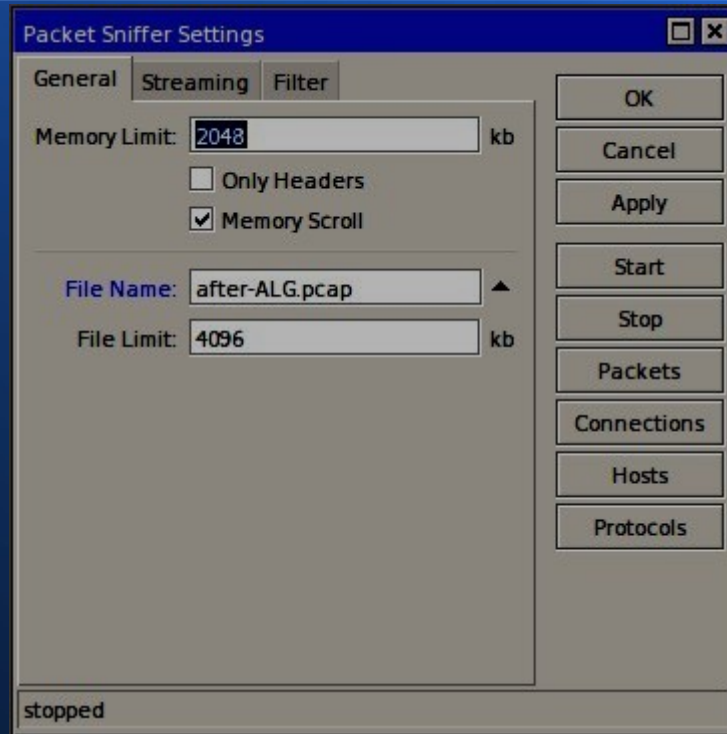
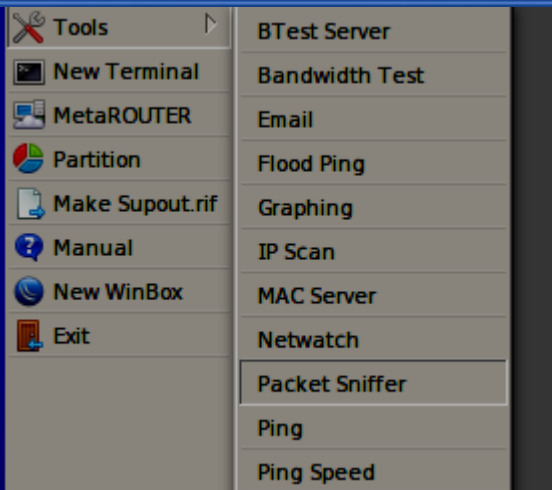
# After ALG modifications pcap



```
/tool sniffer
```

```
set only-headers=no file-name=after-ALG.pcap file-limit=4096 filter-interface=ether1-gateway  
filter-ip-address=207.252.1.148/32 filter-port=5060 filter-direction=any
```

# After ALG modifications pcap



```
/tool sniffer
```

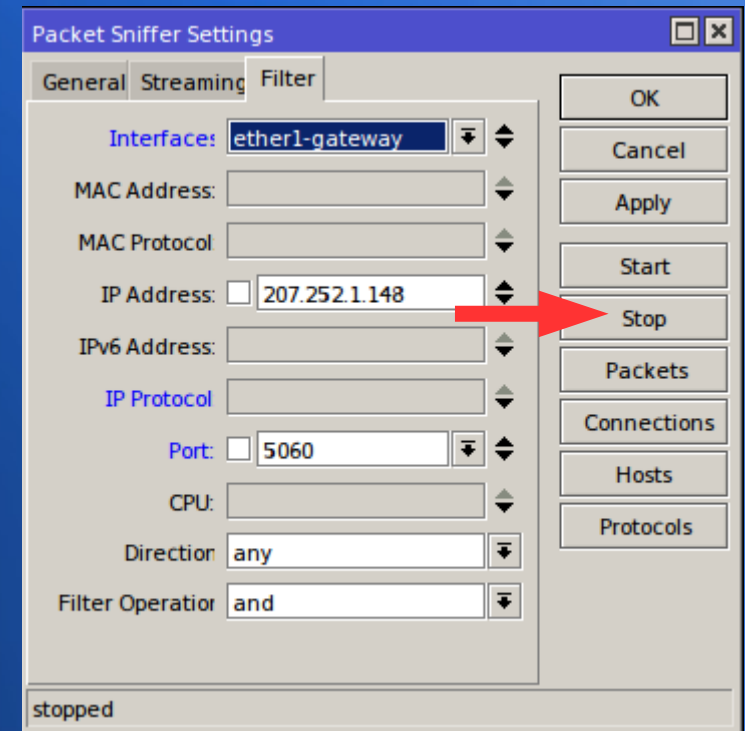
```
set only-headers=no file-name=after-ALG.pcap file-limit=4096 filter-interface=ether1-gateway  
filter-ip-address=207.252.1.148/32 filter-port=5060 filter-direction=any
```

```
start
```

