# 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 un-necessary?
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 Gateway (firewall) that re-writes specific Application Layer 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 1~~92.168.20.100~~

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=191914b06beo0
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

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

After ALG modifies layer 7 data

# When is SIP ALG necessary?

# When is SIP ALG necessary?

When the SIP device behind NAT is NOT NAT aware

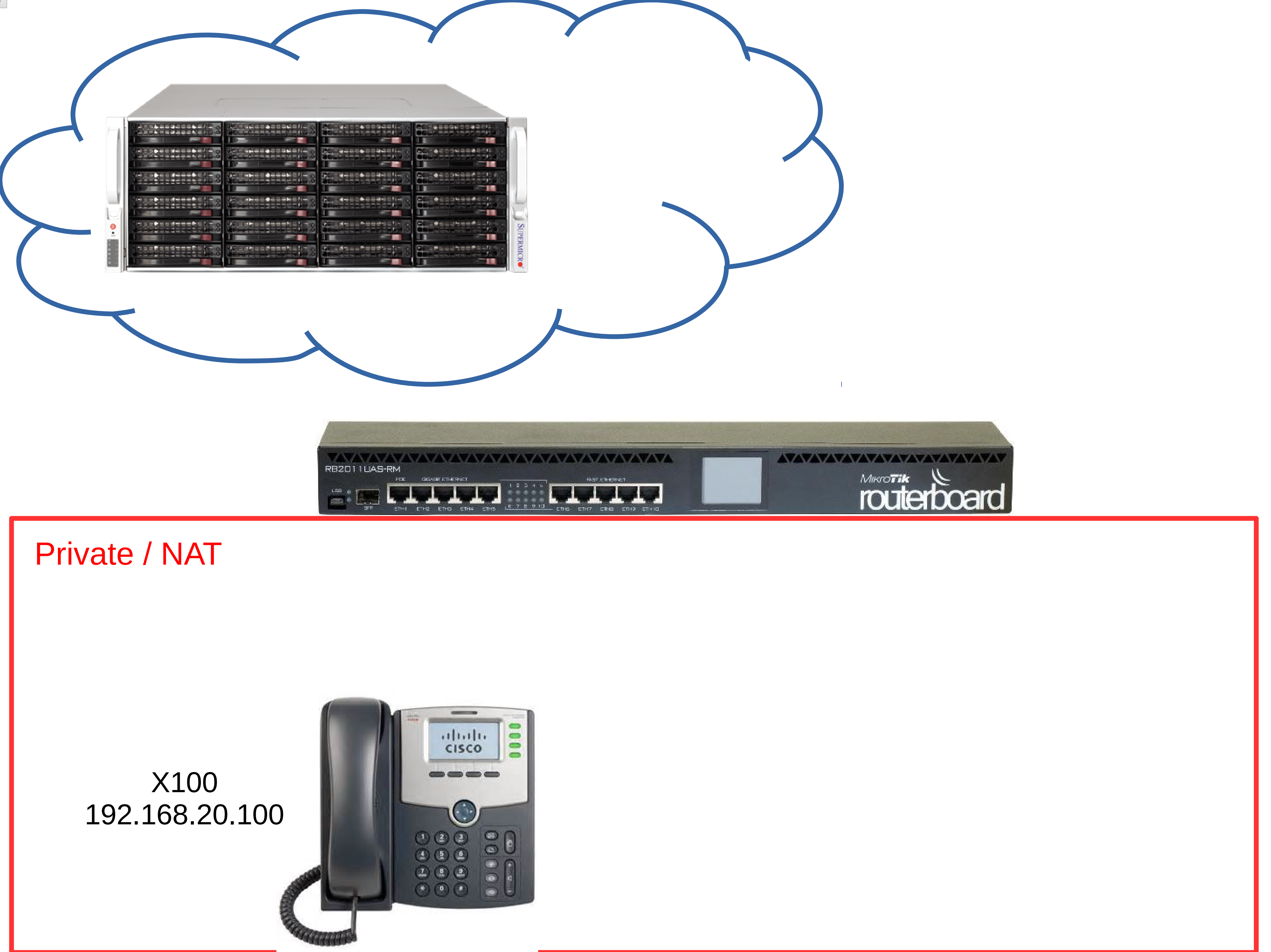
# SIP devices that are not “NAT Aware”

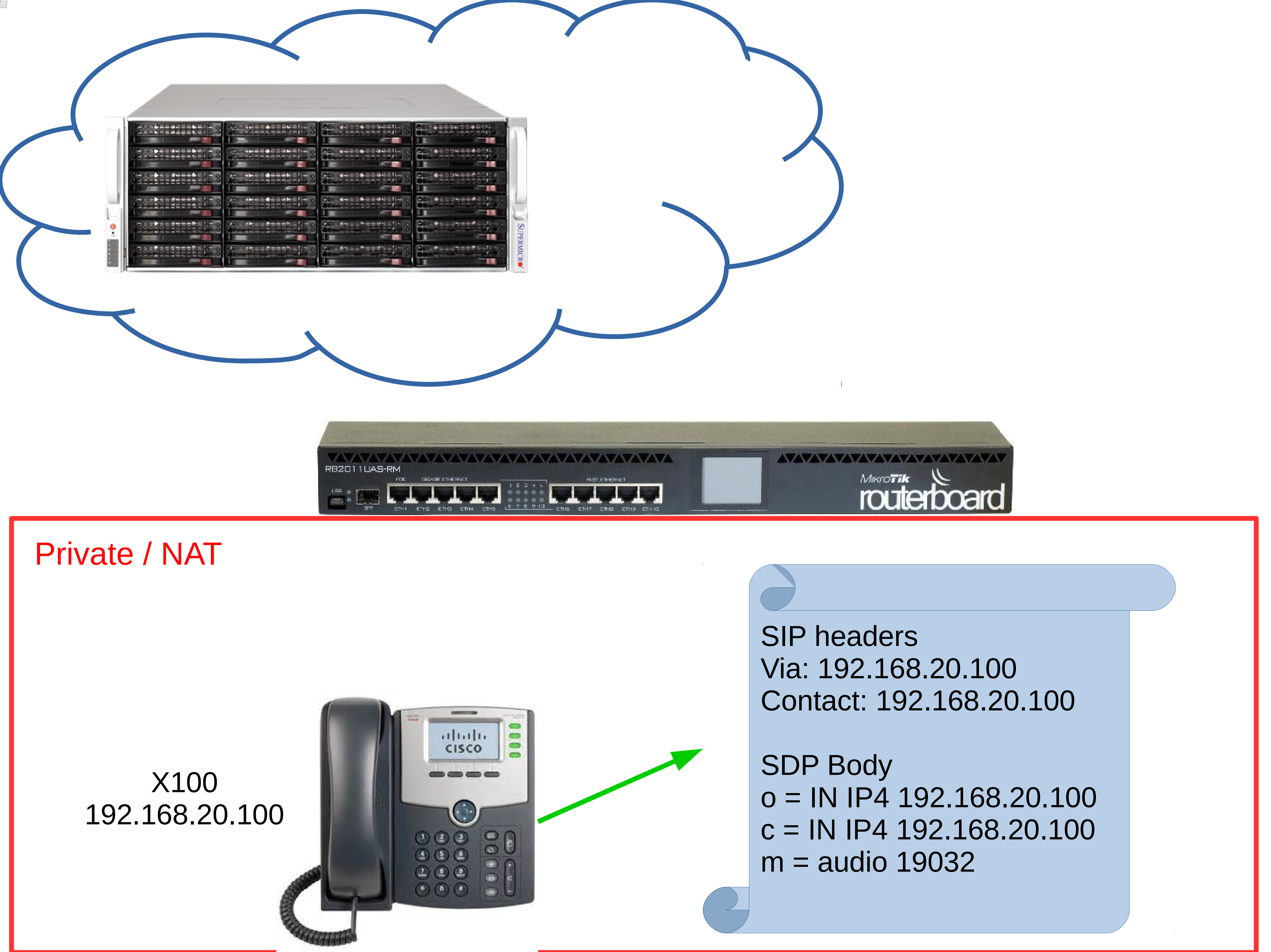
- Some SIP devices are not NAT aware and write their (private) device IP in layer 7 messages to the server.
- The remote SIP server receives the layer 7 message which specifies a private reply address.
- The server sends replies to the private address, which can never be reached.

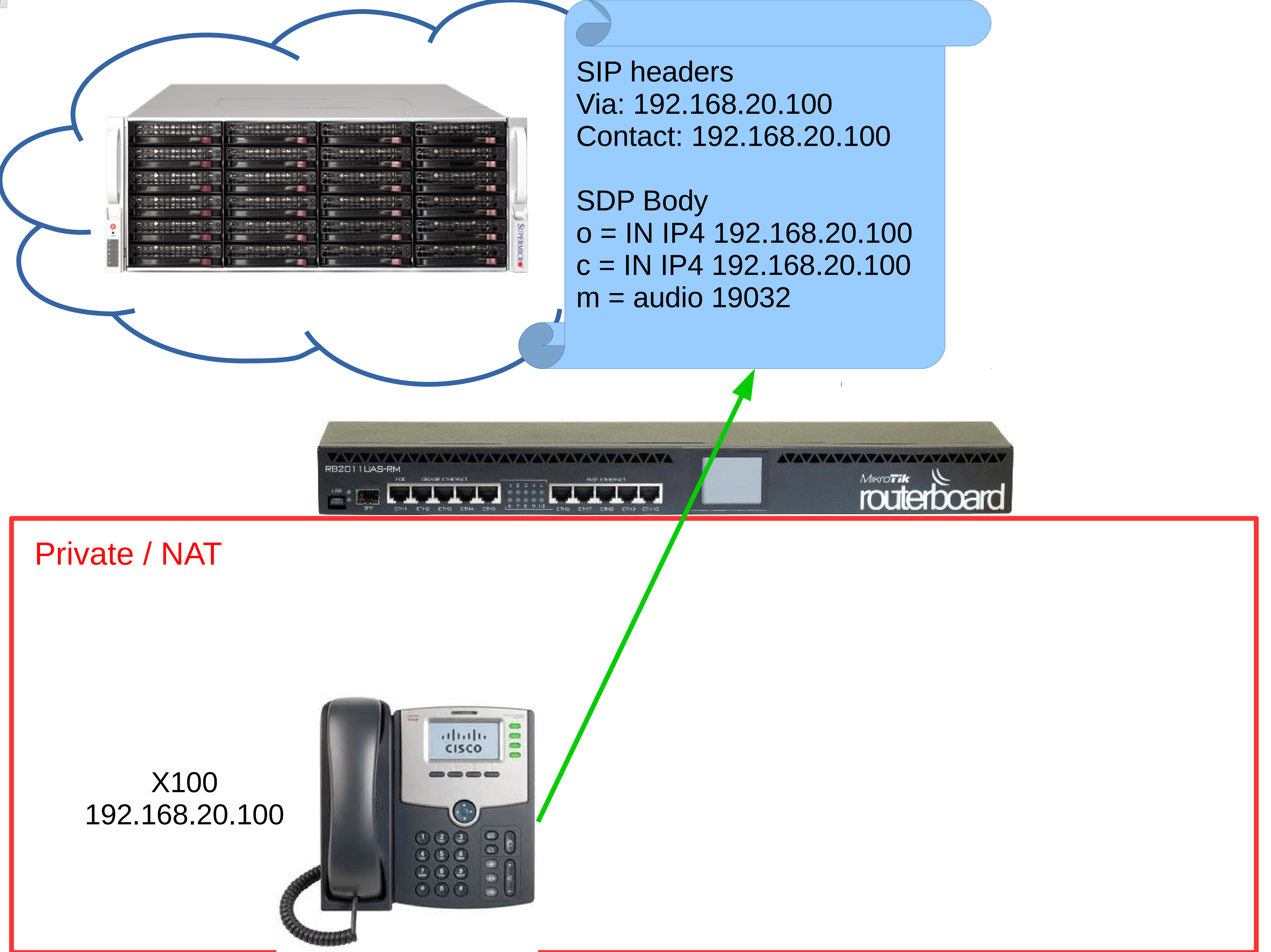
# SIP servers that are not “NAT Aware”

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

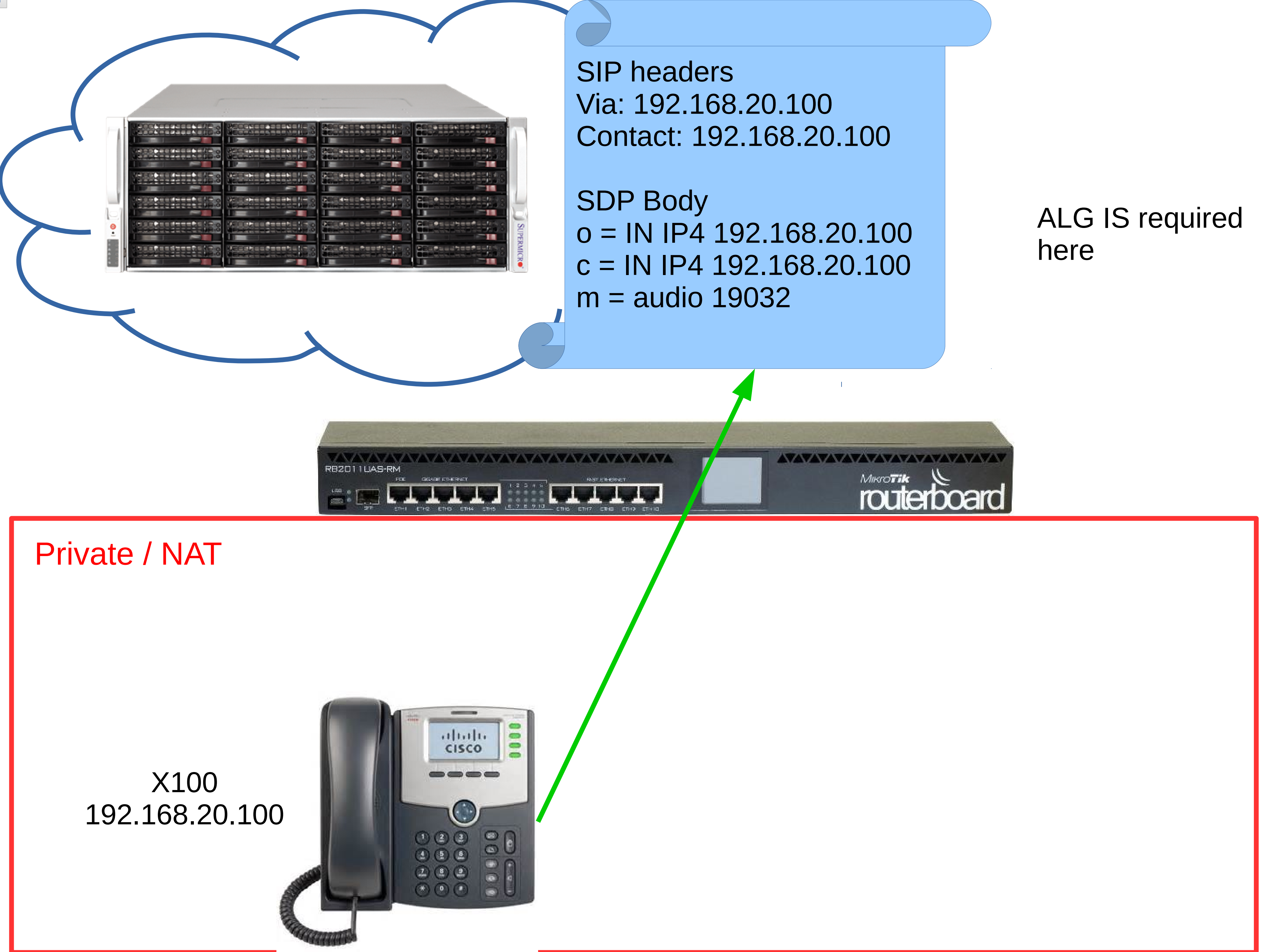
# Example of a SIP device that is not NAT aware


RB2011UAS-RM
MikroTik
routerboard
Private / NAT
X100
192.168.20.100
cisco

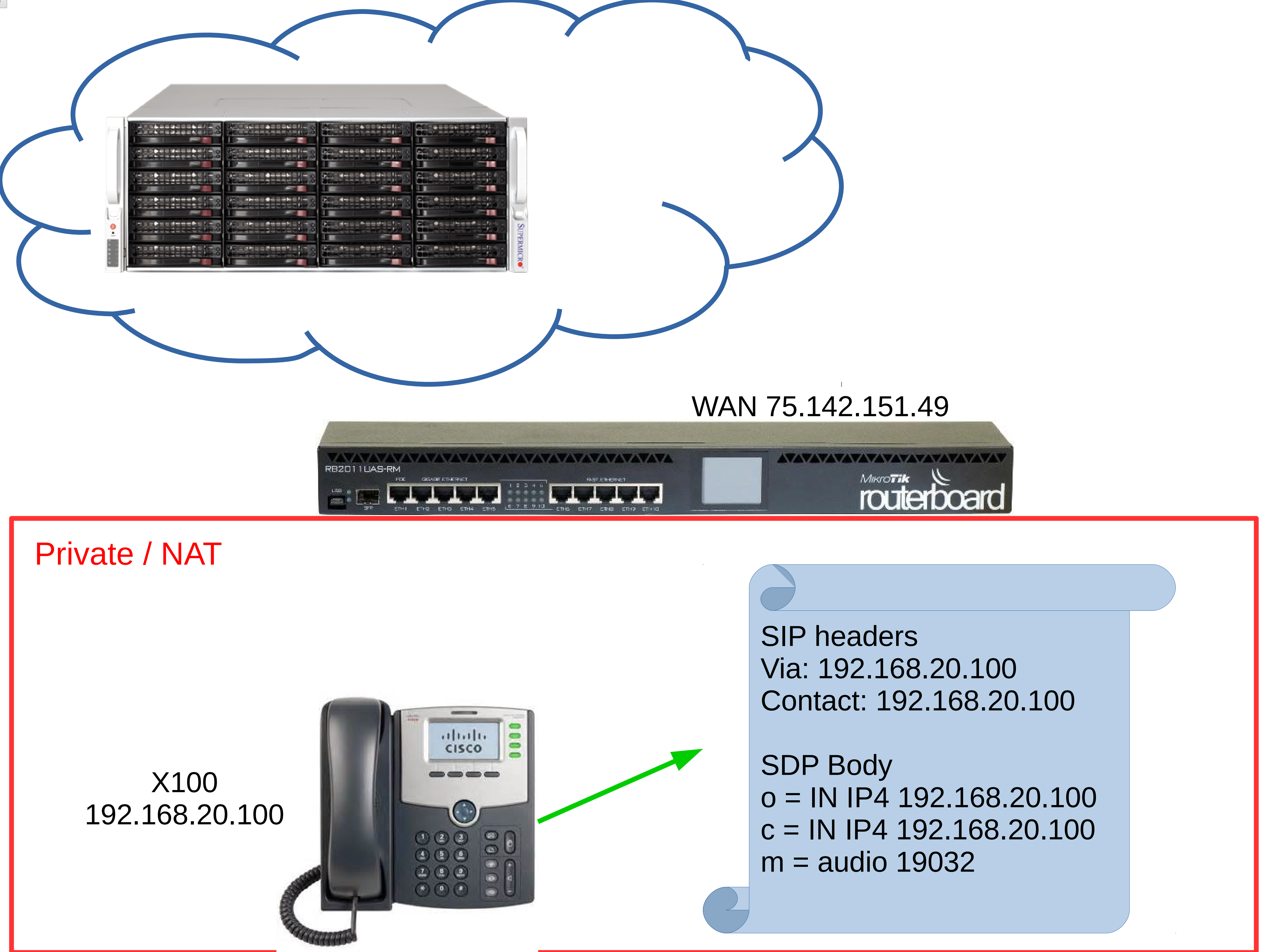

RB2011UAS-RM
Private / NAT
X100
192.168.20.100
SIP headers
Via: 192.168.20.100
Contact: 192.168.20.100
SDP Body
o = IN IP4 192.168.20.100
c = IN IP4 192.168.20.100
m = audio 19032

