Common VoIP problems, How to detect, correct and avoid them.

Who am I ?

- David Attias
- Installing VoIP systems for over 7 years
- Mikrotik user for 5 years
- Mikrotik certifications MTCNA, MTCRE & MTCWE

Purpose of this lecture

To inform Mikrotik users on how to identify and resolve voip problems

Agenda

1) Identify factors which will affect VoIP call quality
2) Correct call quality issues with RouterOS QoS
    - Marking packets with Mangle
    - Shaping VoIP traffic with Queues
3) Detecting VoIP call quality problems
    - Check for dropped packets
    - Using RouterOS packet sniffer & wireshark
4) Avoid call quality issues

What is VoIP?

Several protocols used together to send and receive REAL TIME voice calls through an IP network(s).

Identify factors that affect call quality

Considerations about VoIP call quality

- VoIP calls are REAL TIME!!
- Connection between phones and voip servers must have low delay and very low Jitter.
- Must have enough available symmetrical bandwidth.

g711 uLaw codec = 87.2k per channel

[20ms voice payload per packet]

Sip = 65k (max sip message size)

The Problems

What can affect call quality?

*Not considering hardware problems

- High jitter levels
  - What is Jitter? Packet Delay Variation / The time lapse between each packet for a given data stream
- Packet Loss
- Delay

In the real world.

- Jitter, packet loss and delay can happen anywhere between the phone and (hosted) server.
- However, 90% of call quality issues happen at the customer location.

Why? Because customer networks are rarely configured properly – if configured at all for VoIP QoS.
USE MIKROTIK!!!

Correcting issues with RouterOS QoS

Quality Of Service (QoS)

- Techniques that categorize and prioritize packets
- Ensures sufficient bandwidth, controls latency, jitter, and reduces data loss.
- Regulate data flows by classifying, scheduling, and marking packets based on priority and by shaping traffic

MikroTik Mangle

Mikrotik Mangle

- Mangle is a Mikrotik routerOS facility that marks packets for future processing.
- The mangle marks exist only within the router.
- Also used to modify some fields in the IP header, like DSCP and TTL fields
- Only 1 packet mark per packet
  Only 1 connection mark per packet

To conserve processor resources:

First mark the connection

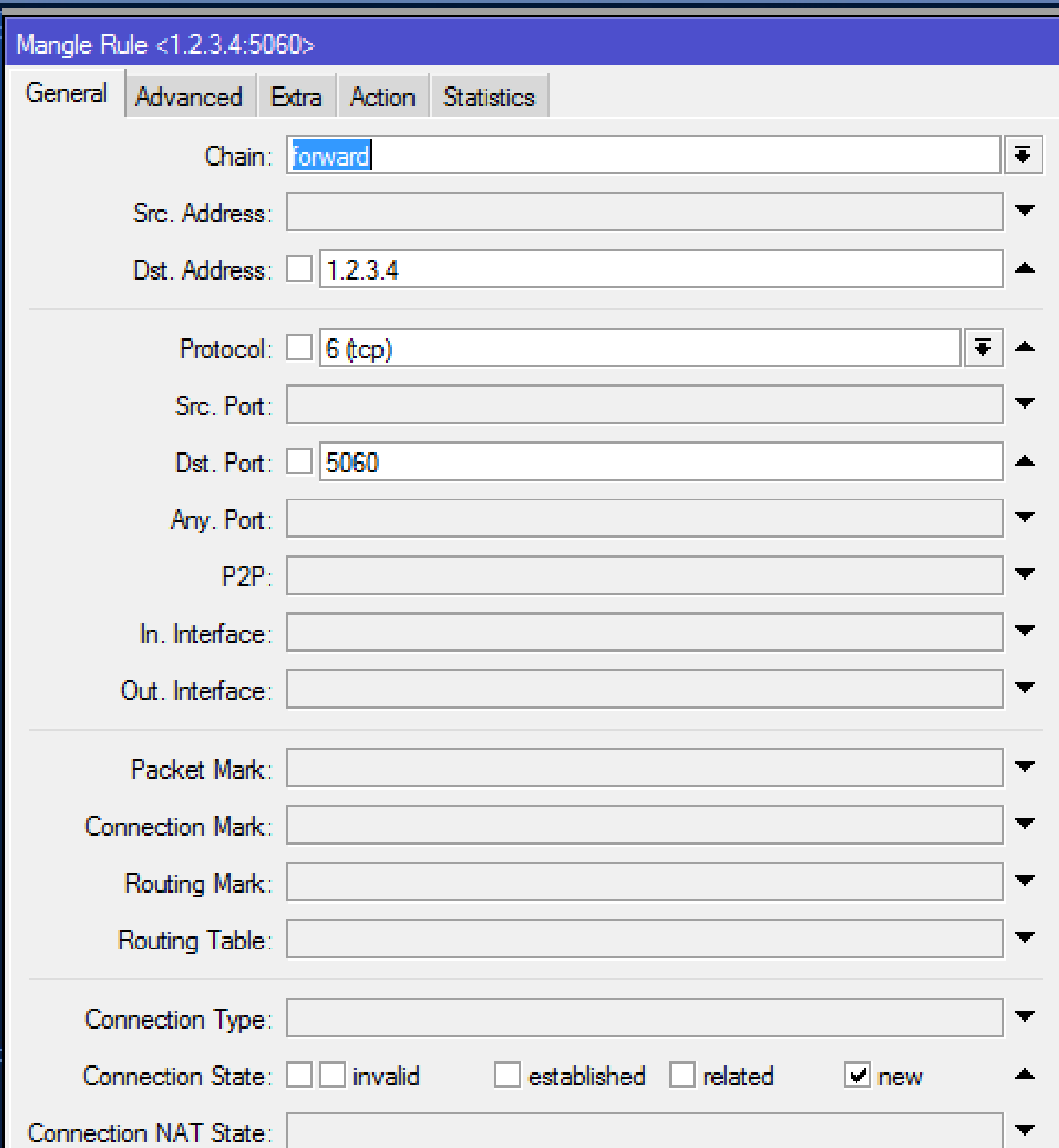
Once the session is in “connection tracker” all packets for that session are marked.