Generate some traffic while the sniffer is capturing packets.

# After ALG modifications pcap

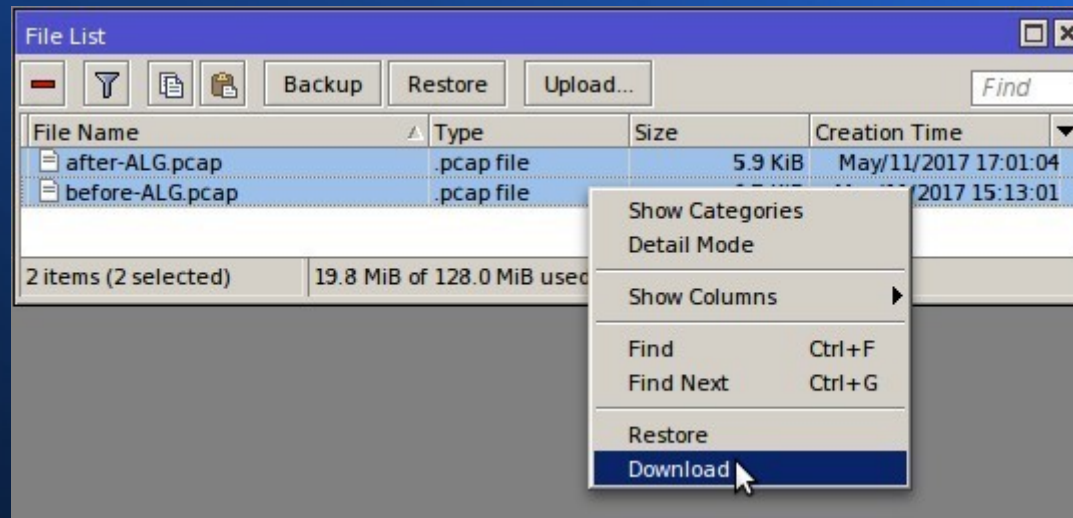
Make sure to stop the sniffer



```
/tool sniffer
```

```
stop
```

# Download pcap files



# Decode in wireshark

Wireshark - Follow UDP Stream (udp.stream eq 2) - after-ALG

```
REGISTER sip:207.252.1.148 SIP/2.0
Via: SIP/2.0/UDP 207.252.1.145:1024;branch=z9hG4bK-82c1a276
From: "100" <sip:100@207.252.1.148>;tag=1d8c3b3fa6a8a34co1
To: "100" <sip:100@207.252.1.148>
Call-ID: e68aae07-897b8a71@192.168.20.100
CSeq: 18716 REGISTER
Max-Forwards: 70
Contact: "100" <sip:100@207.252.1.145:1024>;expires=3600
User-Agent: Cisco/SPA504G-7.6.2b
Content-Length: 0
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER, UPDATE
Supported: replaces

REGISTER sip:207.252.1.148 SIP/2.0
Via: SIP/2.0/UDP 207.252.1.145:1024;branch=z9hG4bK-3586ccde
From: "100" <sip:100@207.252.1.148>;tag=1d8c3b3fa6a8a34co1
To: "100" <sip:100@207.252.1.148>
Call-ID: e68aae07-897b8a71@192.168.20.100
CSeq: 18717 REGISTER
Max-Forwards: 70
Authorization: Digest
username="100", realm="asterisk", nonce="6c91b3a5", uri="sip:
207.252.1.148", algorithm=MD5, response="81d37d55728c12cd6ef39e01d6ee928d"
Contact: "100" <sip:100@207.252.1.145:1024>;expires=3600
User-Agent: Cisco/SPA504G-7.6.2b
Content-Length: 0
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER, UPDATE
Supported: replaces
```

Packet 5. 4 client pkt(s), 0 server pkt(s), 0 turns. Click to select.