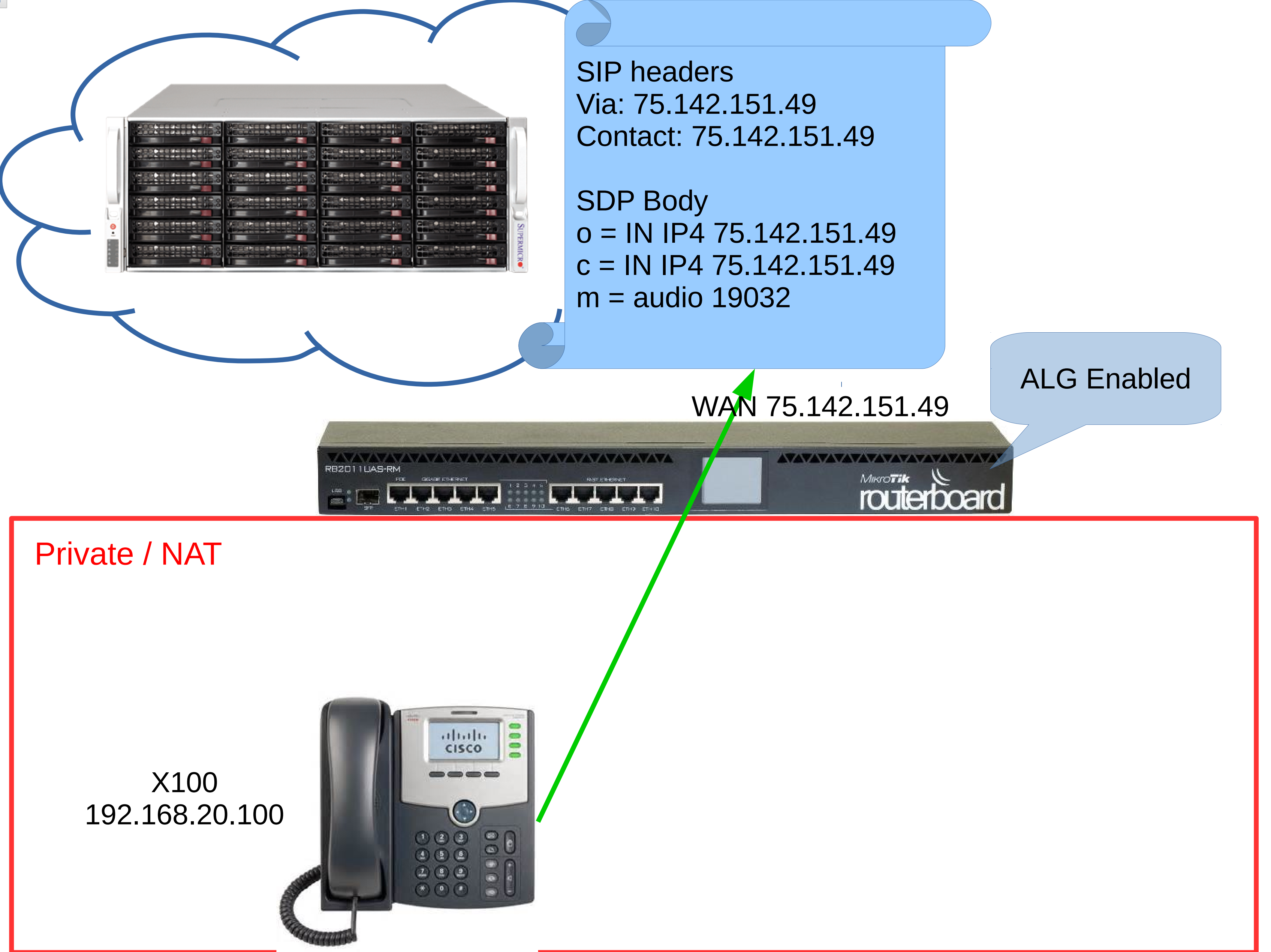

SIP headers
Via: 192.168.20.100
Contact: 192.168.20.100

SDP Body
o = IN IP4 192.168.20.100
c = IN IP4 192.168.20.100
m = audio 19032
RB2011UAS-RM
MikroTik
routerboard
Private / NAT
cisco
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
RB2011UAS-RM
MikroTik
routerboard
Private / NAT
cisco
X100
192.168.20.100

# With RouterOS SIP ALG enabled


WAN 75.142.151.49
RB2011UAS-RM
MikroTik
routerboard
Private / NAT
X100
192.168.20.100
cisco
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

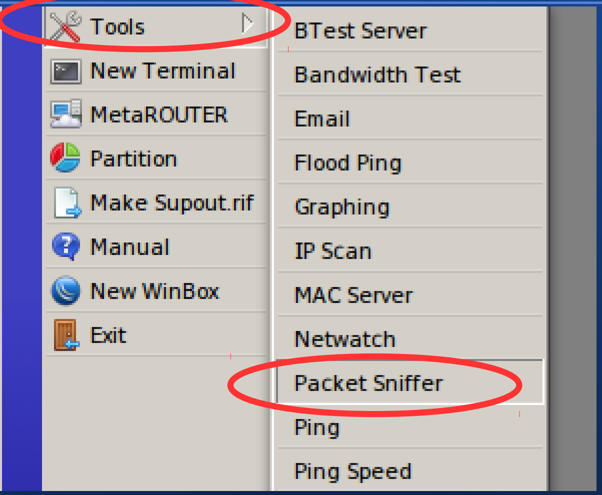

SIP headers
Via: 75.142.151.49
Contact: 75.142.151.49
SDP Body
o = IN IP4 75.142.151.49
c = IN IP4 75.142.151.49
m = audio 19032
ALG Enabled
WAN 75.142.151.49
RB2011UAS-RM
MikroTik
routerboard
Private / NAT
X100
192.168.20.100
CISCO

# 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

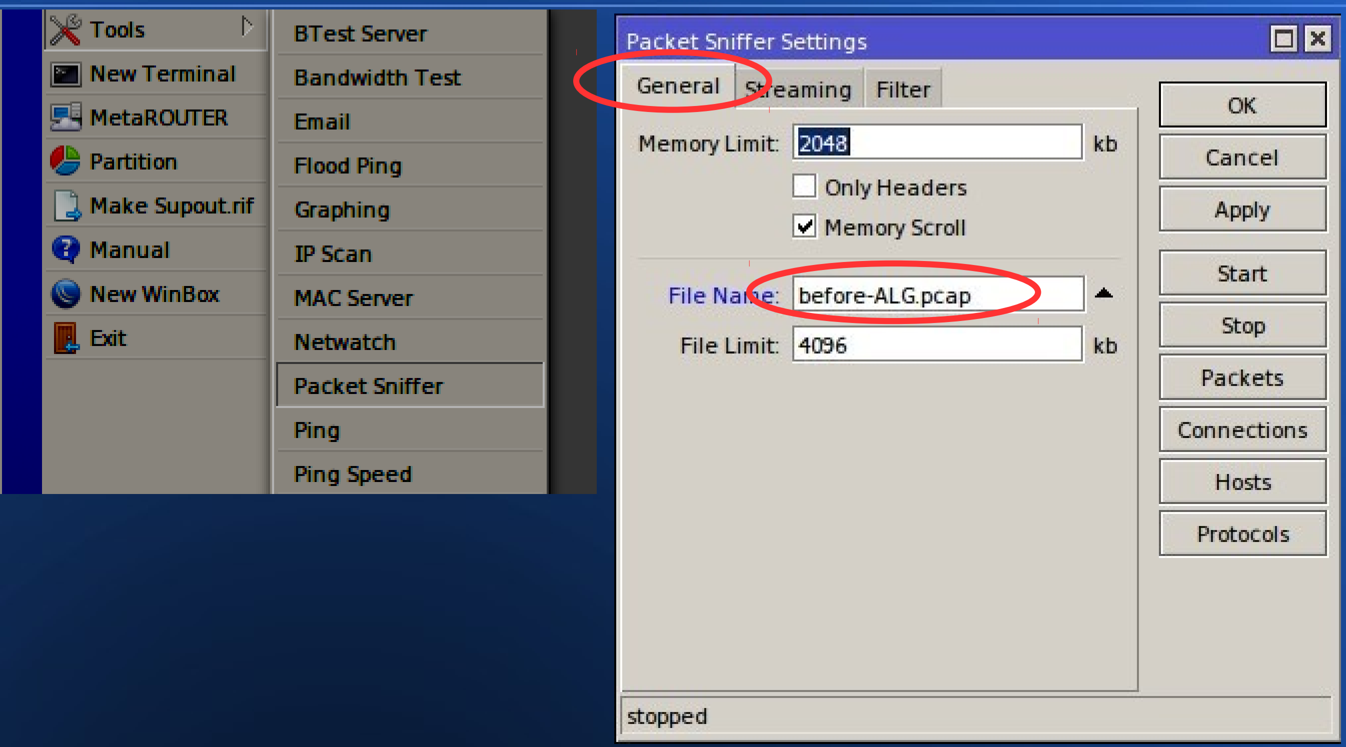
# Setting up the packet capture

# Before ALG modifications pcap



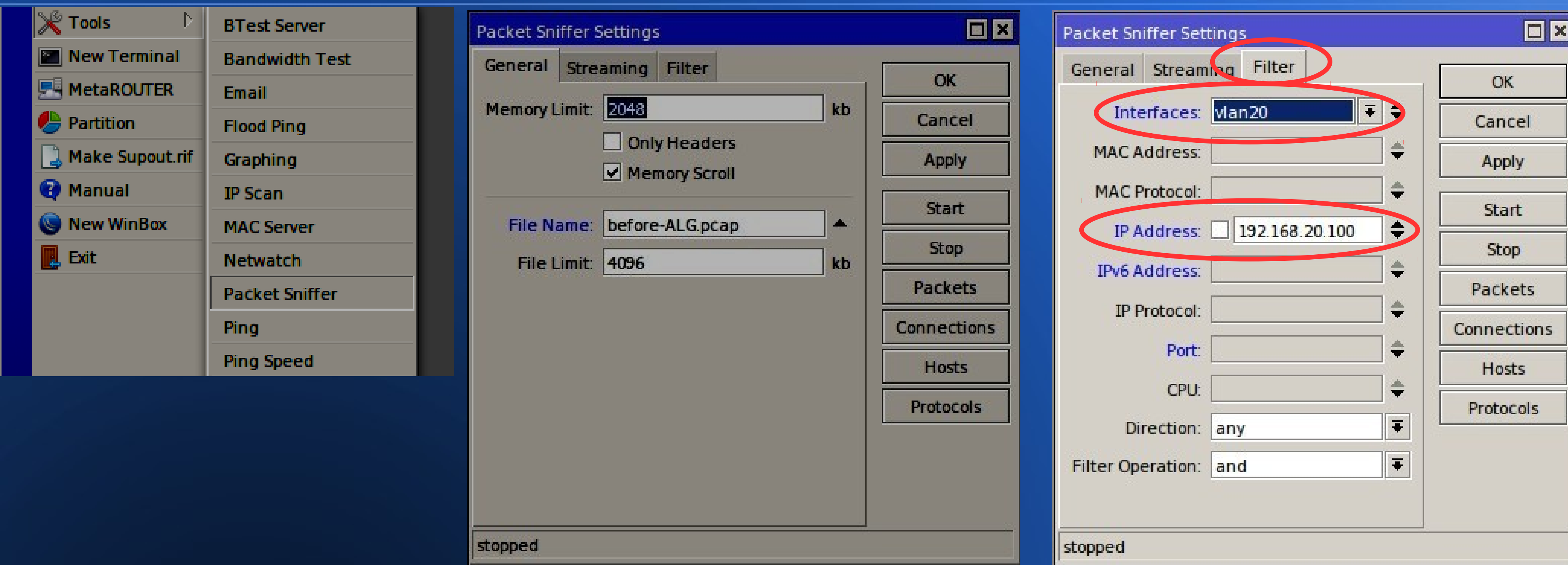

/tool sniffer

# Before ALG modifications pcap



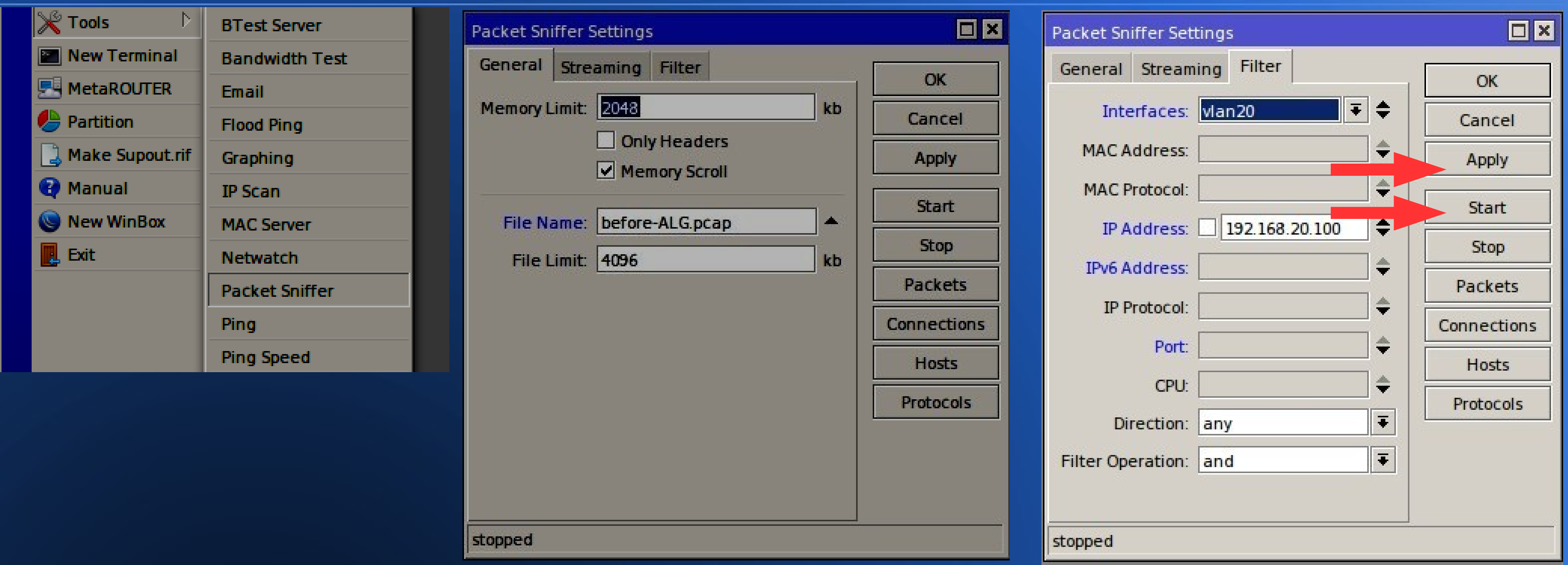

/tool sniffer
set only-headers=no file-name=before-ALG.pcap file-limit=4096

# 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



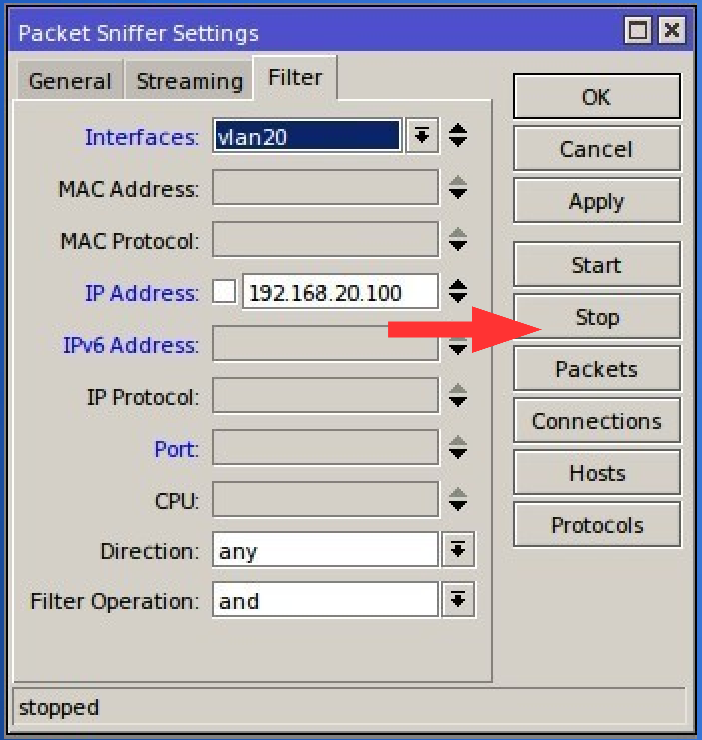

/tool sniffer
set only-headers=no file-name=before-ALG.pcap file-limit=4096 filter-interface=vlan20 filter-ip-address=192.168.20.100/32 filter-direction=any

start

Generate some traffic while the sniffer is capturing packets.