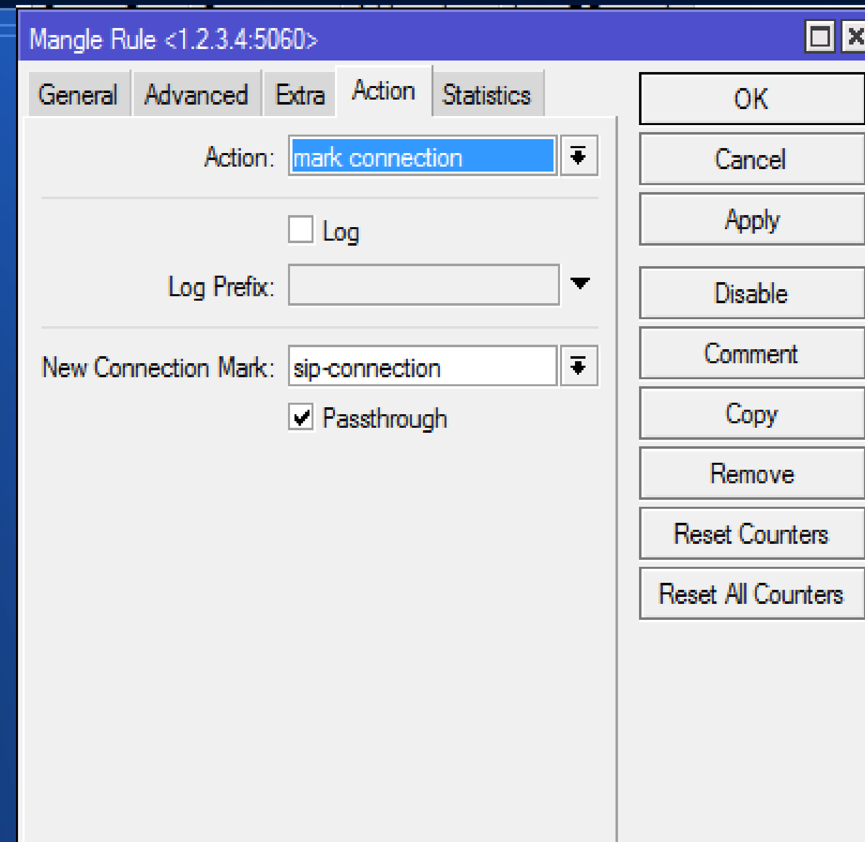
This is more efficient because Mangle doesn't need to scrutinize every packet. It just needs to know if the packet is in “that” connection.