207.252.1.145:1024 → 207.252.1.148:5060 (1) Show data as ASCII Stream 2

Find:  Find Next

Help Hide this stream Print Save as... Close

# ALG Enabled & after modification

Before modification

Wireshark · Follow UDP Stream (udp.stream eq 4) · before-ALG

```
REGISTER sip:207.252.1.148 SIP/2.0
Via: SIP/2.0/UDP 192.168.20.100:5060;branch=z9hG4bK-eefddbff
From: "100" <sip:100@207.252.1.148>;tag=36d71bb091995583o1
To: "100" <sip:100@207.252.1.148>
Call-ID: 6593bf0b-e1cc0d64@192.168.20.100
CSeq: 60159 REGISTER
Max-Forwards: 70
Contact: "100" <sip:100@192.168.20.100:5060>;expires=3600
User-Agent: Cisco/SPA504G-7.6.2b
Content-Length: 0
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER, UPDATE
Supported: replaces

REGISTER sip:207.252.1.148 SIP/2.0
Via: SIP/2.0/UDP 192.168.20.100:5060;branch=z9hG4bK-e9fcb8df
From: "100" <sip:100@207.252.1.148>;tag=36d71bb091995583o1
To: "100" <sip:100@207.252.1.148>
Call-ID: 6593bf0b-e1cc0d64@192.168.20.100
CSeq: 60160 REGISTER
Max-Forwards: 70
Authorization: Digest username="100",realm="asterisk",nonce="01a48834",
207.252.1.148",algorithm=MD5,response="36cd61d13d1c58ed830cf64e3b47b82a
Contact: "100" <sip:100@192.168.20.100:5060>;expires=3600
User-Agent: Cisco/SPA504G-7.6.2b
Content-Length: 0
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER, UPDATE
Supported: replaces
```

4 client pkt(s), 0 server pkt(s), 0 turns.

192.168.20.100:5060 → 207.252.1.148:5060 (9940 by ▼

Show data as ASCII ▼

Find:

Help

Hide this stream

Print

Save as...

Wireshark · Follow UDP Stream (udp.stream eq 2) · after-ALG

```
REGISTER sip:207.252.1.148 SIP/2.0
Via: SIP/2.0/UDP 207.252.1.145:1024;branch=z9hG4bK-82c1a276
From: "100" <sip:100@207.252.1.148>;tag=1d8c3b3fa6a8a34co1
To: "100" <sip:100@207.252.1.148>
Call-ID: e68aae07-897b8a71@192.168.20.100
CSeq: 18716 REGISTER
Max-Forwards: 70
Contact: "100" <sip:100@207.252.1.145:1024>;expires=3600
User-Agent: Cisco/SPA504G-7.6.2b
Content-Length: 0
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER, UPDATE
Supported: replaces

REGISTER sip:207.252.1.148 SIP/2.0
Via: SIP/2.0/UDP 207.252.1.145:1024;branch=z9hG4bK-3586ccde
From: "100" <sip:100@207.252.1.148>;tag=1d8c3b3fa6a8a34co1
To: "100" <sip:100@207.252.1.148>
Call-ID: e68aae07-897b8a71@192.168.20.100
CSeq: 18717 REGISTER
Max-Forwards: 70
Authorization: Digest
username="100",realm="asterisk",nonce="6c91b3a5",uri="sip:
207.252.1.148",algorithm=MD5,response="81d37d55728c12cd6ef39e01d6ee928d
Contact: "100" <sip:100@207.252.1.145:1024>;expires=3600
User-Agent: Cisco/SPA504G-7.6.2b
Content-Length: 0
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER, UPDATE
Supported: replaces
```

Packet 5. 4 client pkt(s), 0 server pkt(s), 0 turns. Click to select.