# Before ALG modifications pcap

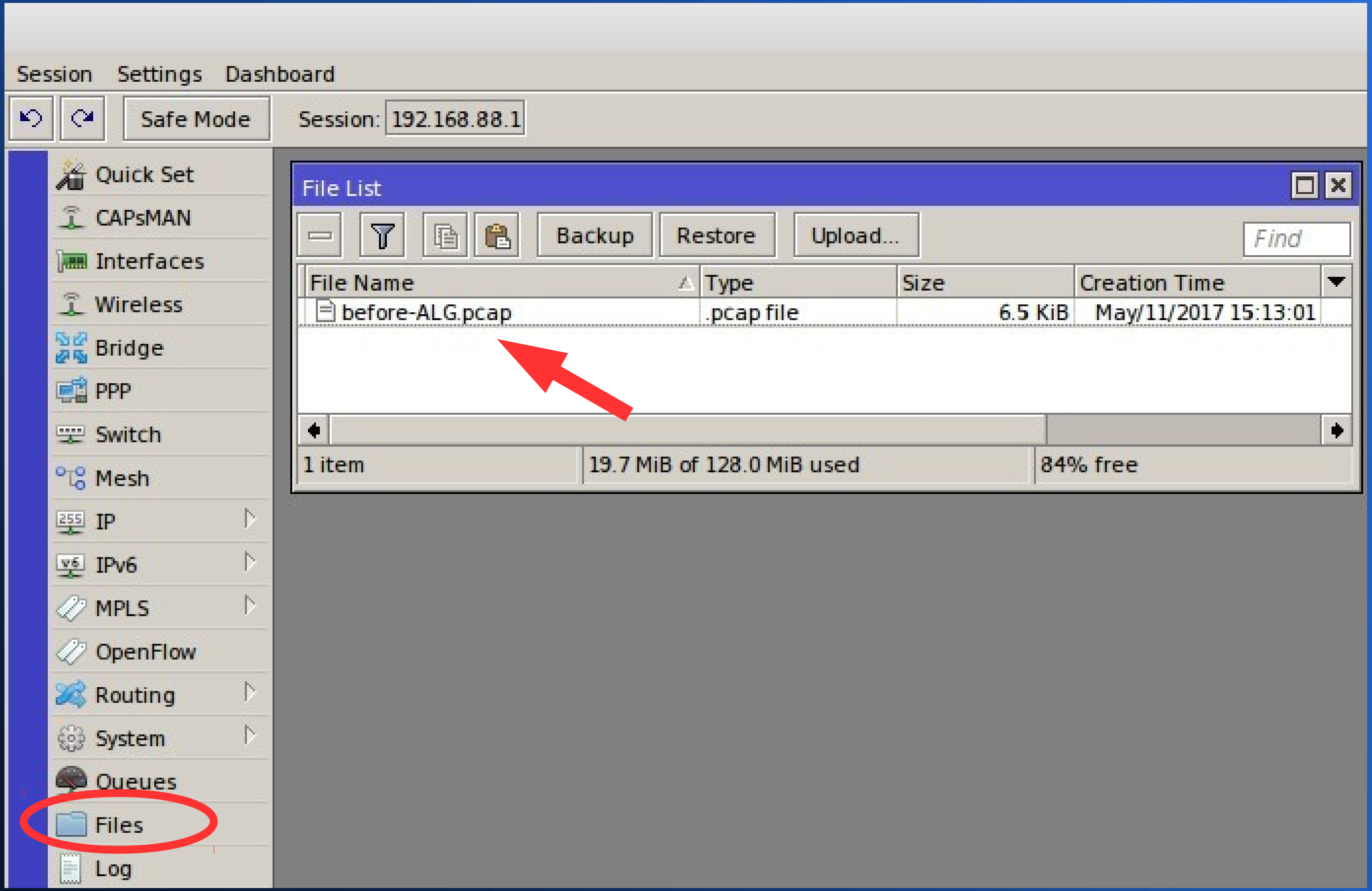
Make sure to stop the sniffer



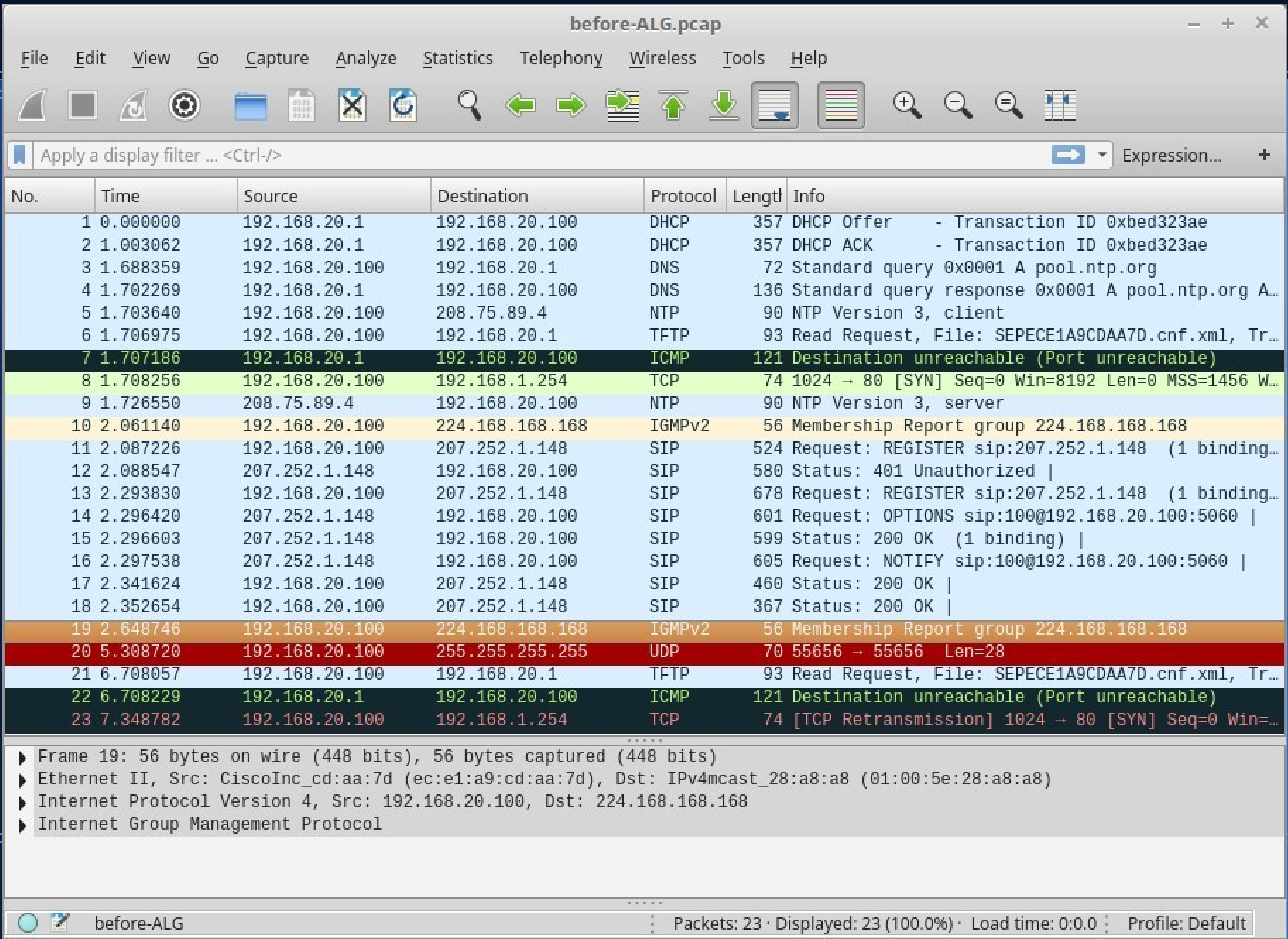

/tool sniffer

stop

# Download pcap files



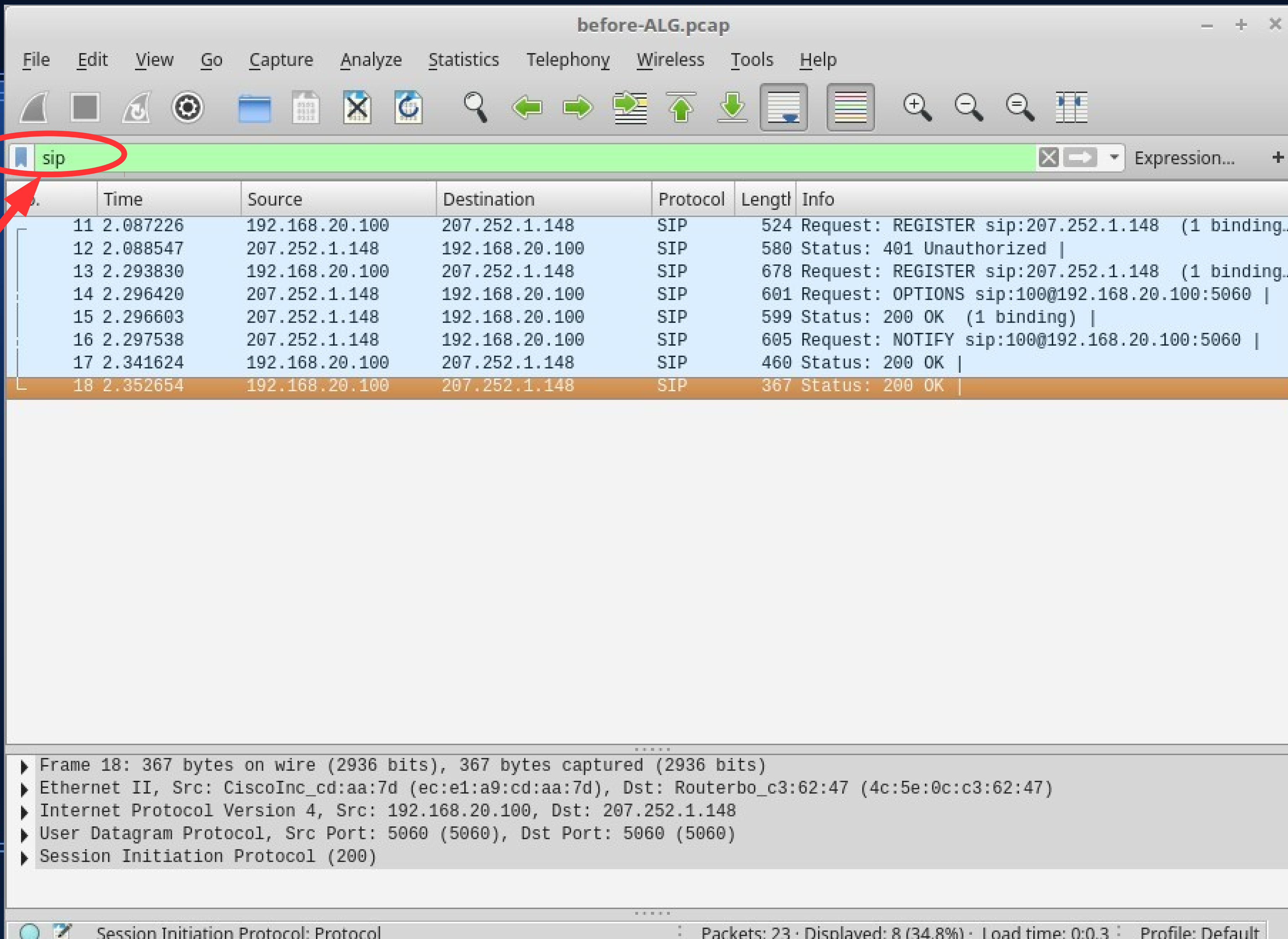
# Decode in wireshark



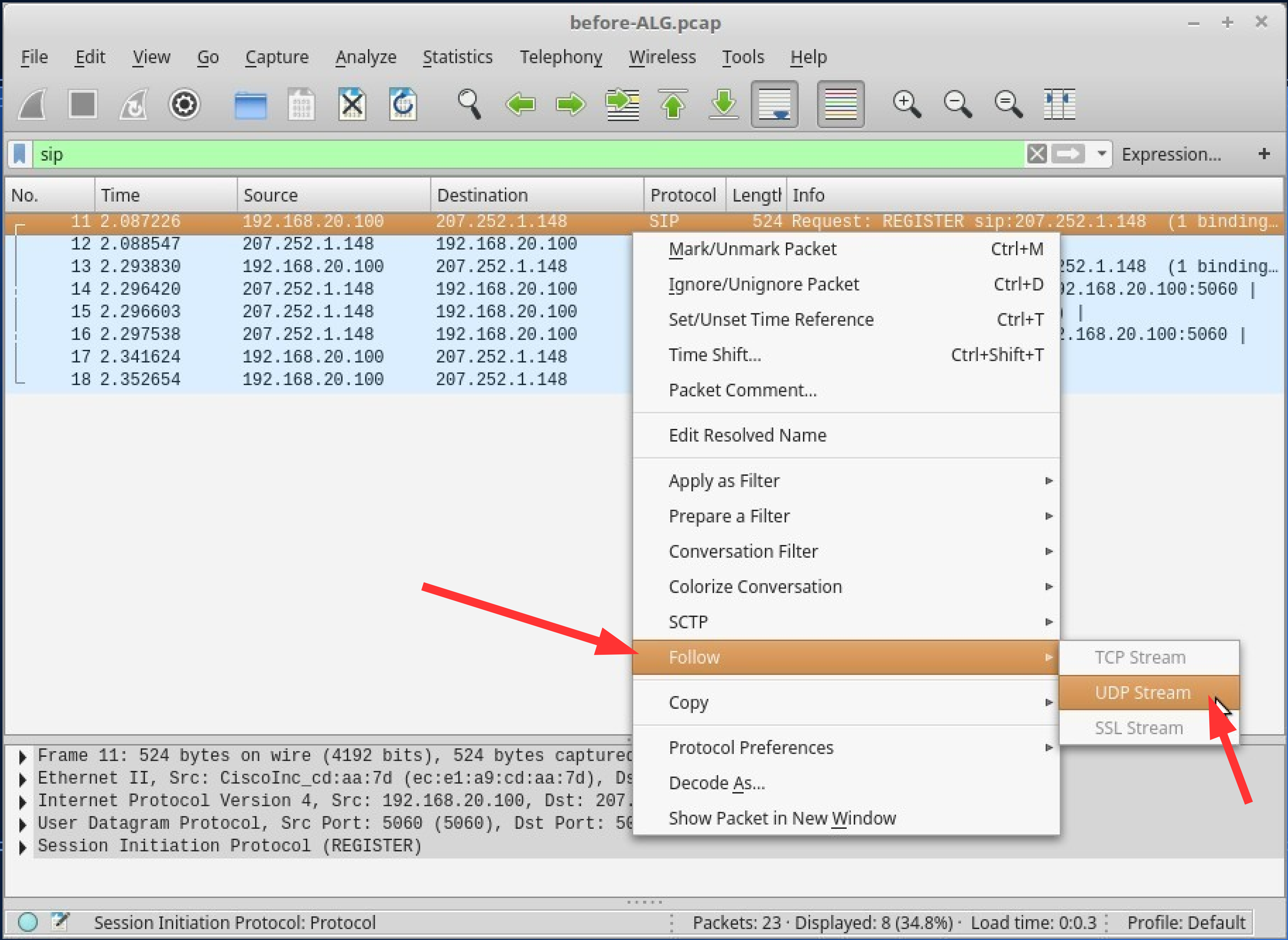

| No. | Time | Source | Destination | Protocol | Lengtl | Info |
|---|---|---|---|---|---|---|
| 1 | 0.000000 | 192.168.20.1 | 192.168.20.100 | DHCP | 357 | DHCP Offer - Transaction ID 0xbed323ae |
| 2 | 1.003062 | 192.168.20.1 | 192.168.20.100 | DHCP | 357 | DHCP ACK - Transaction ID 0xbed323ae |
| 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
```

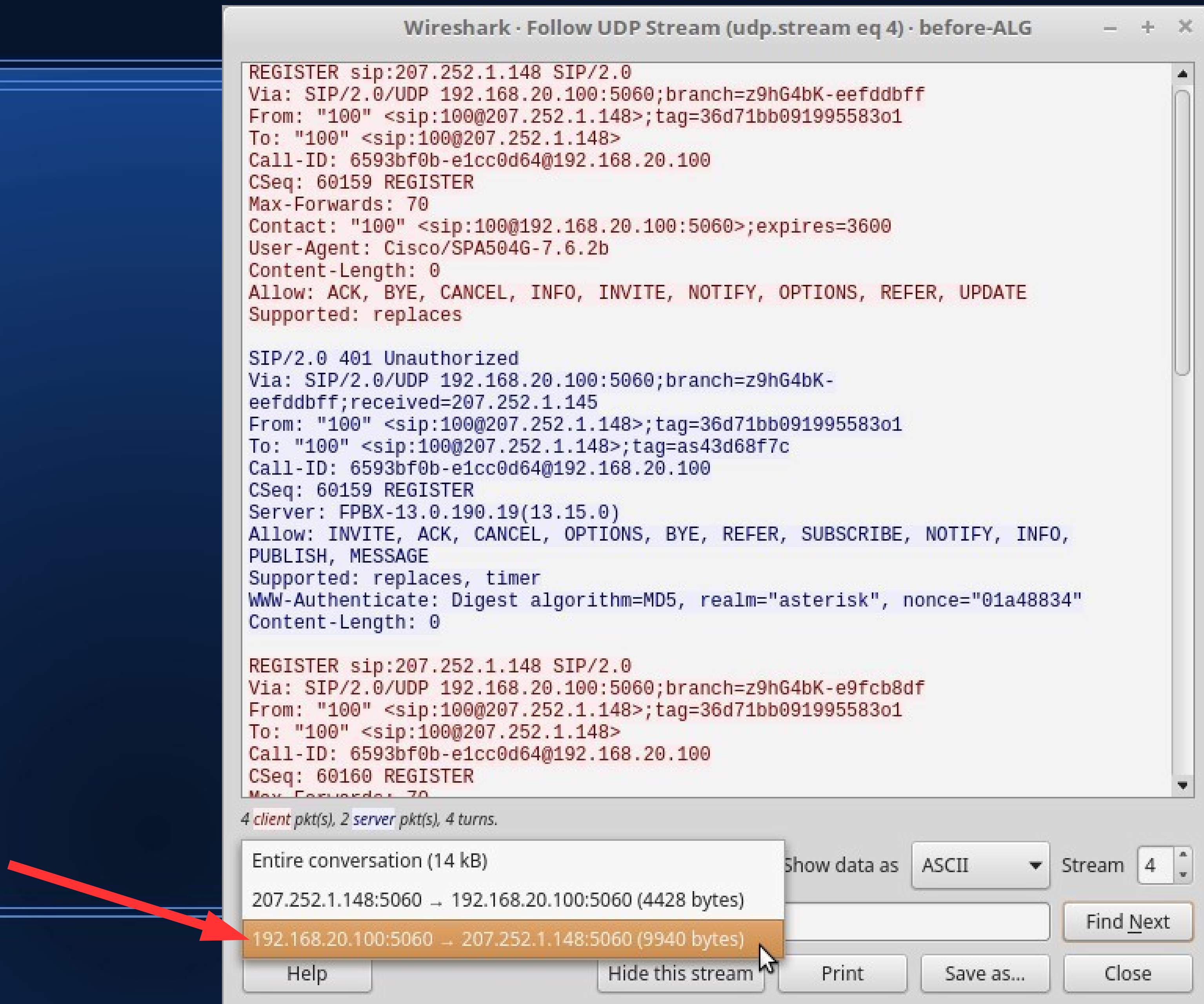
# Decode in wireshark



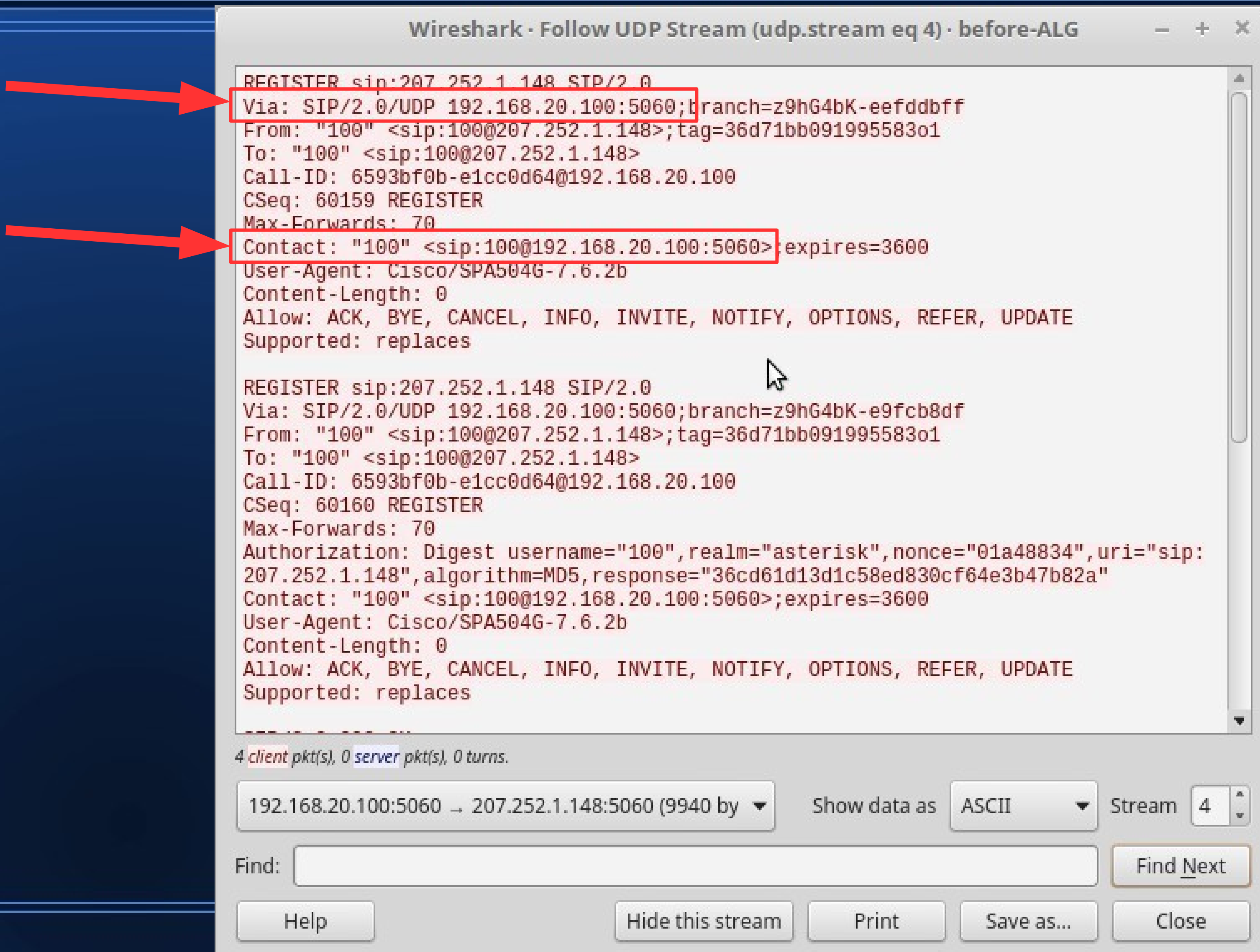
# Decode in wireshark



# Decode in wireshark

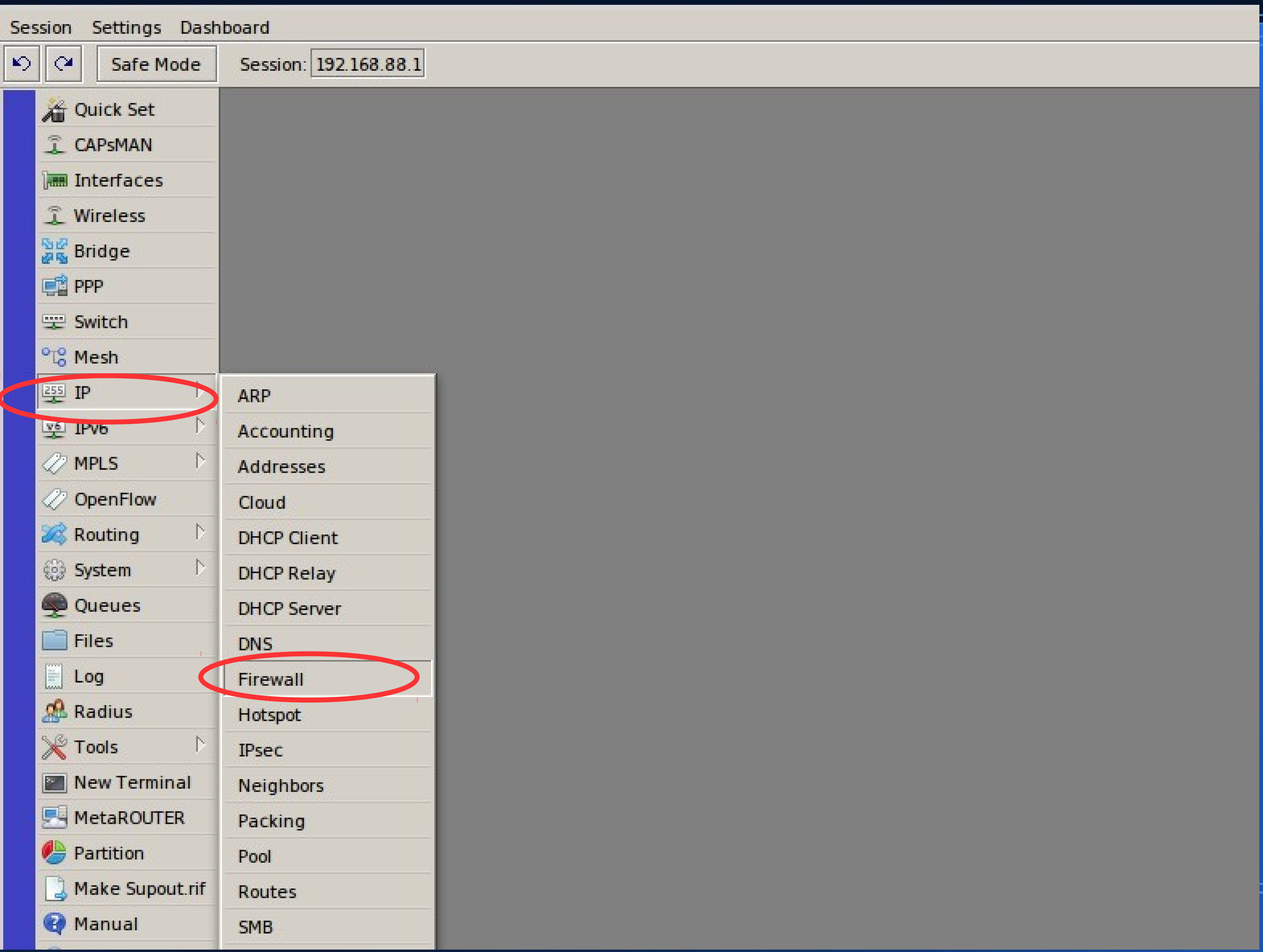


# Decode in wireshark

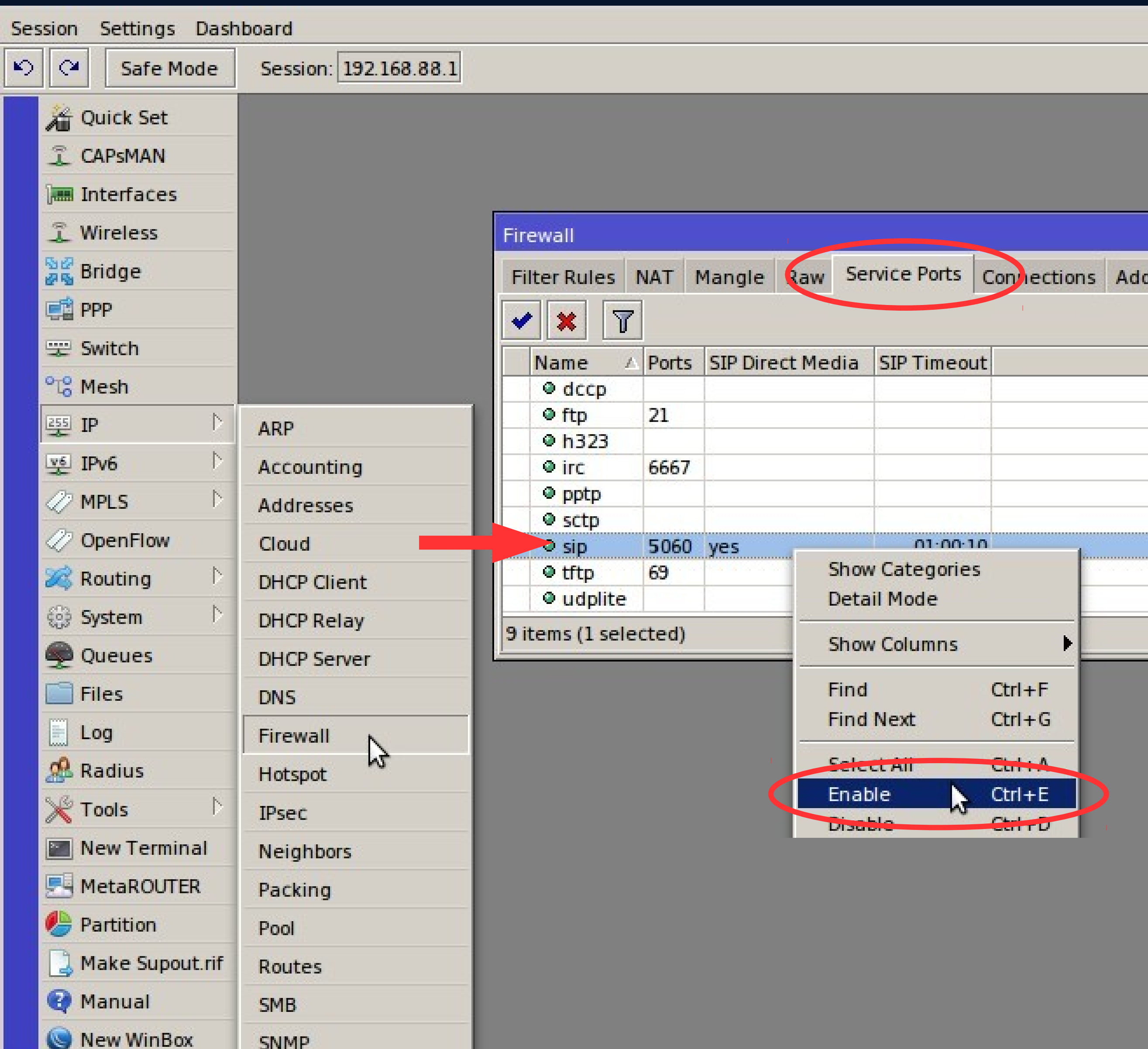


If your SIP device is not NAT Aware