Qualify Traffic

- SiP server = 1.2.3.4
- SiP port = 5060 tcp
- RTP port range = 10000 ~ 20000 udp

Mark the SiP connection



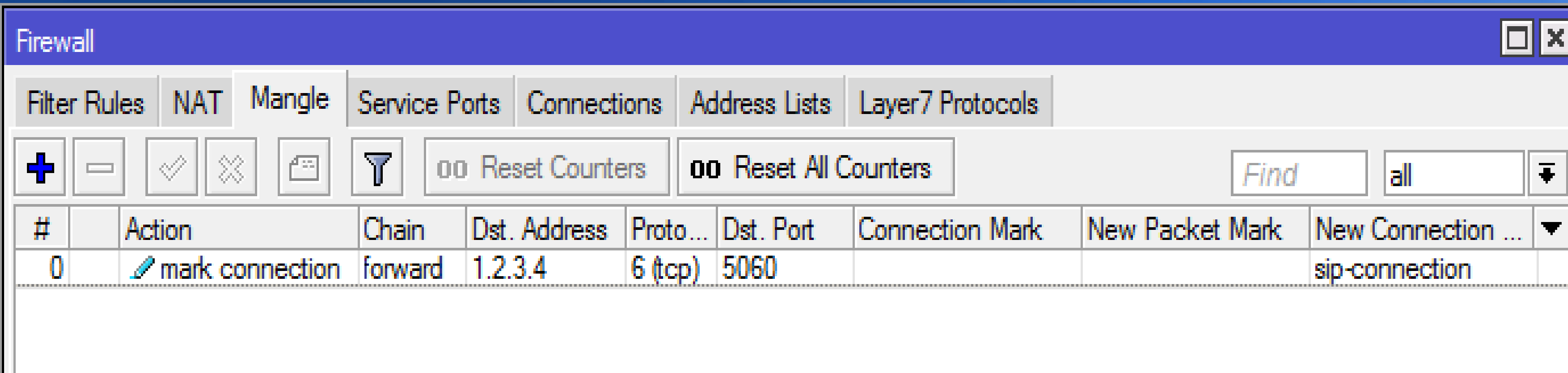


Mark the SiP connection