207.252.1.145:1024 → 207.252.1.148:5060 (1 ▼

Show data as ASCII ▼ Stream

Find:  Find N

Help

Hide this stream

Print

Save as...

Clos



# When is SIP ALG unnecessary?

# When is SIP ALG unnecessary?

When the SIP device is NAT aware.

1. Server is behind NAT
2. Server is outside NAT (public server)

# When is SIP ALG unnecessary?

## SIP servers behind NAT

- Nat aware SIP servers have the option to detect their WAN ip and write it in the SIP/SDP messages where necessary, before sending it.
- FreePBX detects the WAN ip and inserts it in SIP messages where necessary.

WAN  
75.142.151.49



Private / NAT

Server  
192.168.20.2



WAN  
75.142.151.49



Private / NAT

Server  
192.168.20.2



SIP headers  
Via: 75.142.151.49  
Contact: 75.142.151.49

SDP Body  
o = IN IP4 75.142.151.49  
c = IN IP4 75.142.151.49  
m = audio 19032

ALG Disabled

WAN  
75.142.151.49



Private / NAT

Server  
192.168.20.2



SIP headers  
Via: 75.142.151.49  
Contact: 75.142.151.49

SDP Body  
o = IN IP4 75.142.151.49  
c = IN IP4 75.142.151.49  
m = audio 19032

# Servers outside NAT (public server)

# Servers outside NAT (public server)

- SIP servers have NAT options for each extension



# Servers outside NAT (public server)

- SIP servers have NAT options for each extension
- If server side extension states NAT=Yes then send all responses to the client originating IP and Port.