Enable SIP ALG !

# Enable SIP ALG



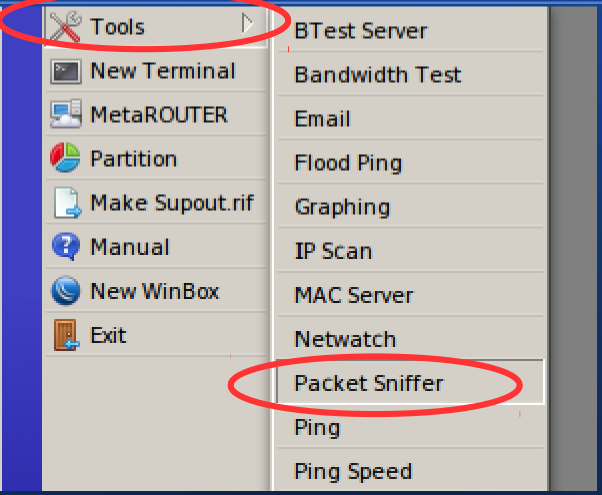
# Enable SIP ALG




/ip firewall service-port
enable sip

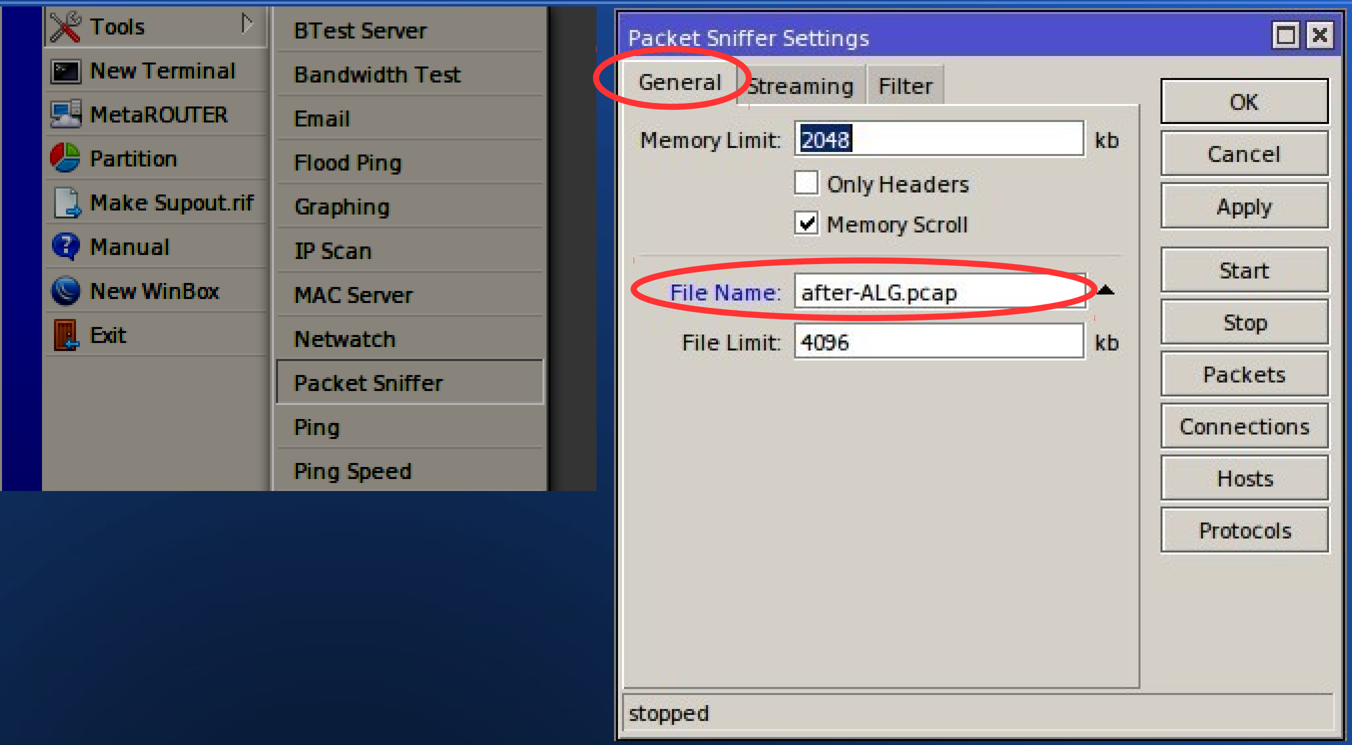
# capture packets after ALG modification

# After ALG modifications pcap




/tool sniffer

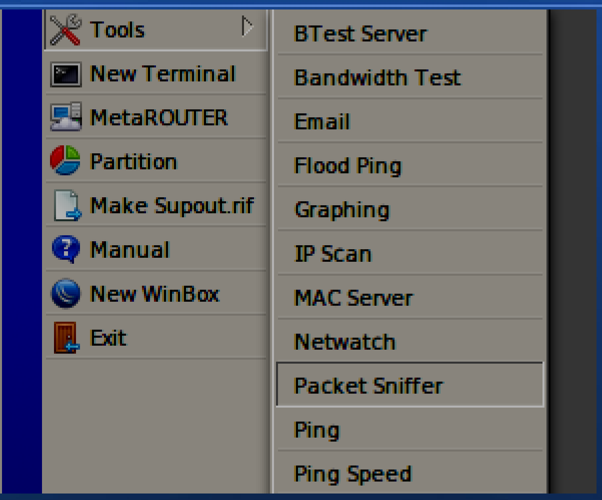
# After ALG modifications pcap

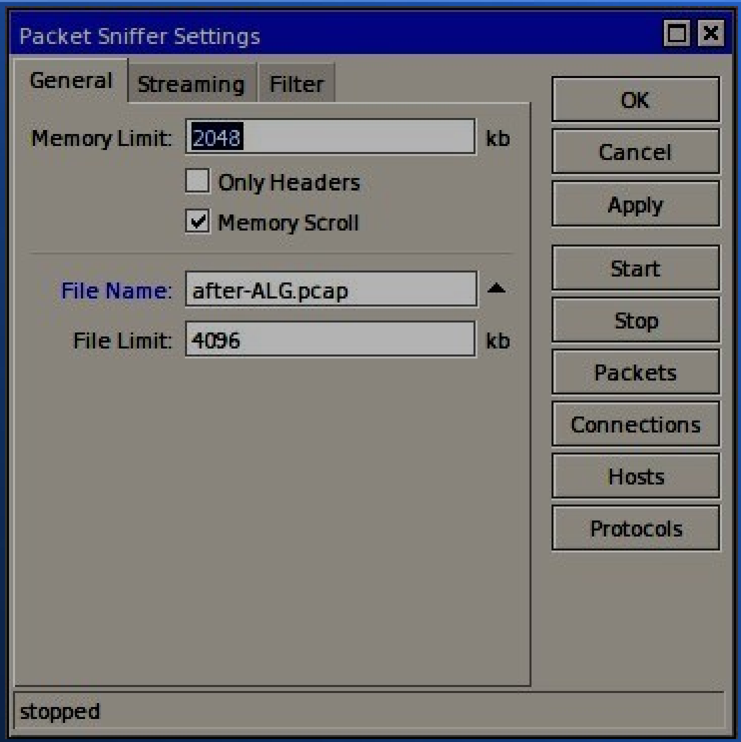


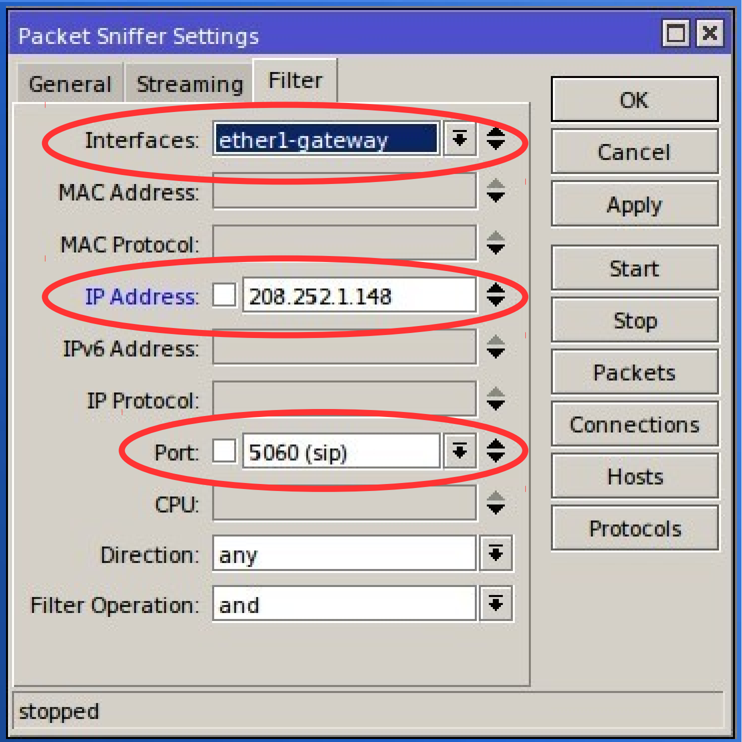

/tool sniffer

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

# After ALG modifications pcap

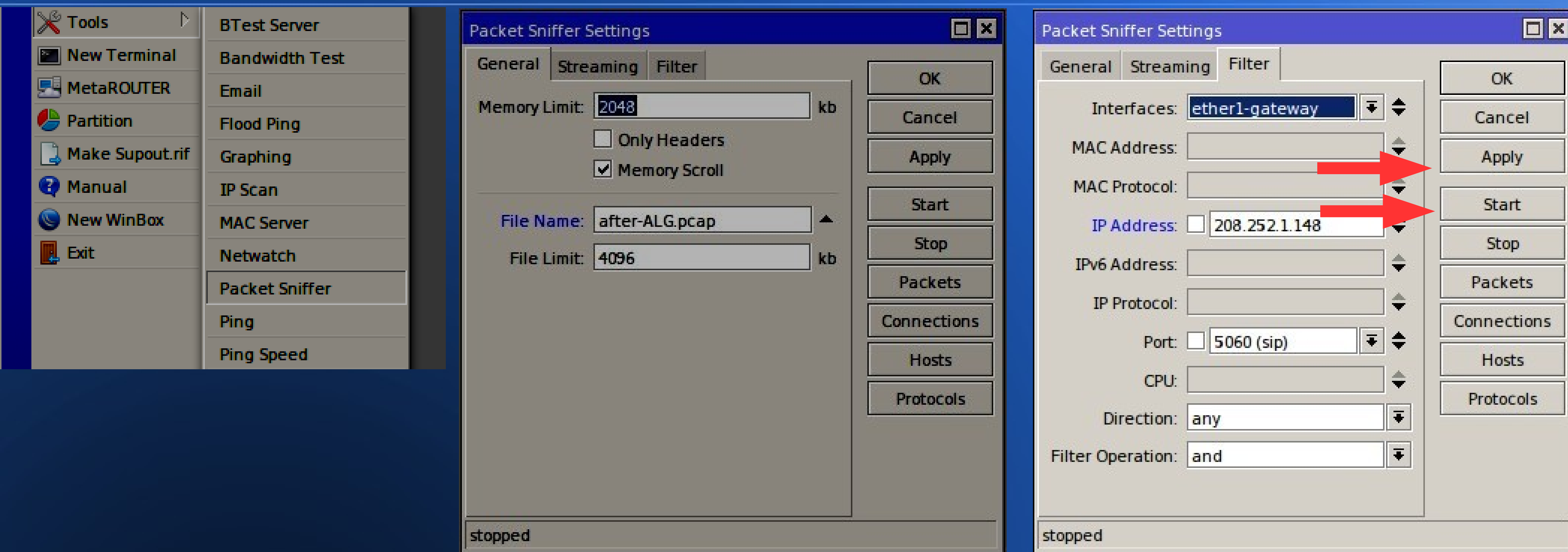



/tool sniffer
set only-headers=no file-name=after-ALG.pcap file-limit=4096 filter-interface=ether1-gateway filter-ip-address=207.252.1.148/32 filter-port=5060 filter-direction=any

# After ALG modifications pcap



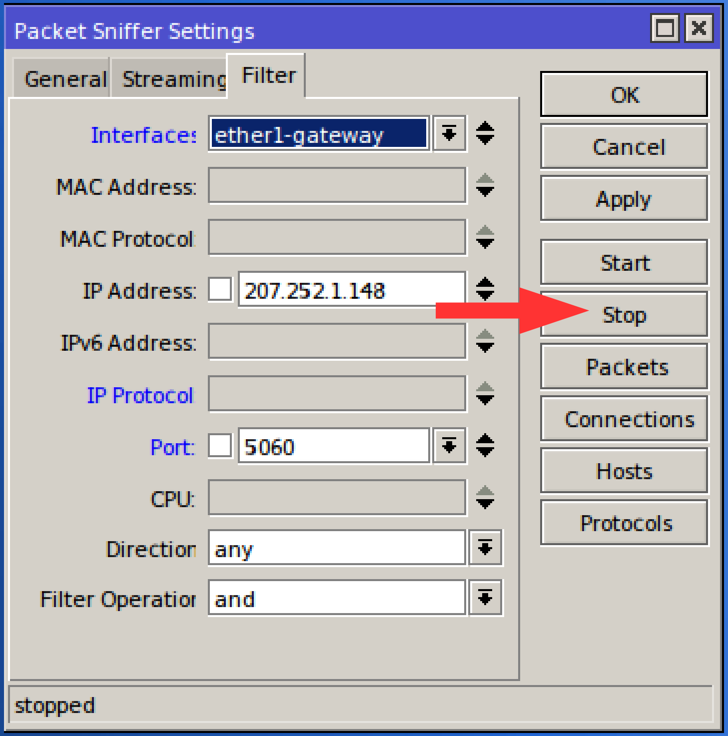

/tool sniffer
set only-headers=no file-name=after-ALG.pcap file-limit=4096 filter-interface=ether1-gateway filter-ip-address=207.252.1.148/32 filter-port=5060 filter-direction=any