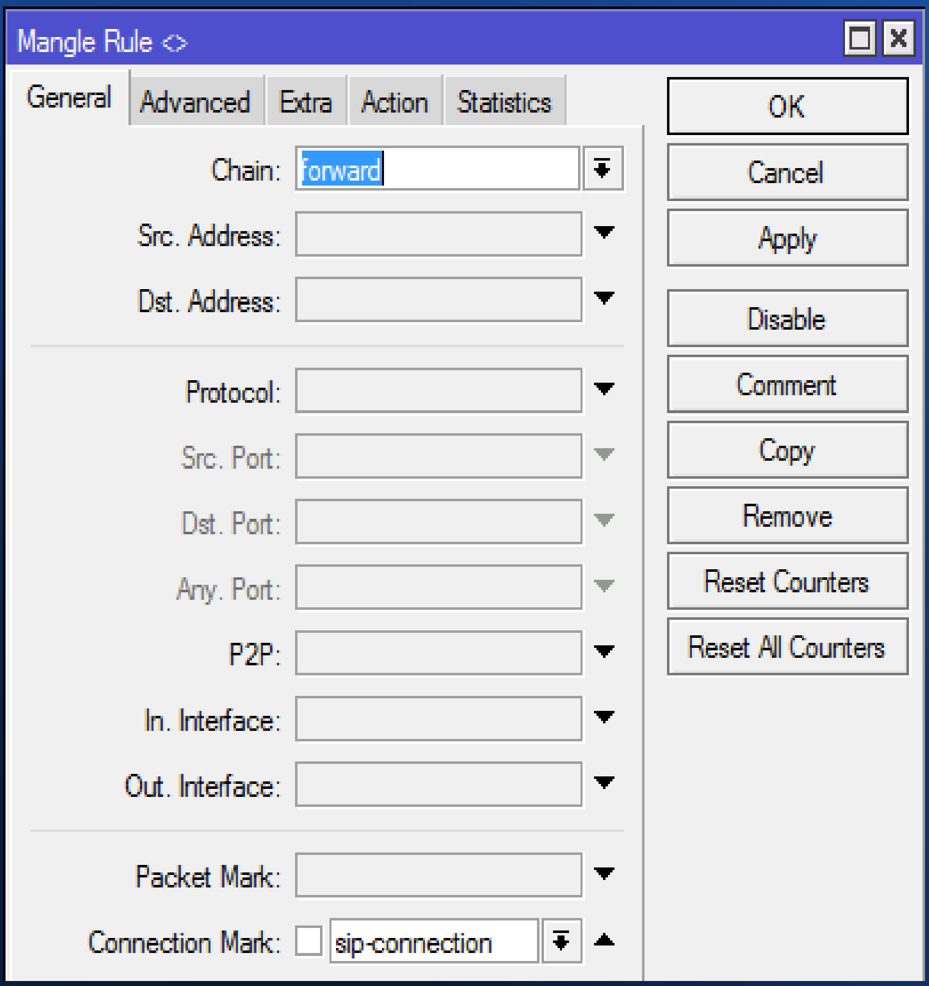
/ip firewall mangle

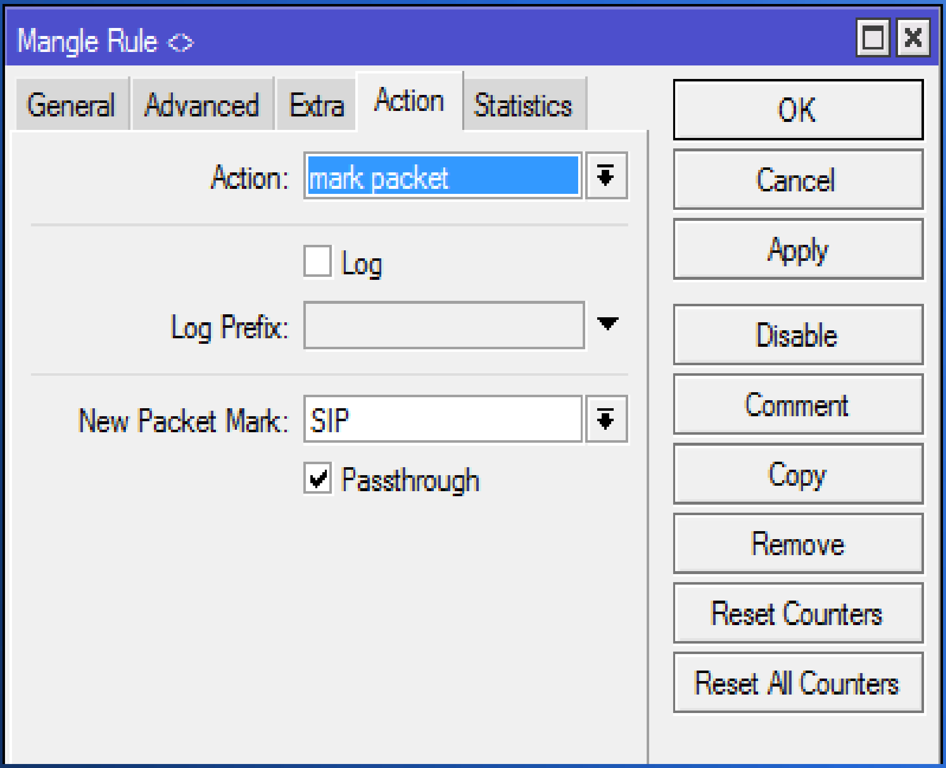
add chain=forward dst-address=1.2.3.4 protocol=tcp dst-port=5060 action=mark-connection new-connection-mark=sip-connection