WAN  
75.142.151.49

ALG Disabled

Private / NAT

WAN  
75.142.151.49

ALG Disabled



Private / NAT

X100  
192.168.20.100





WAN  
75.142.151.49

ALG Disabled



Private / NAT

X100  
192.168.20.100





WAN  
75.142.151.49

ALG Disabled



Private / NAT

X100  
192.168.20.100





Extension 100  
NAT = Yes

WAN  
75.142.151.49

ALG Disabled



Private / NAT

X100  
192.168.20.100





Sip Server  
207.252.1.148



WAN  
75.142.151.49

ALG Disabled

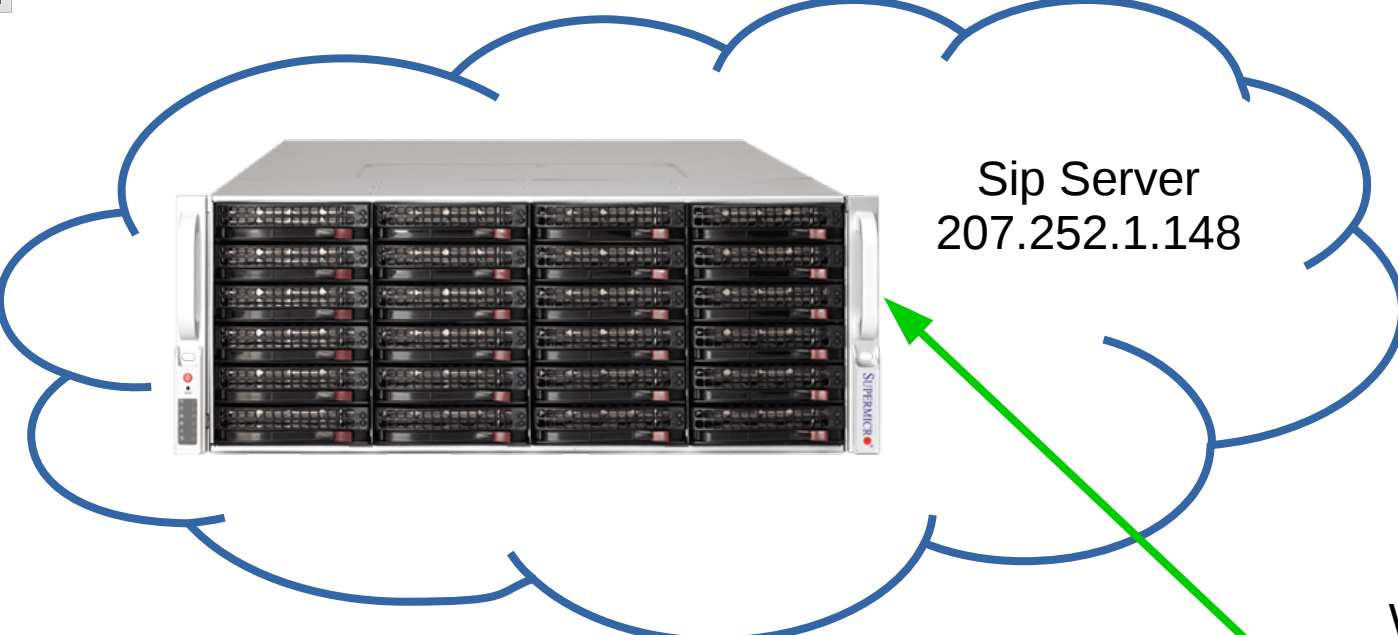
Private / NAT

X100  
192.168.20.100



SIP headers  
Via: 192.168.20.100  
Contact: 192.168.20.100

SDP Body  
o = IN IP4 192.168.20.100  
c = IN IP4 192.168.20.100  
m = audio 19032



Sip Server  
207.252.1.148

ALG Disabled

WAN  
75.142.151.49



Private / NAT

X100  
192.168.20.100



SIP headers  
Via: 192.168.20.100  
Contact: 192.168.20.100  
  
SDP Body  
o = IN IP4 192.168.20.100  
c = IN IP4 192.168.20.100  
m = audio 19032





SIP headers  
Via: 192.168.20.100  
Contact: 192.168.20.100

SDP Body  
o = IN IP4 192.168.20.100  
c = IN IP4 192.168.20.100  
m = audio 19032

WAN  
75.142.151.49



Private / NAT

X100  
192.168.20.100





SIP headers

Via: ~~192.168.20.100~~

Contact: ~~192.168.20.100~~

SDP Body

o = IN IP4 ~~192.168.20.100~~

c = IN IP4 ~~192.168.20.100~~

m = audio ~~19032~~

VWAN

75.142.151.49



Private / NAT

X100  
192.168.20.100





SIP headers

Via: received=75.142.151.49

Contact: ~~192.168.20.100~~

SDP Body

o = IN IP4 ~~192.168.20.100~~

c = IN IP4 ~~192.168.20.100~~

m = audio ~~25481~~

WAN  
75.142.151.49



Private / NAT

X100  
192.168.20.100



# When does ALG break VoIP?

# When does ALG break VoIP?

- DOES NOT HAPPEN WITH Mikrotik RouterOS !
- Poor quality ALG's replace ALL private IP's in SIP headers, including Call-ID

# When does ALG break VoIP?

REGISTER sip:207.252.1.148 SIP/2.0

Via: SIP/2.0/UDP 192.168.20.100:5060;branch=z9hG4bK-8fb0e171

From: "David Attias" <sip:201525@207.252.1.148>;tag=191914b06be0

To: "David Attias" <sip:201525@207.252.1.148>

**Call-ID: 6894e30c-h1c8d357@192.168.20.100**

CSeq: 1373 REGISTER

Max-Forwards: 70

Contact: "David Attias" <sip:201525@192.168.20.100:5060>;expires=3600

User-Agent: Cisco/SPA504G-7.6.2b

Content-Length: 0

Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER, UPDATE

Supported: replaces

# When does ALG break VoIP?

REGISTER sip:207.252.1.148 SIP/2.0

Via: SIP/2.0/UDP 192.168.20.100:5060;branch=z9hG4bK-8fb0e171

From: "David Attias" <sip:201525@207.252.1.148>;tag=191914b06beo0

To: "David Attias" <sip:201525@207.252.1.148>

Call-ID: 6894e30c-h1c8d357@192.168.20.100

NEVER change anything in this field !!!

CSeq: 1373 REGISTER

Max-Forwards: 70

Contact: "David Attias" <sip:201525@192.168.20.100:5060>;expires=3600

User-Agent: Cisco/SPA504G-7.6.2b

Content-Length: 0

Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER, UPDATE

Supported: replaces

# When does ALG break VoIP?

- DOES NOT HAPPEN WITH RouterOS !
- Poor quality ALG's replace ALL private IP's in SIP headers, including Call-ID
- Poor quality ALG's unnecessarily adds a ; which breaks the syntax of sip requests.