start

Generate some traffic while the sniffer is capturing packets.

# After ALG modifications pcap

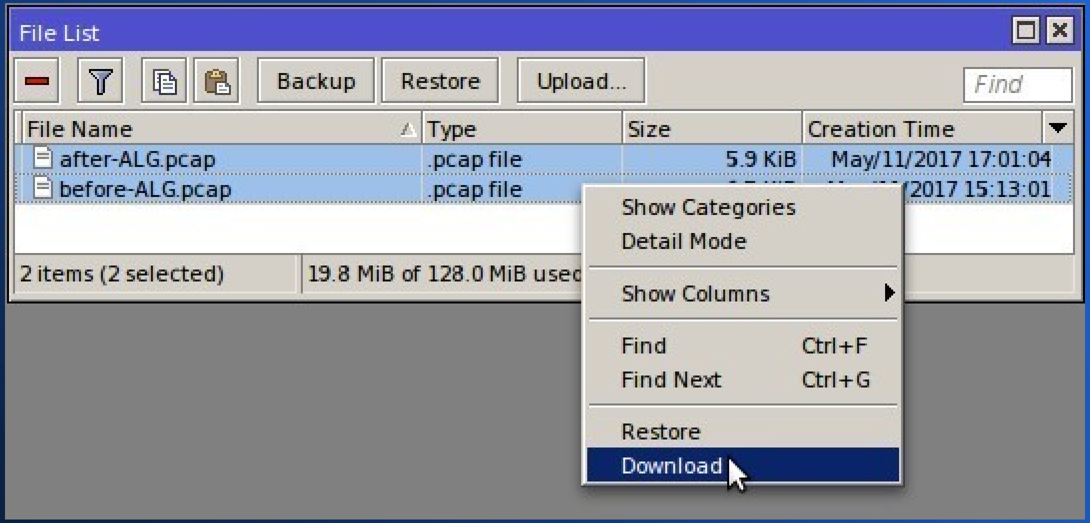
Make sure to stop the sniffer



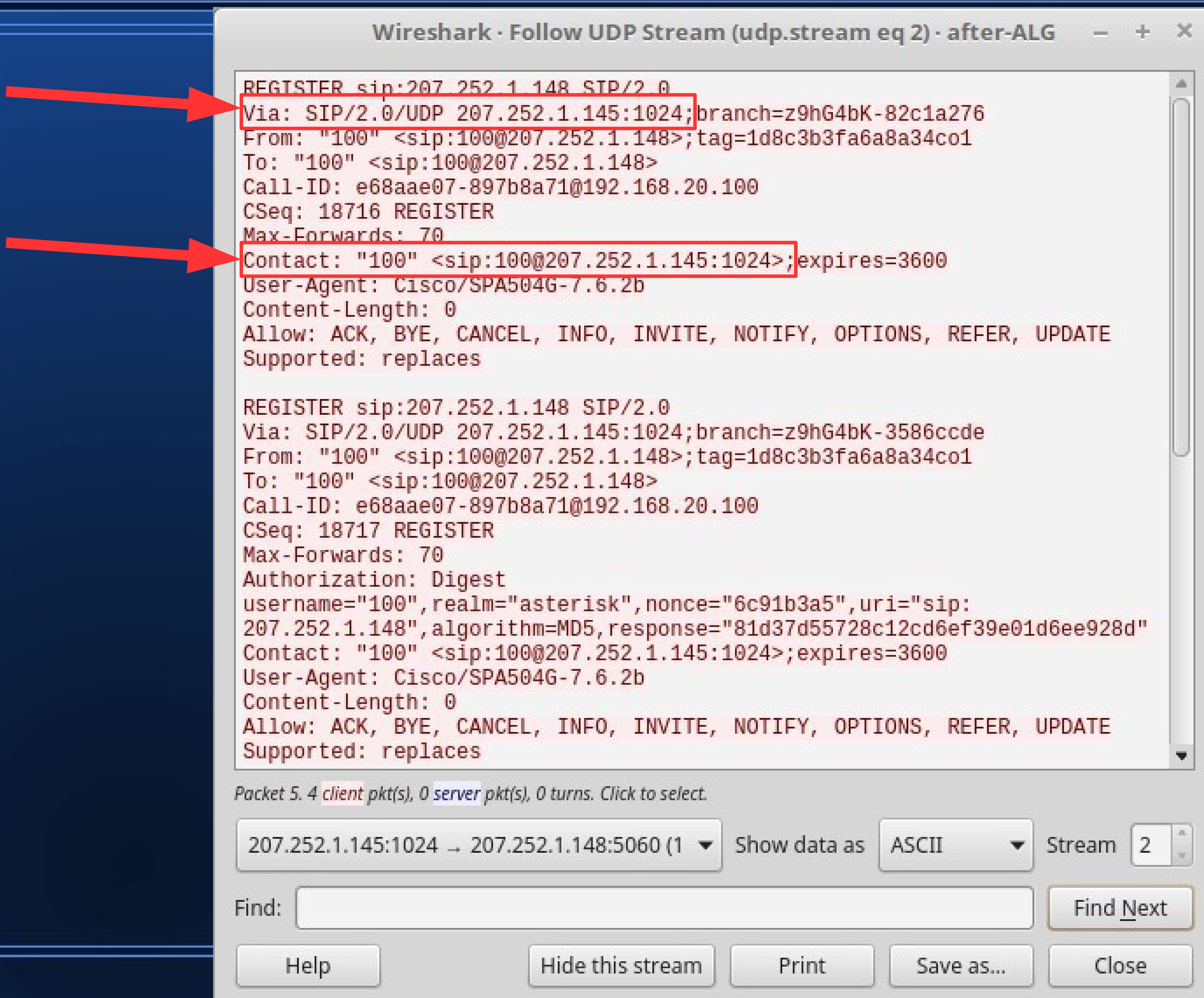

/tool sniffer

stop

# Download pcap files

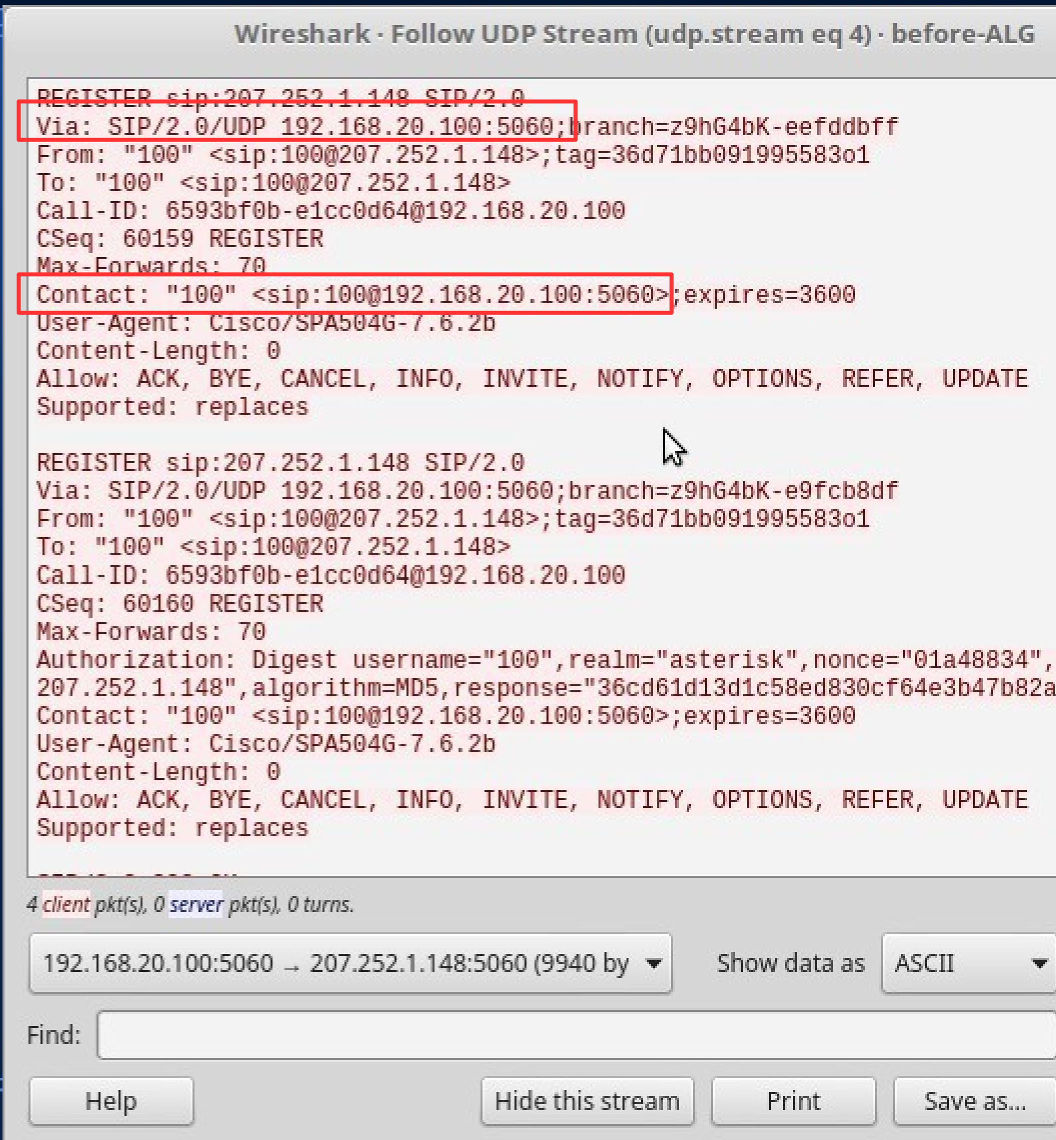


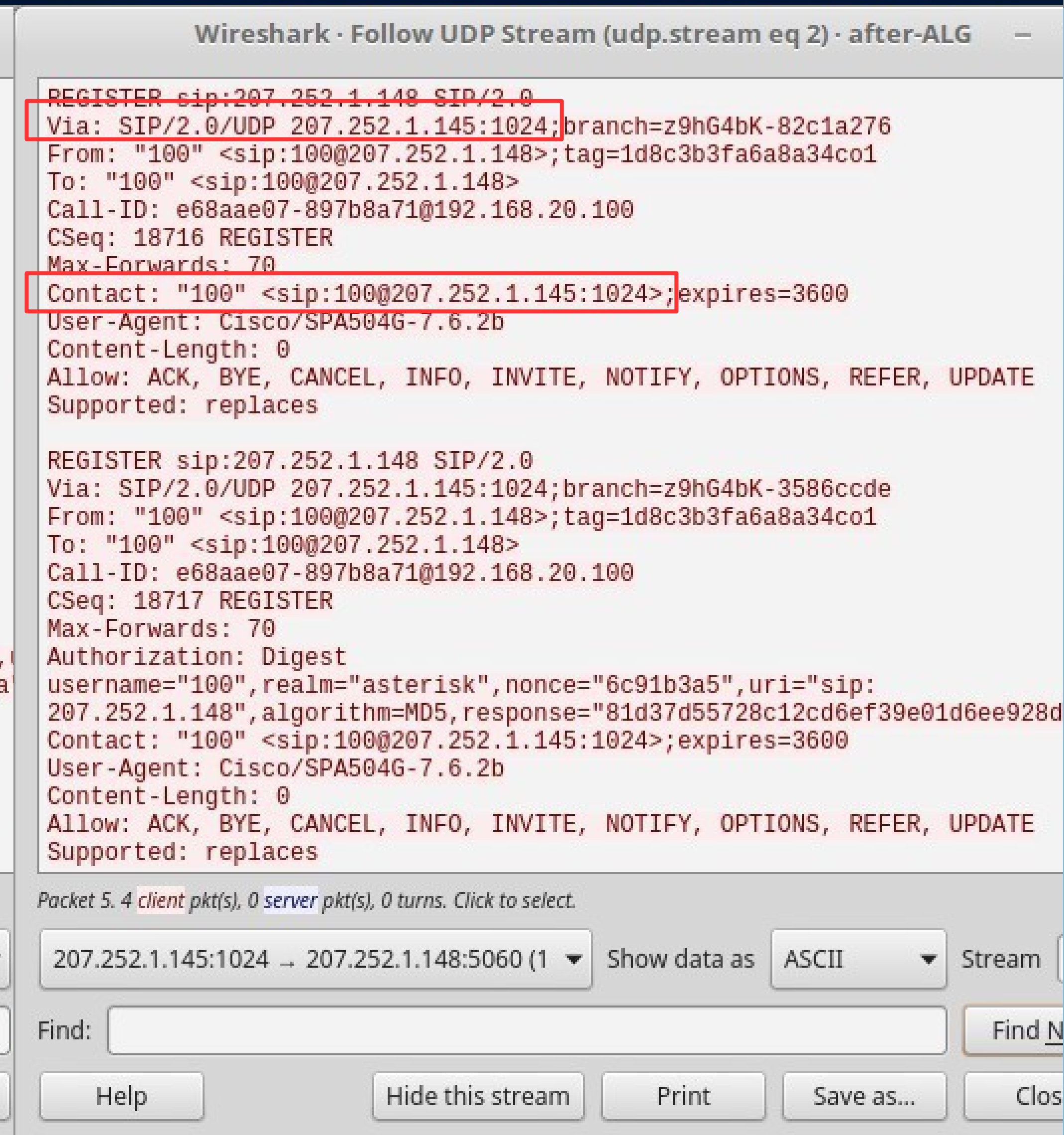
# Decode in wireshark



# ALG Enabled

# Before modification & after modification



# When is SIP ALG unnecessary?

# When is SIP ALG unnecessary?

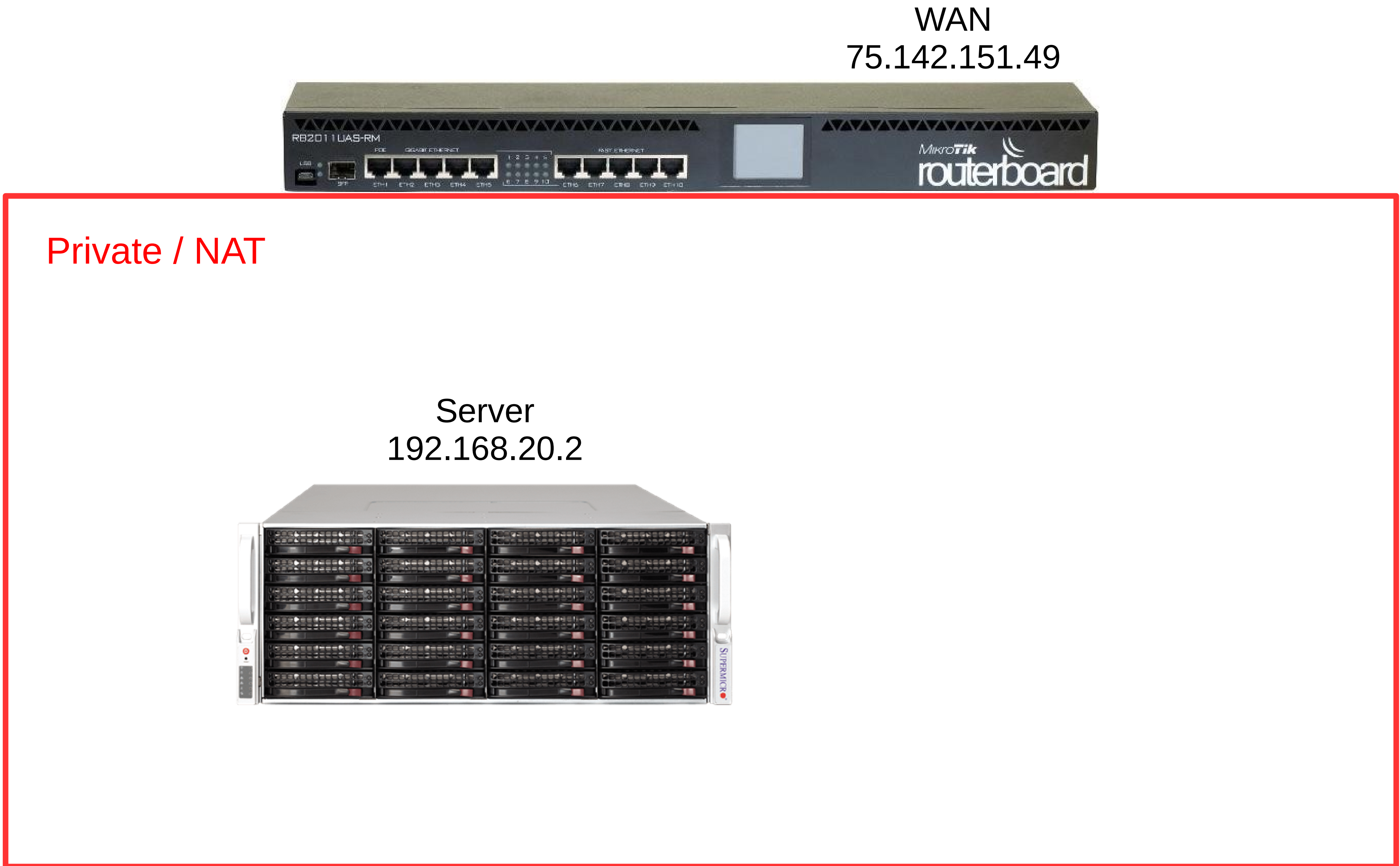
When the SIP device is NAT aware.

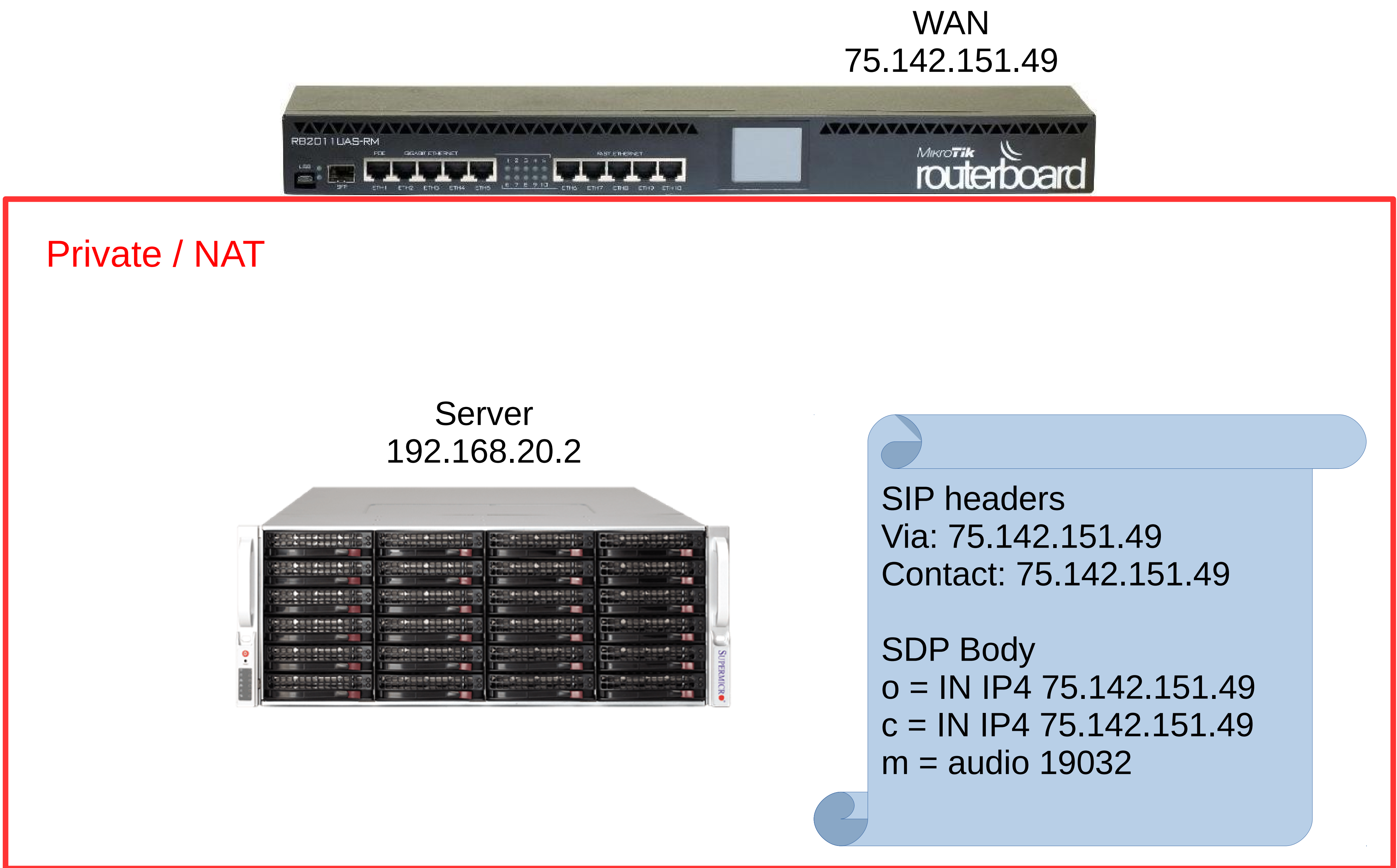
1. Server is behind NAT
2. Server is outside NAT (public server)

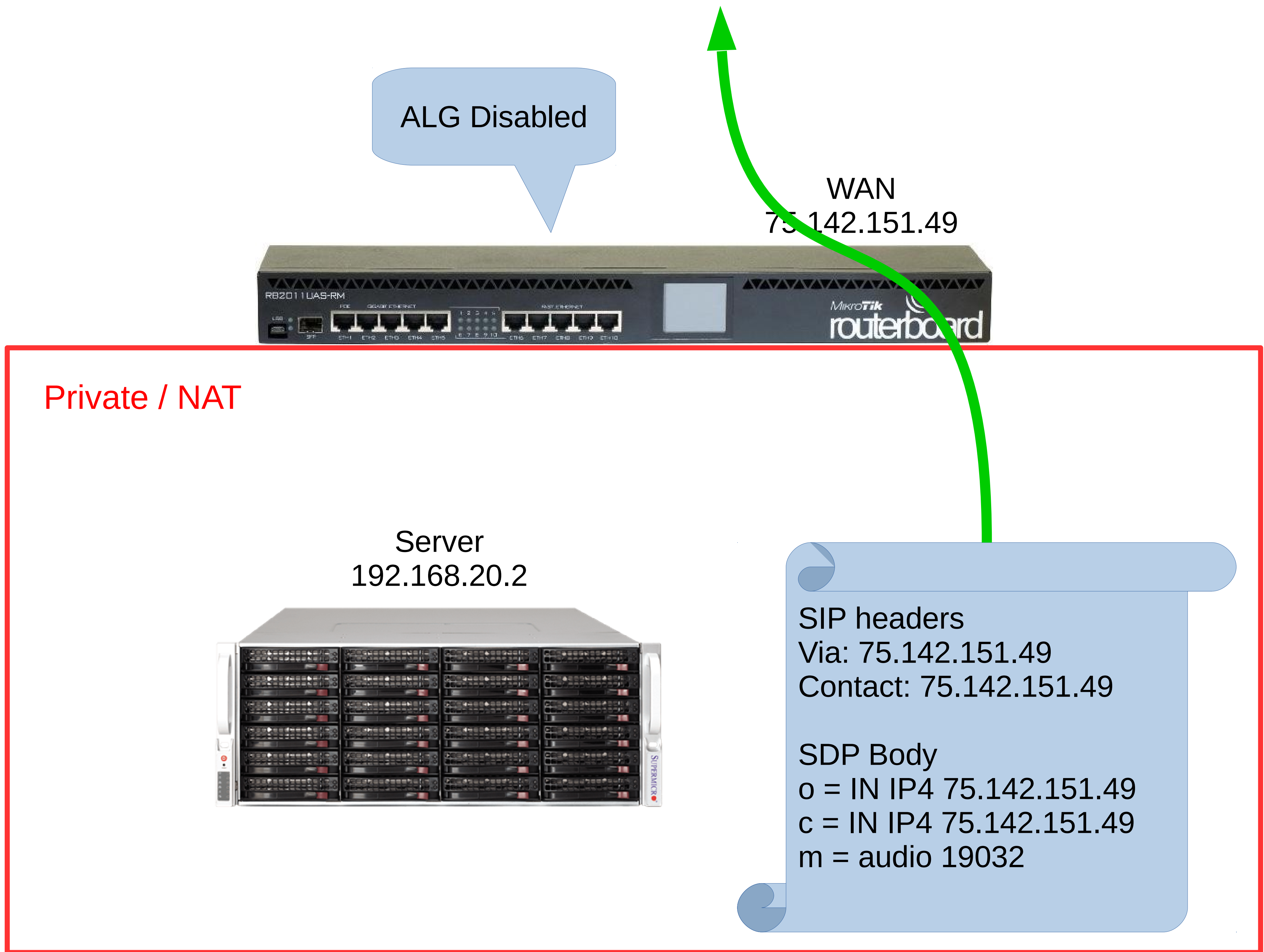
# When is SIP ALG unnecessary?

## 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
RB2011UAS-RM
MikroTik
routerboard
Private / NAT
Server
192.168.20.2