Mark the SiP connection



Mark the SiP packets



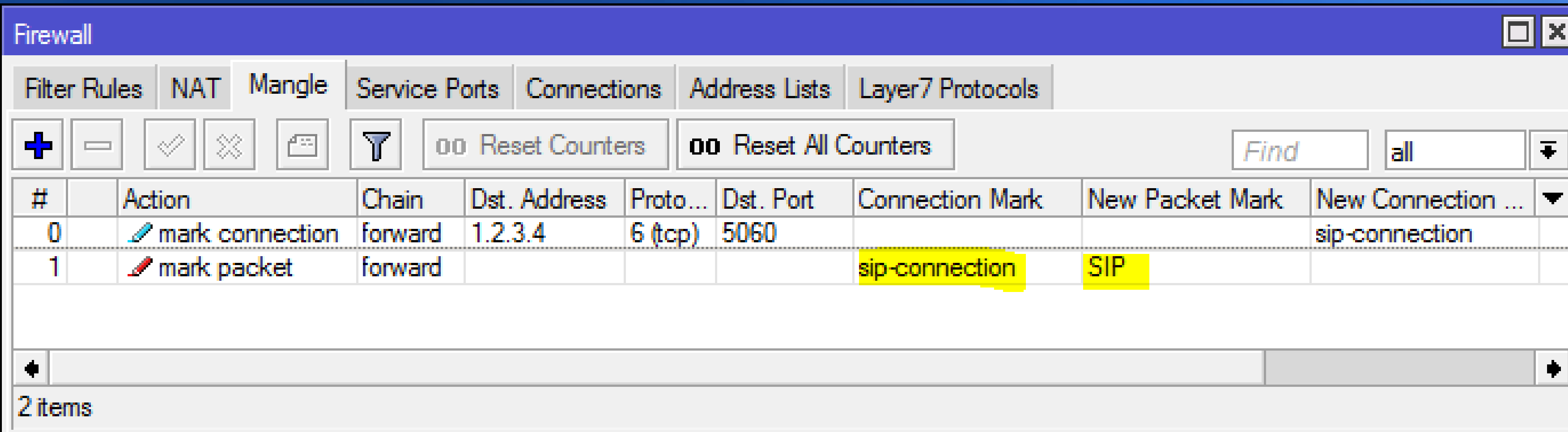


Mark the SiP packets

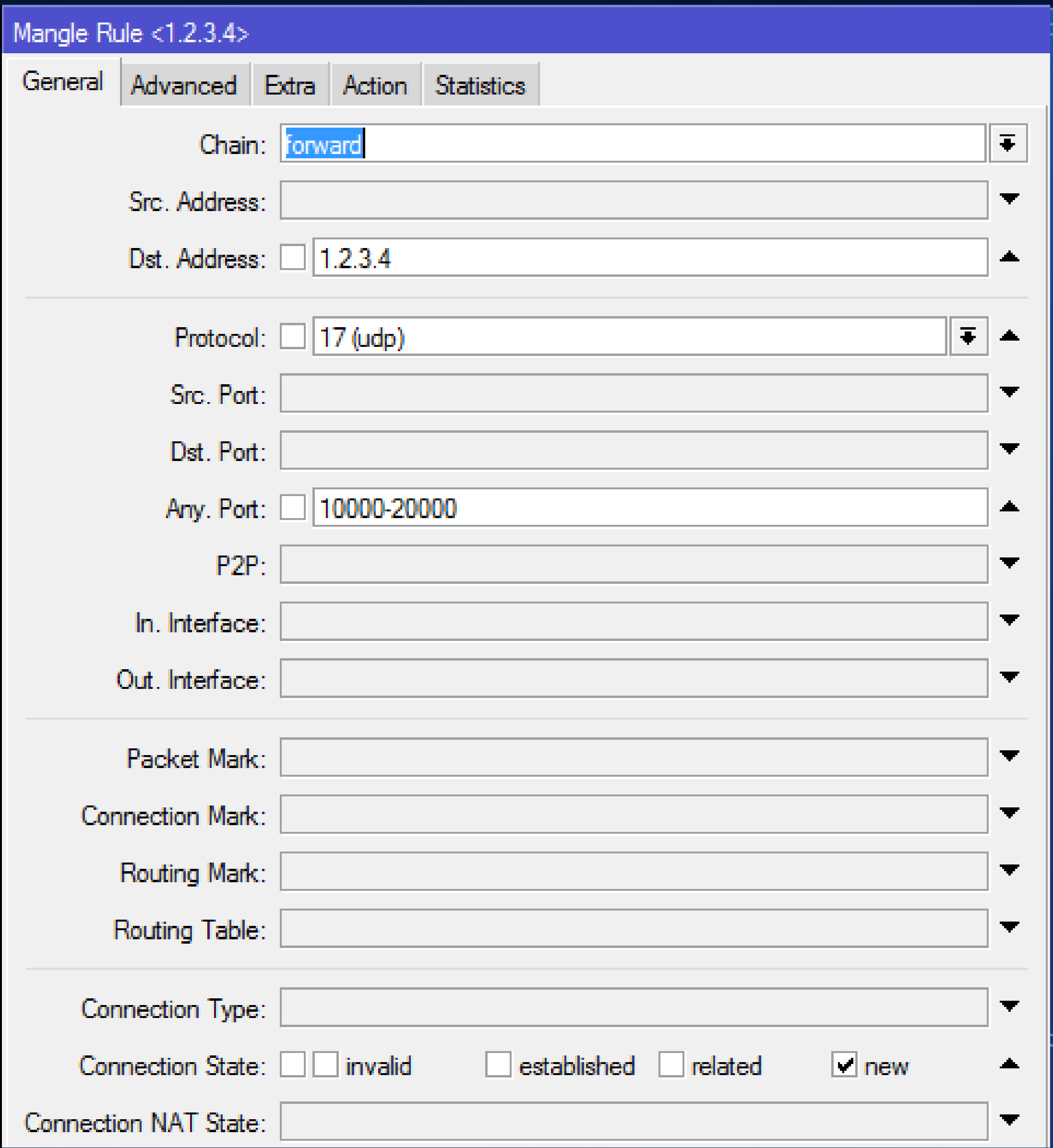