# SIP ALG Timeout

## The problem:

- The phone sets layer 7 session timeout on the server.
- The router sets the UDP timeout for the session.

# SIP ALG Timeout

## The problem:

- The phone sets layer 7 session timeout on the server.
- The router sets the UDP timeout for the session.
- If the router session timeout expires before the server session timeout, the server would send data to an expired session (closed return port)

# SIP ALG Timeout

## The Solution:

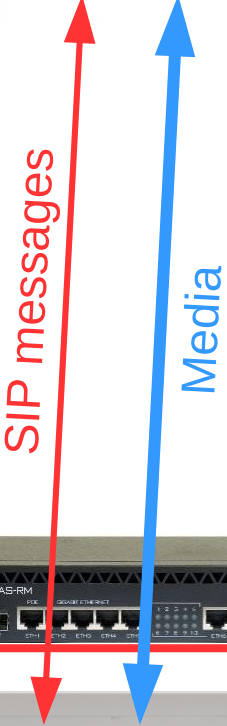
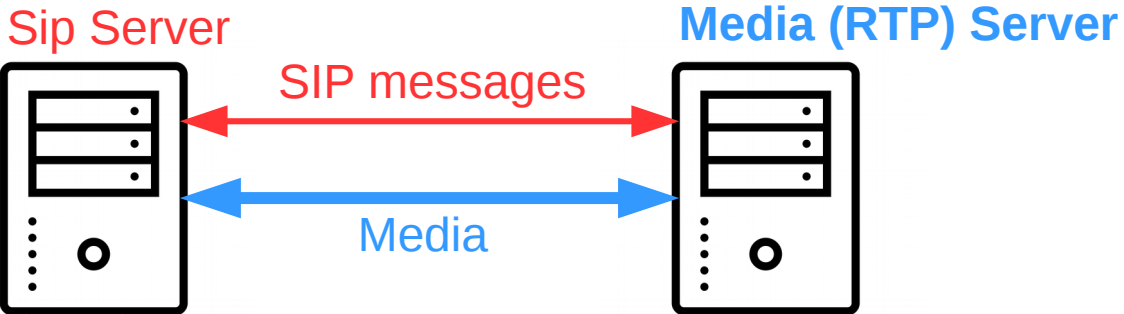
- Manually set SIP ALG timeout
- Set it higher than your lowest sip keepalive message interval (register, invite, options)

# SIP Direct Media

# SIP Direct Media

- Allows a redirect of the RTP media stream to go directly from SIP device to SIP device, “cutting out the middle man”
- The SIP servers are responsible for setting up the direct media stream.
- After the initial call is established the NAT’ed SIP server will re-invite the public media server to establish a direct media connection, bypassing the middle server