SUPERMICRO


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
RB2011UAS-RM
MikroTik
routerboard
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)

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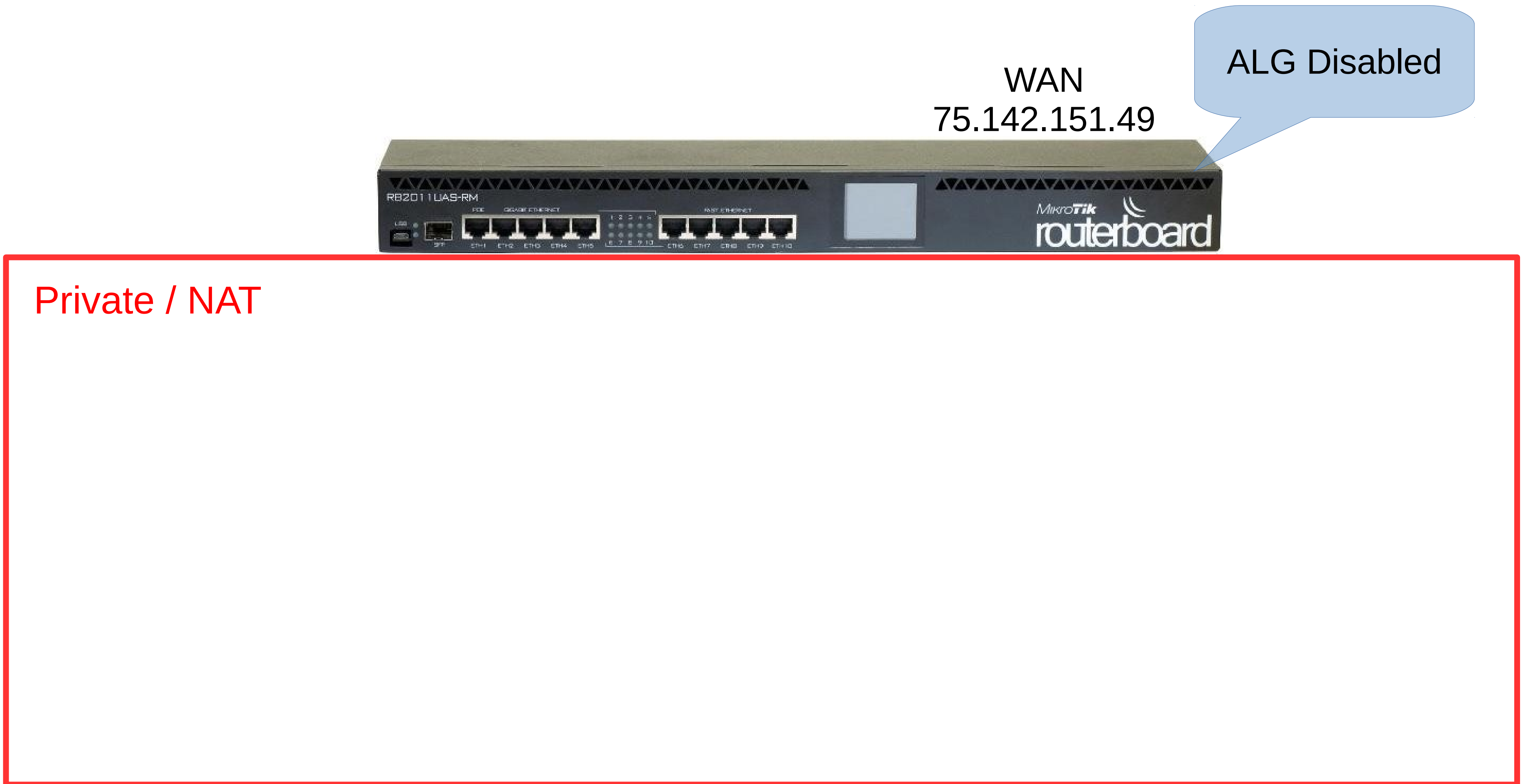
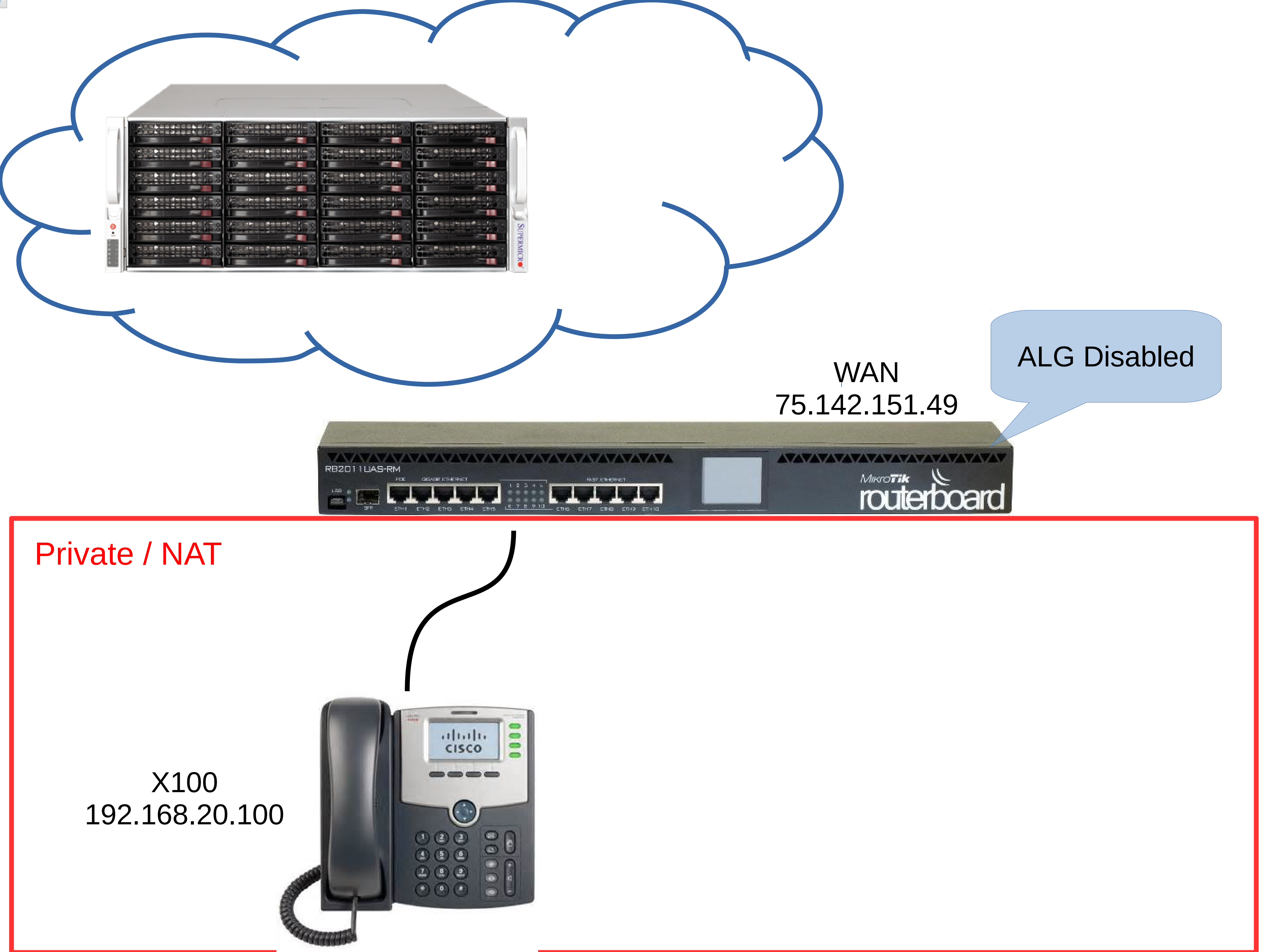
- SIP servers have NAT options for each extension

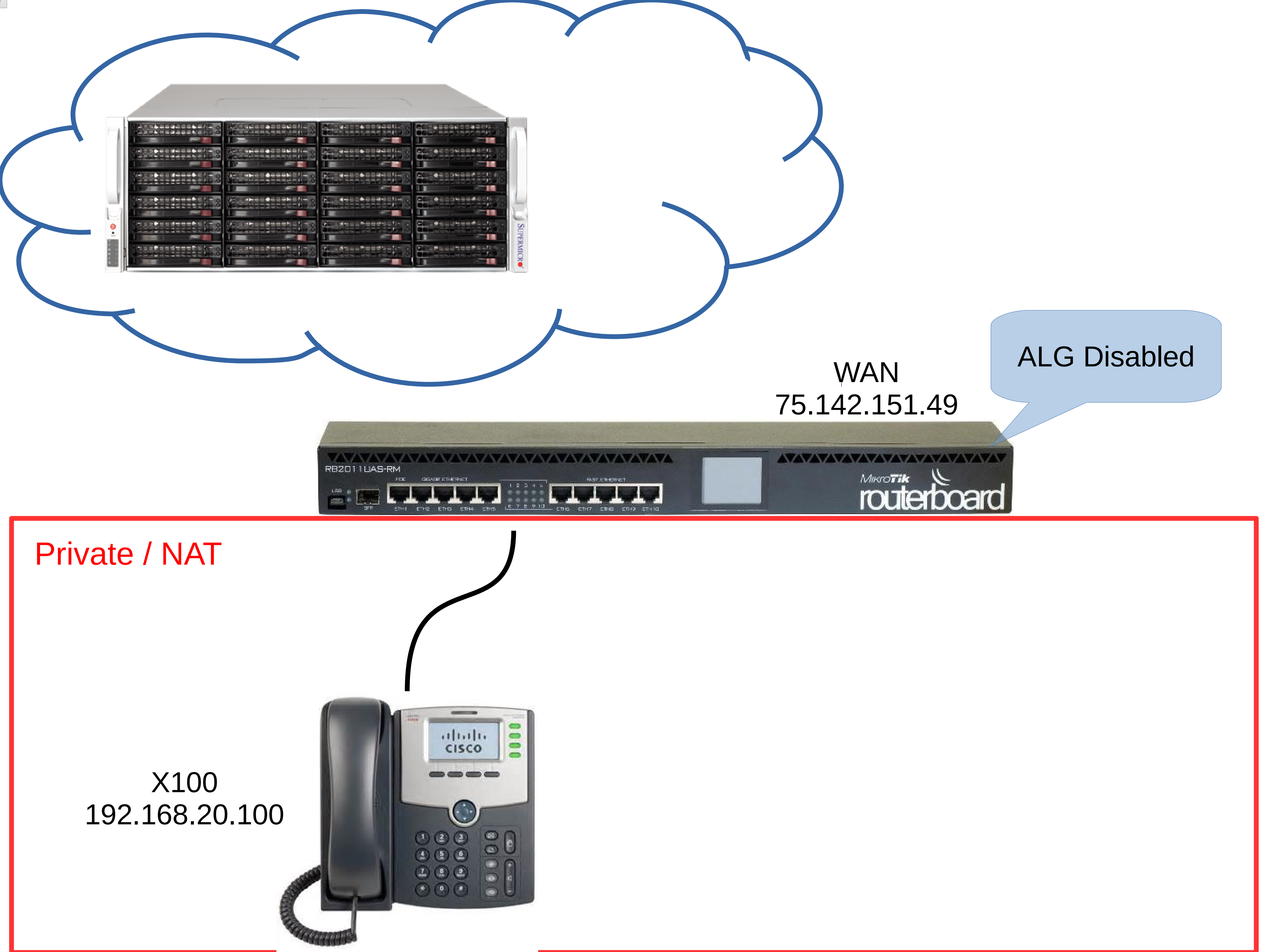
# Servers outside NAT (public server)

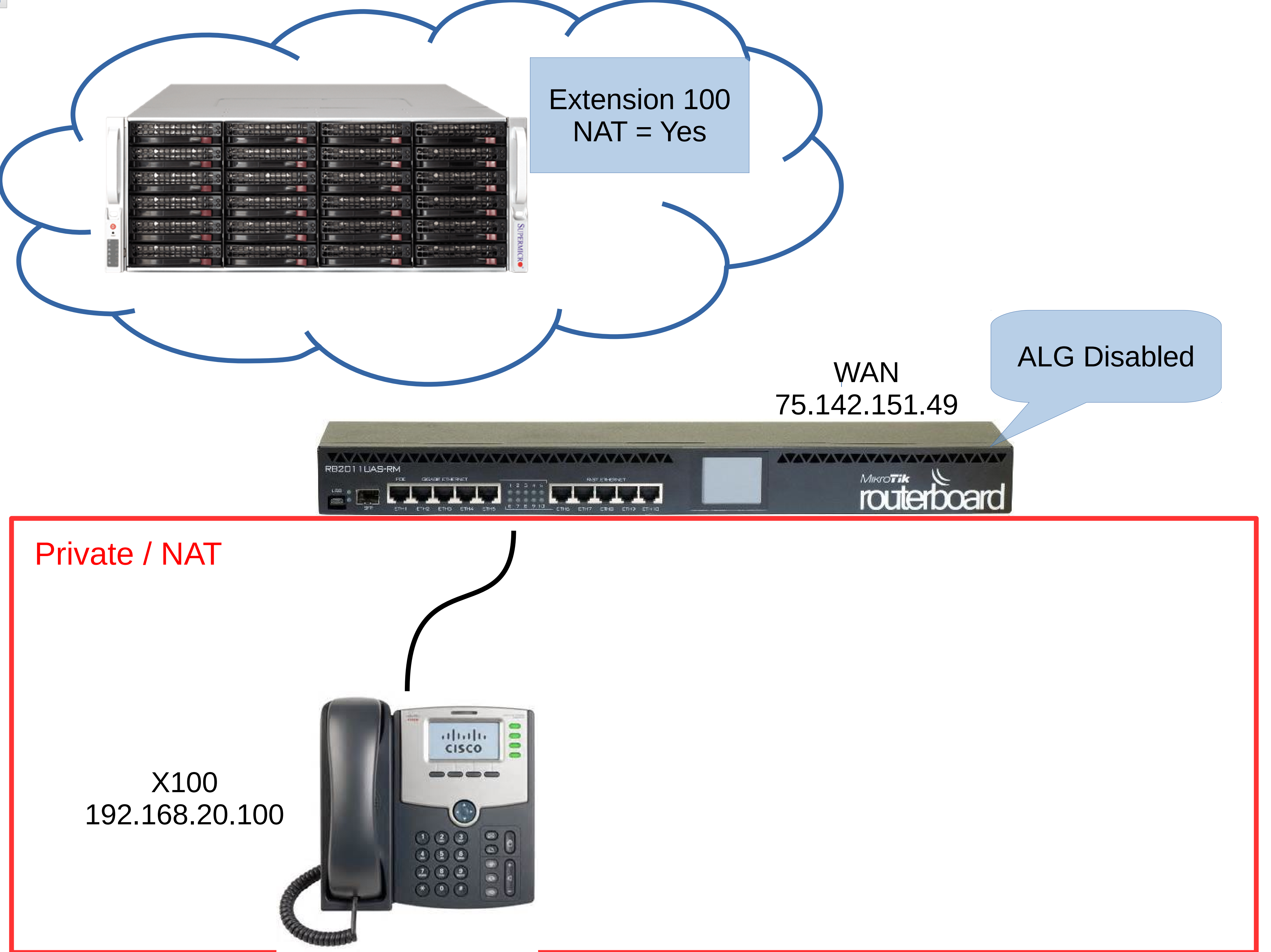
- SIP servers have NAT options for each extension
- If server side extension states NAT=Yes then send all responses to the client originating IP and Port.


WAN
75.142.151.49
ALG Disabled
RB2011UAS-RM
MikroTik
routerboard
Private / NAT

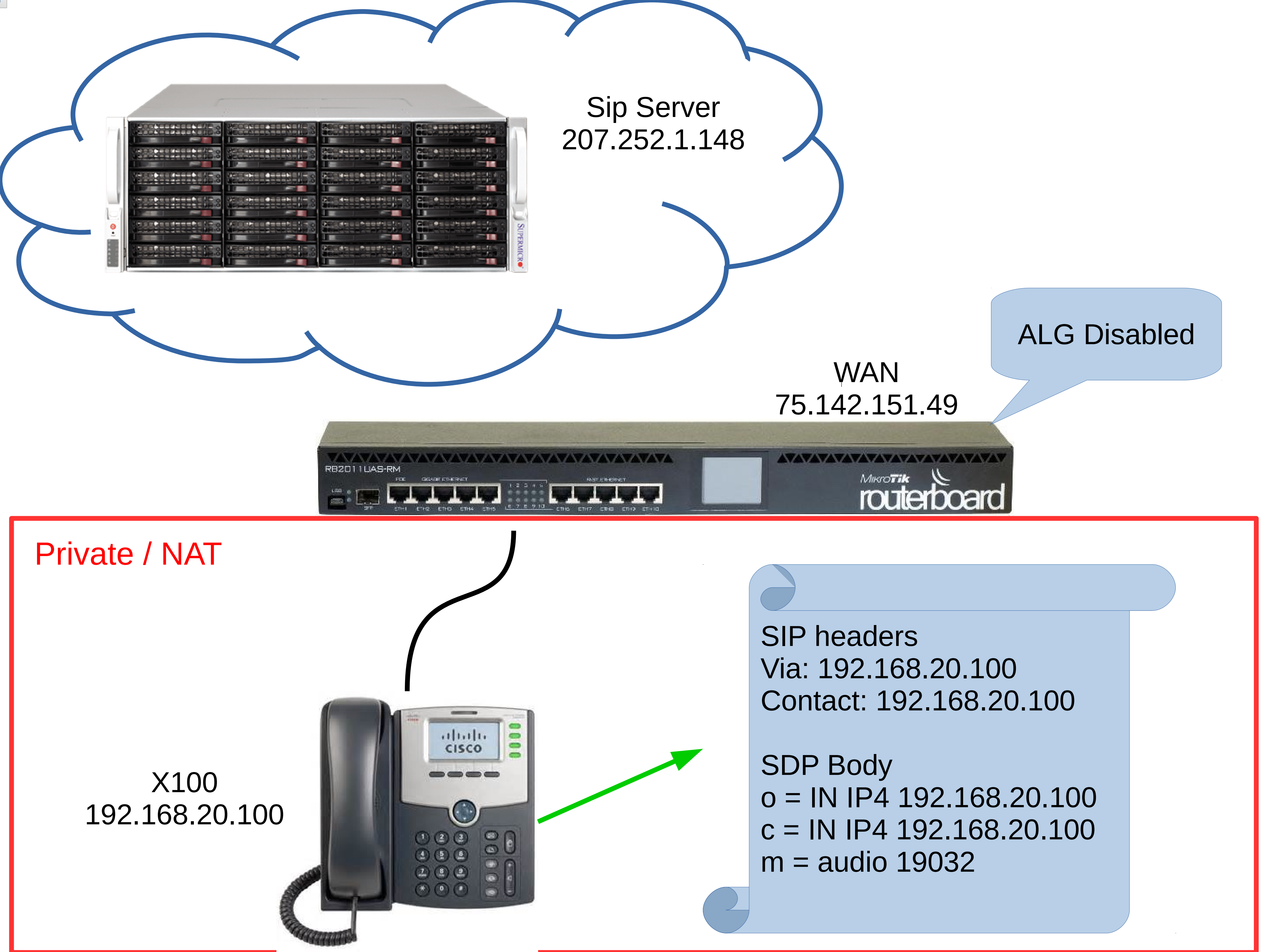

WAN
75.142.151.49
ALG Disabled
RB2011UAS-RM
MikroTik
routerboard
Private / NAT
cisco
X100
192.168.20.100


WAN
75.142.151.49
ALG Disabled
RB2011UAS-RM
MikroTik
routerboard
Private / NAT
X100
192.168.20.100