/ip firewall mangle

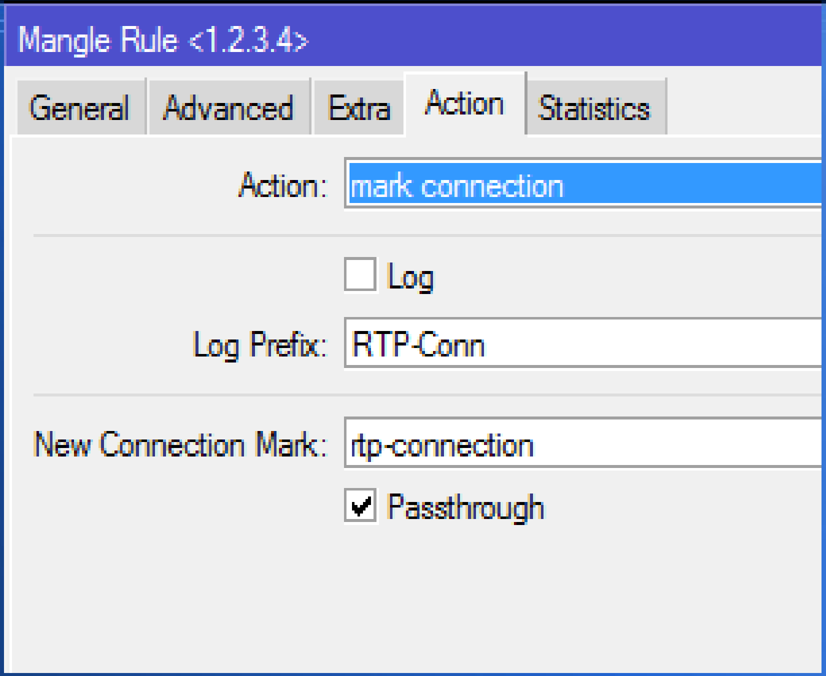
add chain=forward
connection-mark=sip-connection
add action=mark-packet
new-packet-mark=SIP