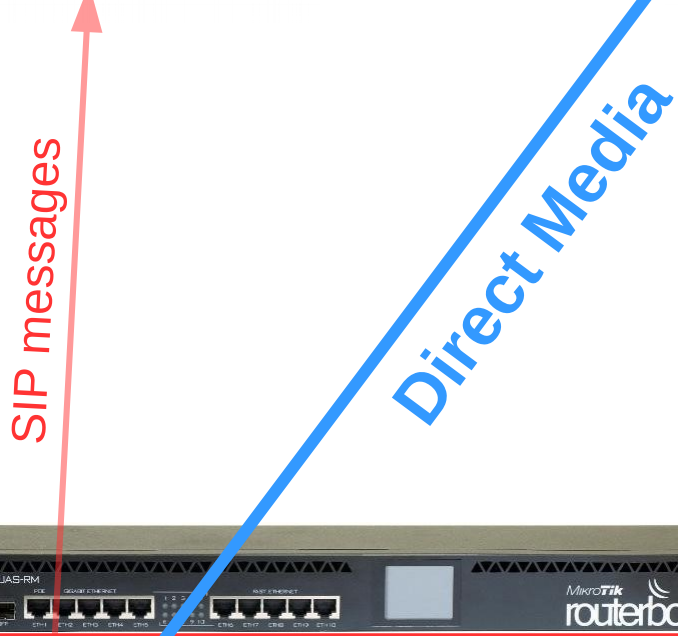
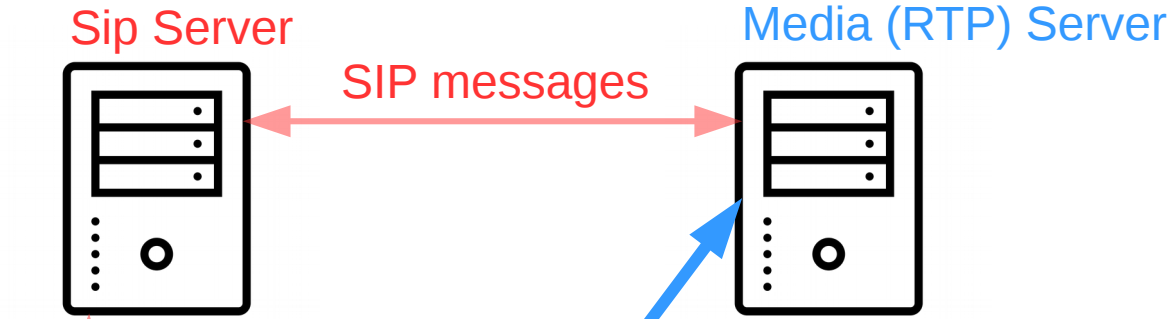
# Standard Flow



Private / NAT



# Direct-Media



Private / NAT

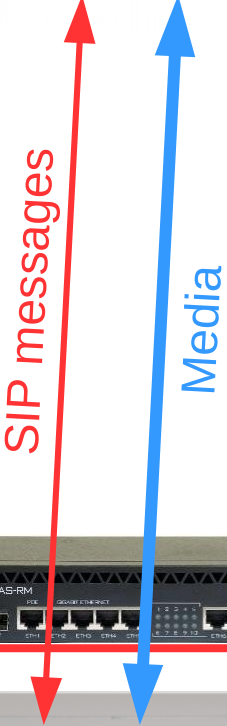
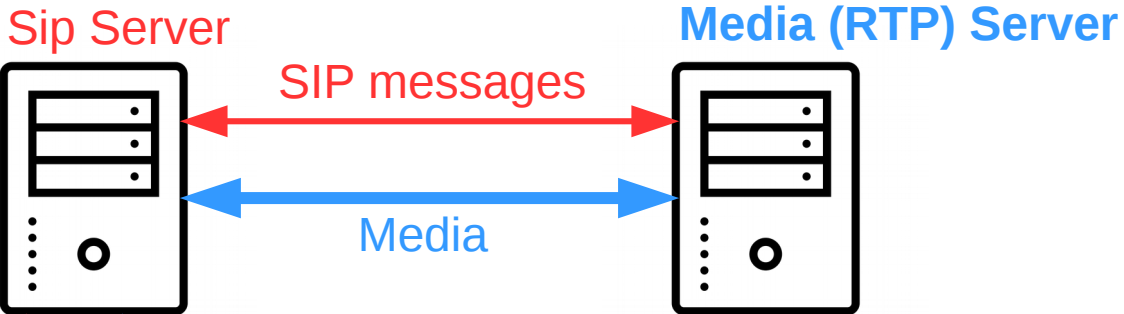


# SIP Direct Media

- The sip-direct-media option has the ability to block or allow the NAT'ed server from re-inviting the media server for a direct media session.
- sip-direct-media yes  
Allows direct media re-invites
- sip-direct-media no  
Blocks direct media re-invites



# Standard Flow



Private / NAT



# Re-cap

- 1- Use SIP ALG when your NAT'ed sip device is NOT NAT aware!
- 2- Make sure you set your SIP-Server ports correctly
- 3- Set your UDP timeout higher than your sip keep alive
- 4- Don't fear SIP ALG, it's designed to make your job easier !

# Agenda

- 1- What is ALG & what does it do.
- 2- The problem with VoIP and NAT
- 3- When is SIP ALG necessary and unnecessary?
- 4- How SIP ALG corrects problems.
- 5- Testing with Wireshark
- 6- SIP ALG Timeout
- 7- SIP ALG direct-media