CISCO

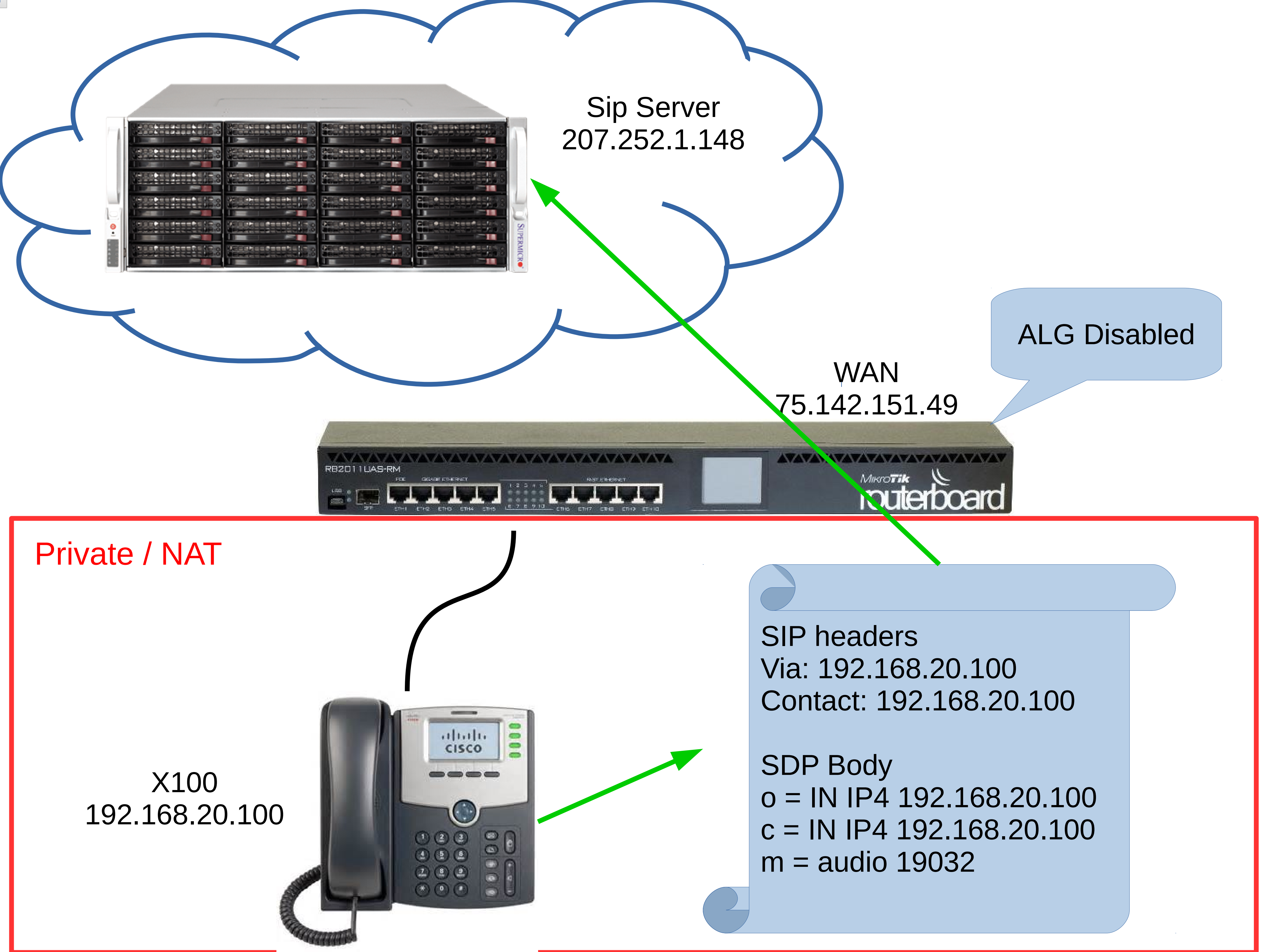

ALG Disabled
WAN
75.142.151.49
RB2011UAS-RM
MikroTik
routerboard
Private / NAT
X100
192.168.20.100
cisco

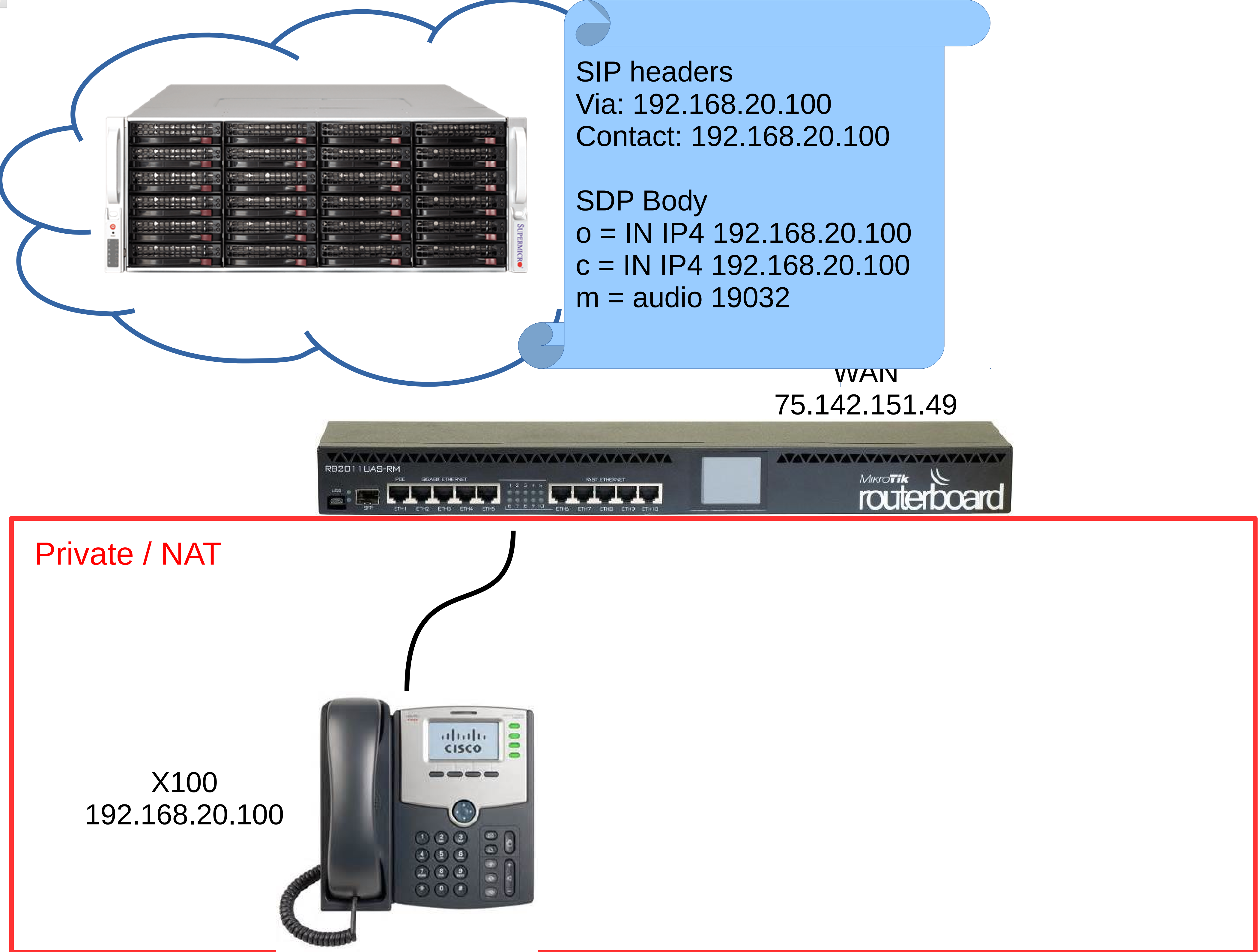


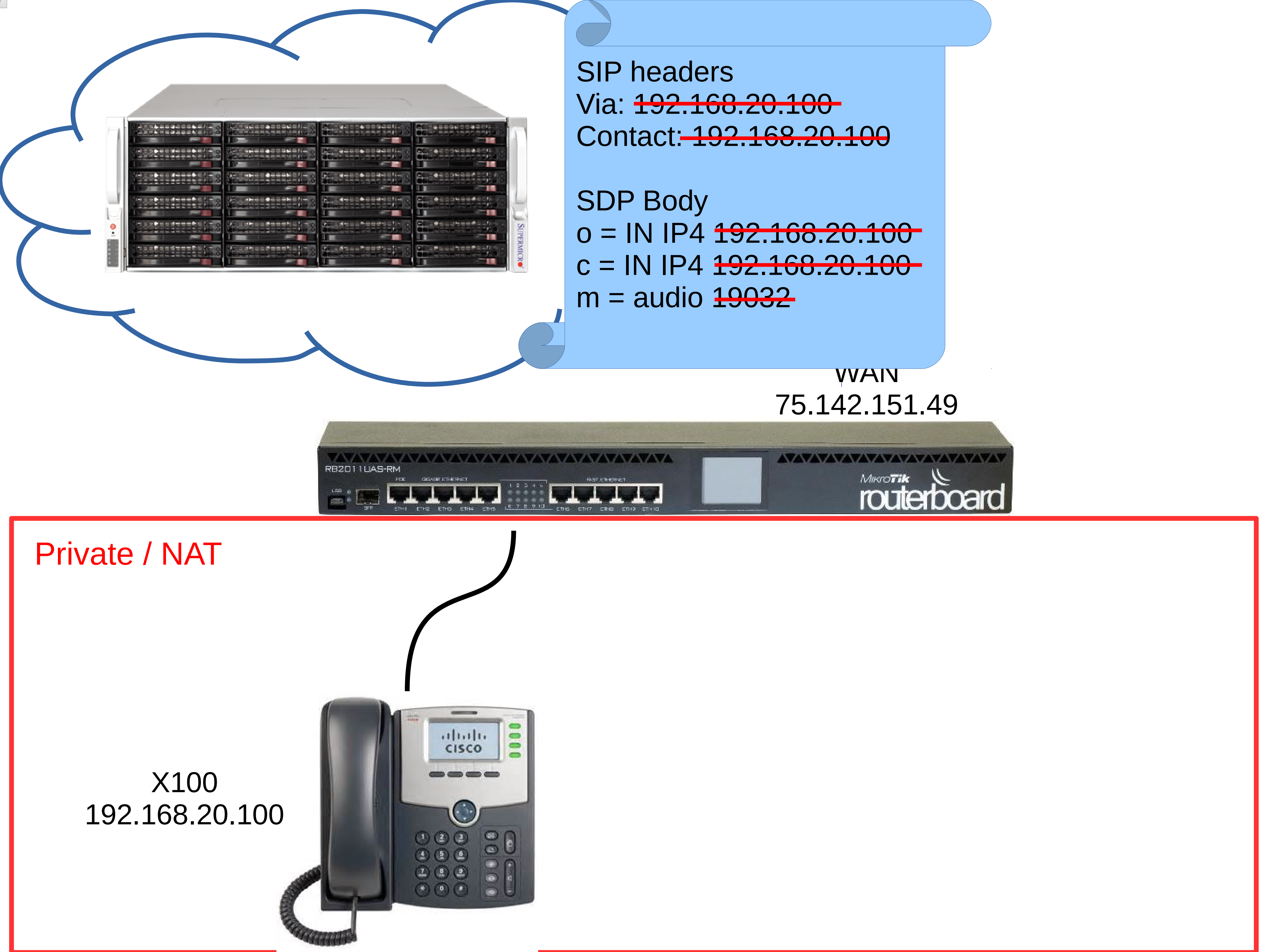
Extension 100
NAT = Yes
SUPERMICRO
WAN
75.142.151.49
ALG Disabled
RB2011UAS-RM
MikroTik
routerboard
Private / NAT
X100
192.168.20.100
cisco

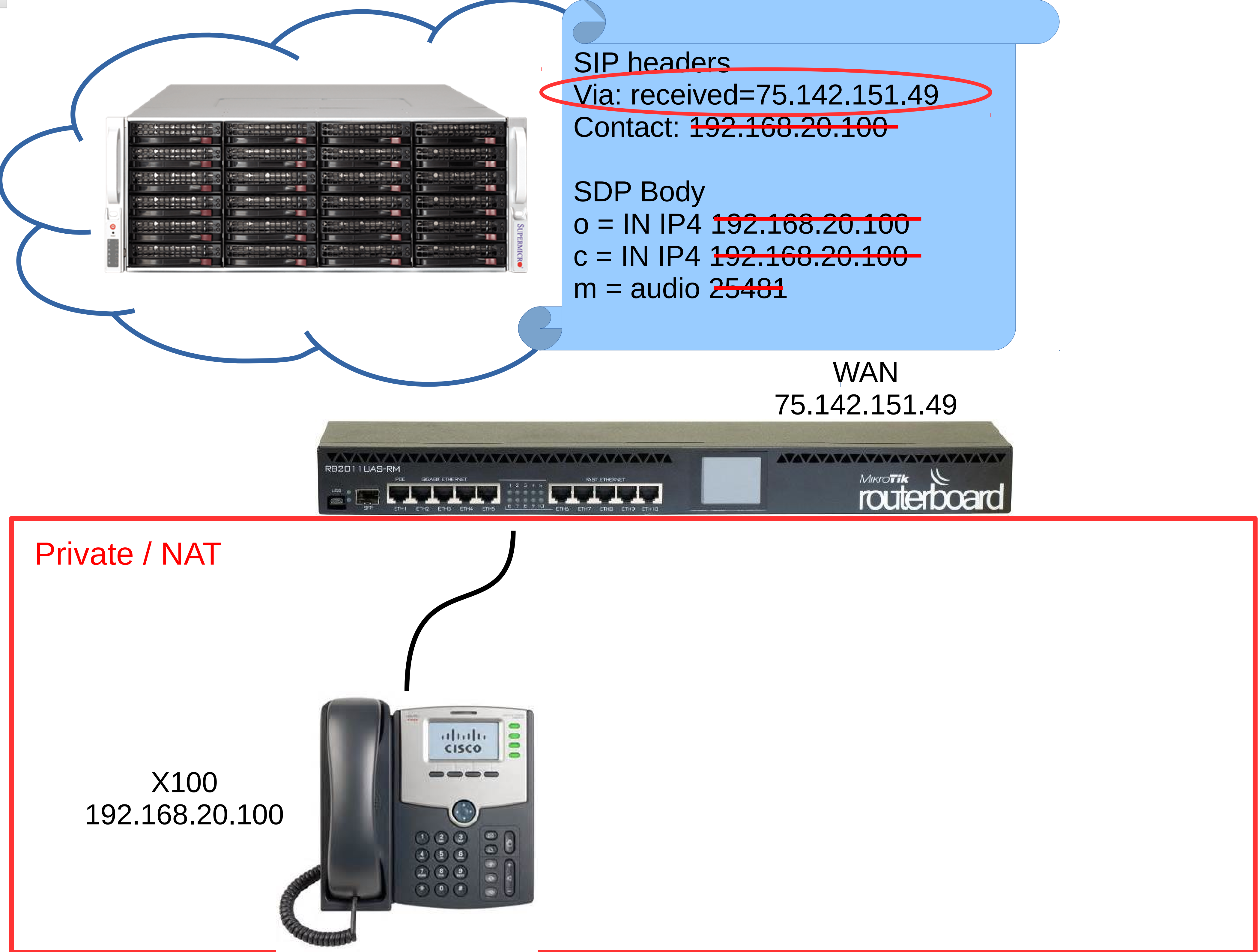


Sip Server
207.252.1.148
ALG Disabled
WAN
75.142.151.49
RB2011UAS-RM
MikroTik
routerboard
Private / NAT
SIP headers
Via: 192.168.20.100
Contact: 192.168.20.100
SDP Body
o = IN IP4 192.168.20.100
c = IN IP4 192.168.20.100
m = audio 19032
X100
192.168.20.100


Sip Server
207.252.1.148
ALG Disabled
WAN
75.142.151.49
RB2011UAS-RM
MikroTik
routerboard
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
RB2011UAS-RM
MikroTik
routerboard
Private / NAT
X100
192.168.20.100
cisco


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