Mark the SiP packets



Mark the RTP connection



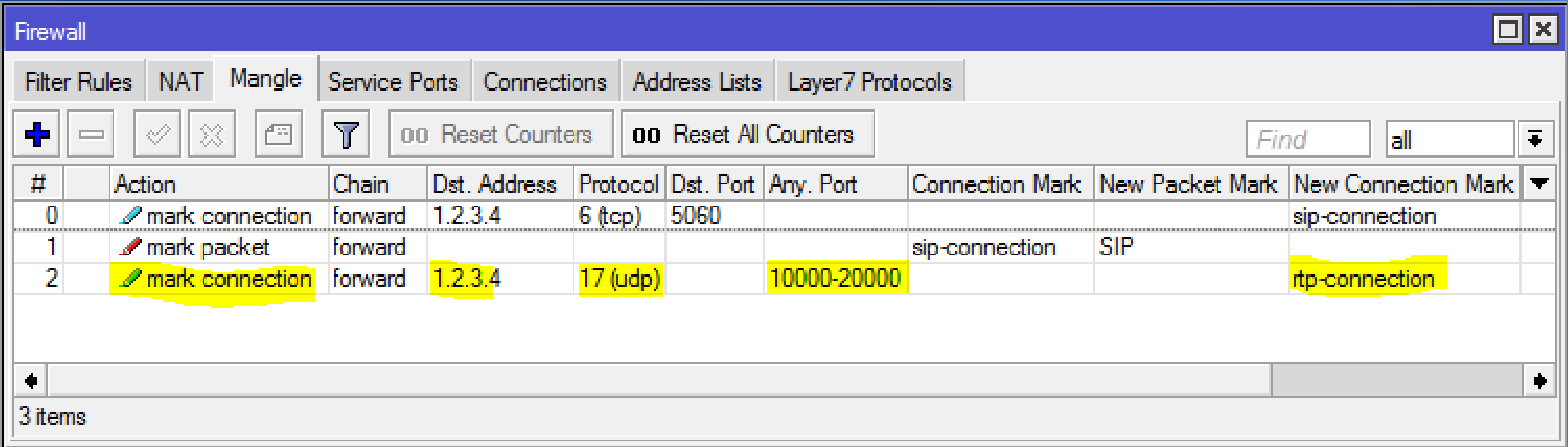


Mark the RTP connection

/ip firewall mangle

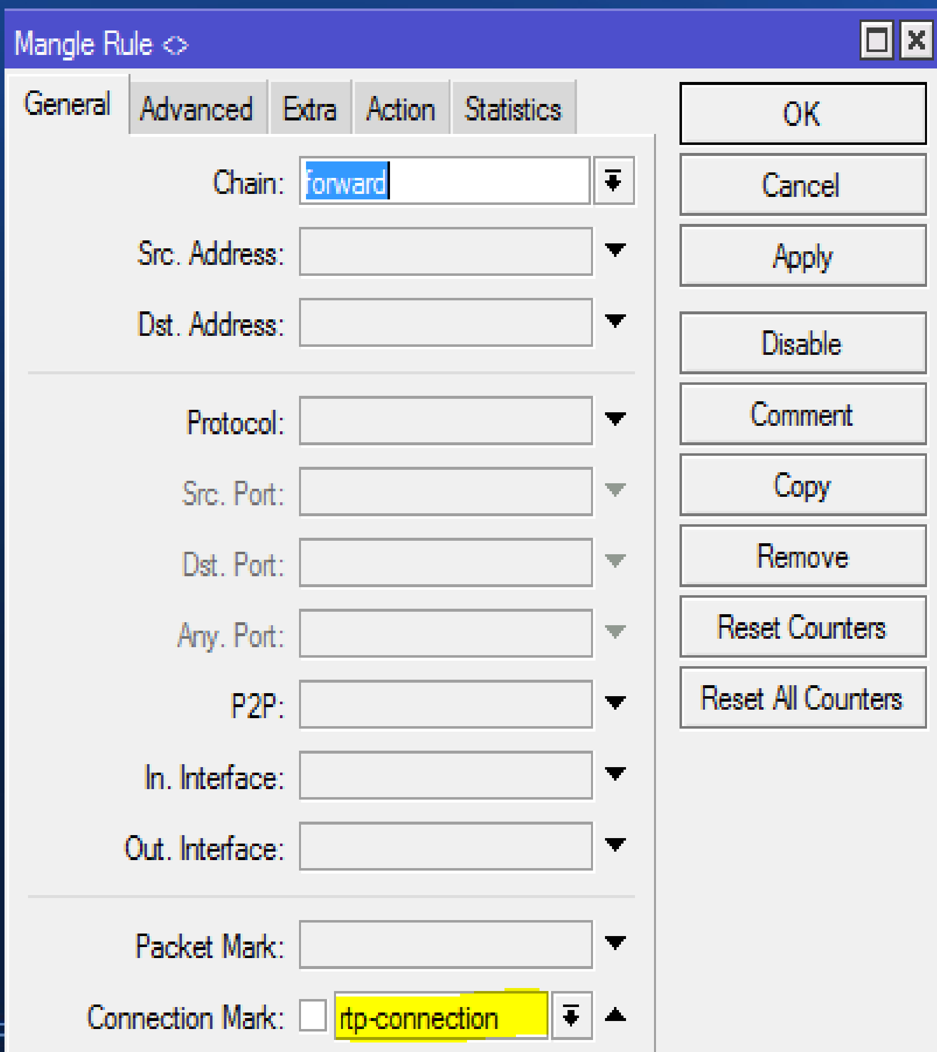
add action=mark-connection chain=forward dst-address=1.2.3.4 new-connection-mark=rtp-connection port=10000-20000 protocol=udp