RB2011UAS-RM
MikroTik
routerboard
Private / NAT
X100
192.168.20.100
CISCO


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
RB2011UAS-RM
MikroTik
routerboard
Private / NAT
X100
192.168.20.100
cisco

# 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=191914b06beo0

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:**

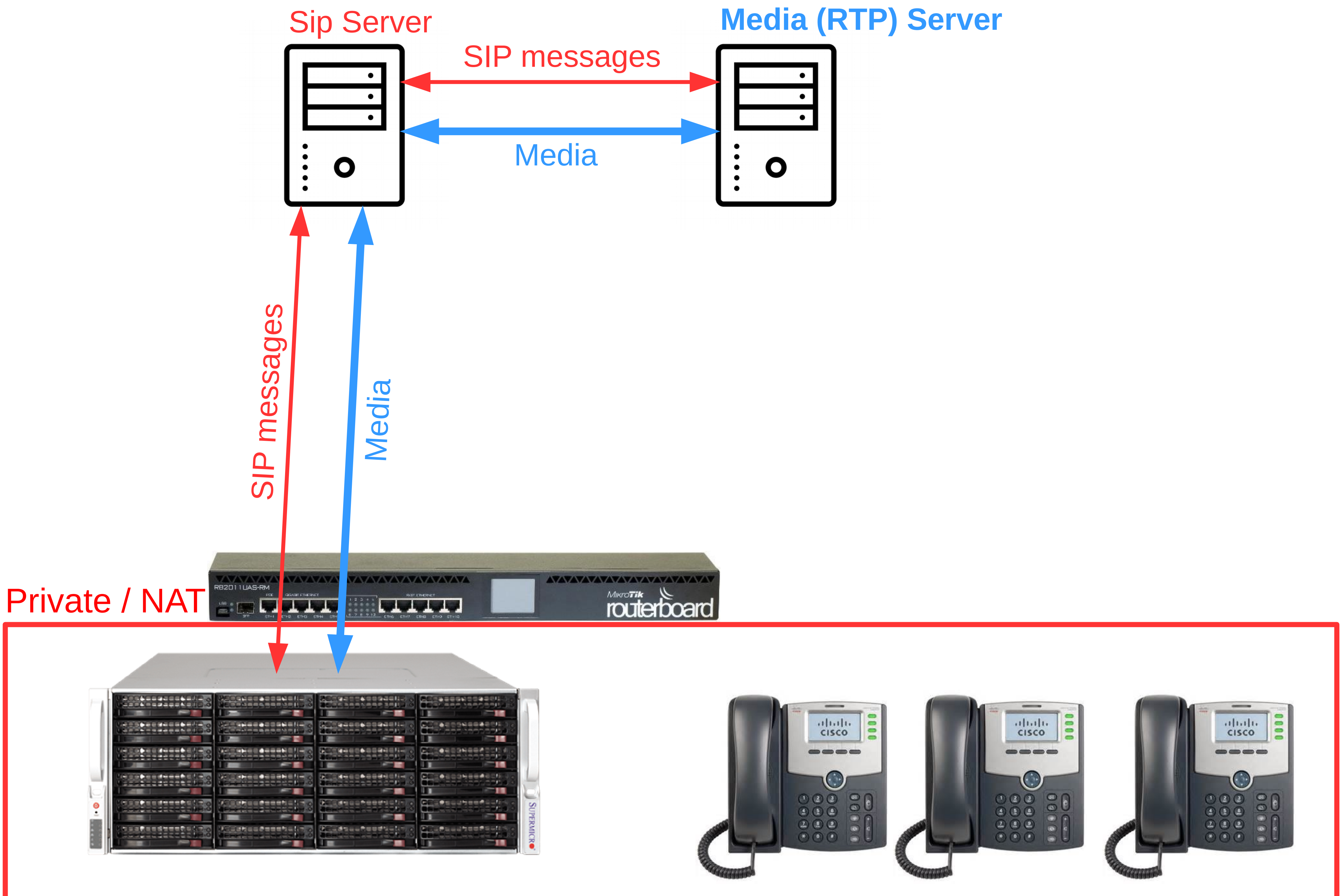
- Manually set SIP ALG timeout
- Set it higher than your lowest sip keepalive message interval (register, invite, options)

# SIP Direct Media

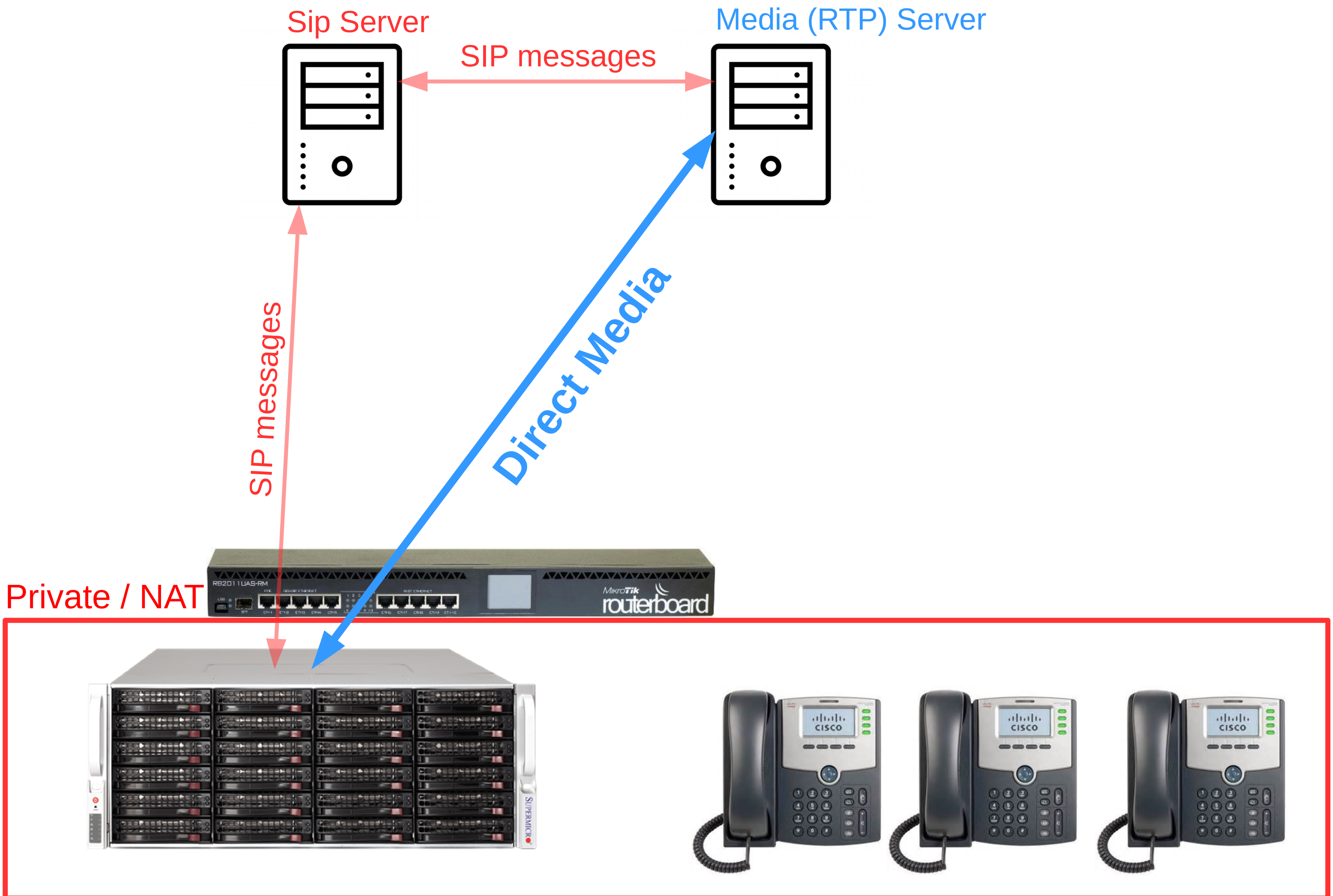
# SIP Direct Media

- 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



# Direct-Media



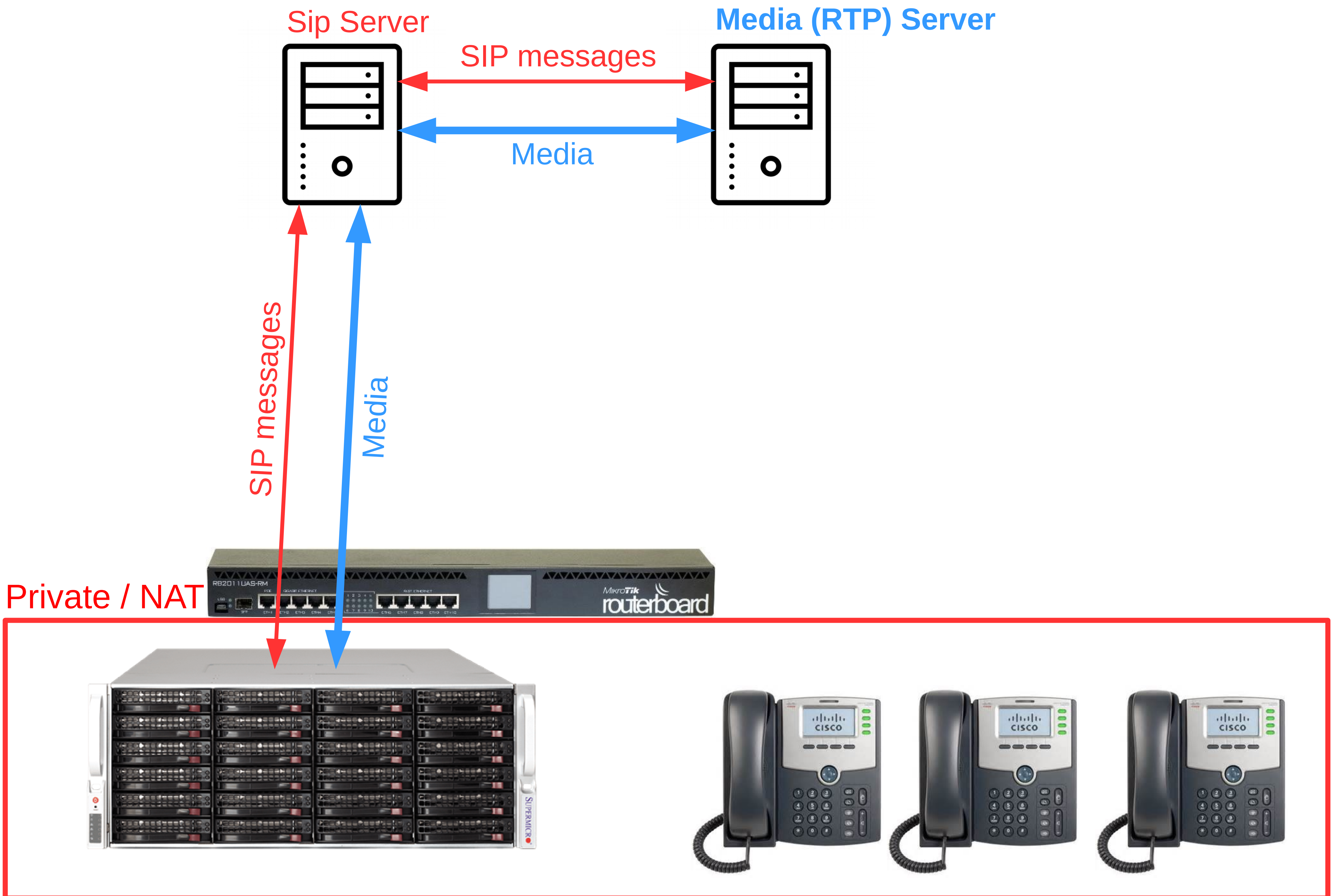
# SIP Direct Media

- 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



# 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 un-necessary?
4- How SIP ALG corrects problems.
5- Testing with wireshark
6- SIP ALG Timeout
7- SIP ALG direct-media