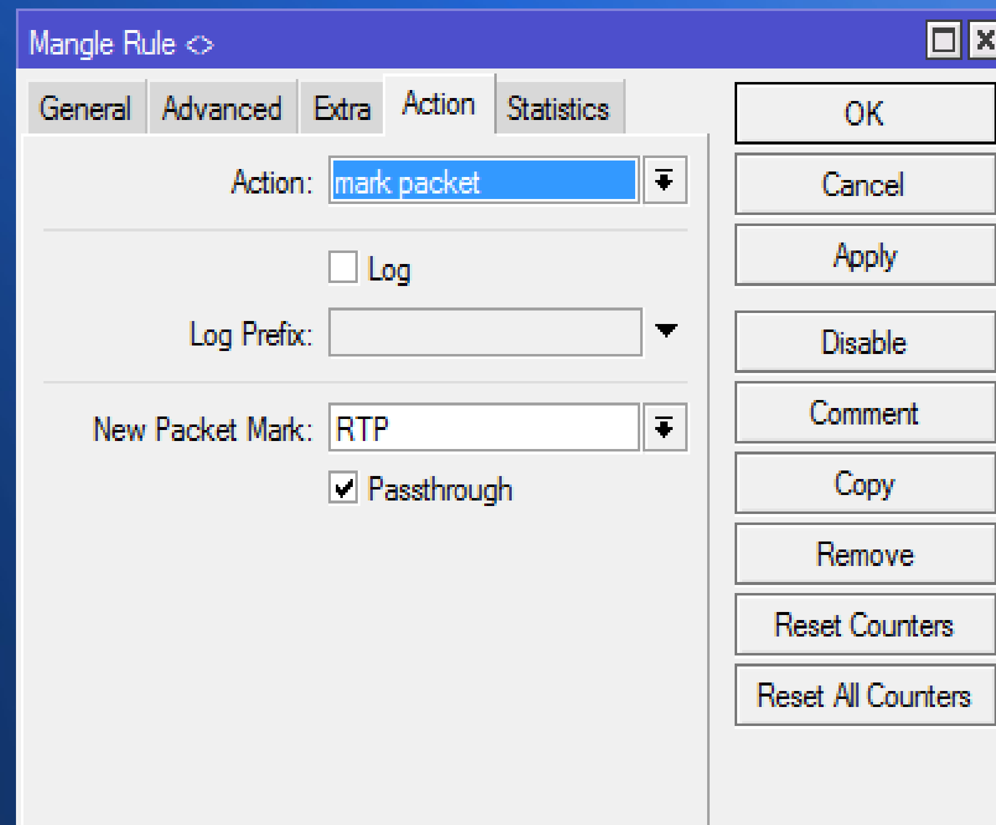
Mark the RTP connection



#		Action	Chain	Dst. Address	Protocol	Dst. Port	Any. Port	Connection Mark	New Packet Mark	New Connection Mark
0		mark connection	forward	1.2.3.4	6 (tcp)	5060				sip-connection
1		mark packet	forward					sip-connection	SIP	
2		mark connection	forward	1.2.3.4	17 (udp)		10000-20000			rtp-connection

Mark the RTP packets



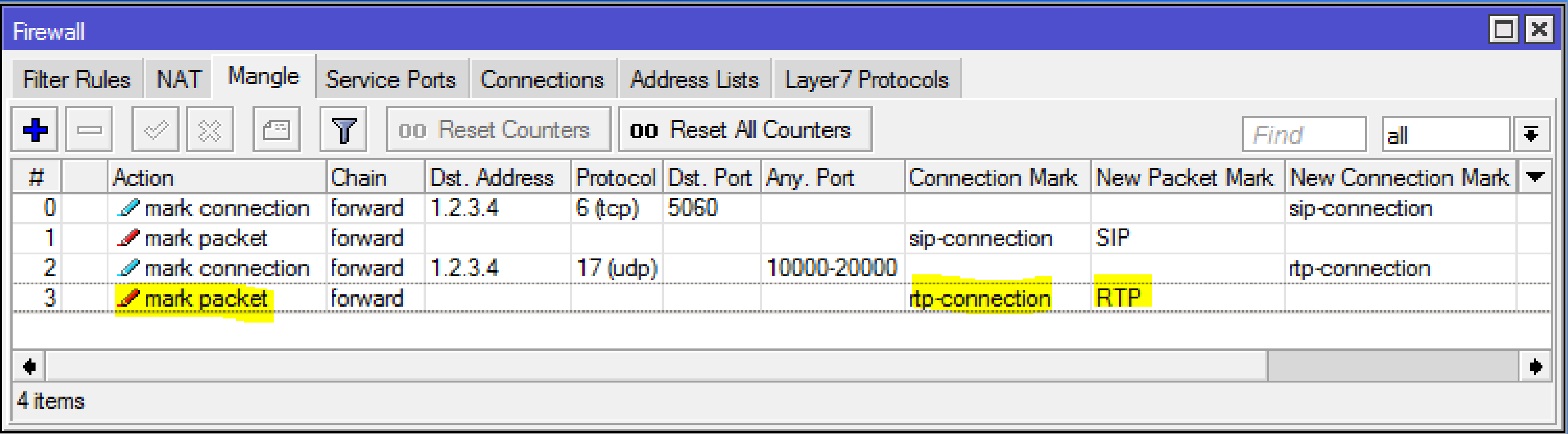


Mark the RTP connection

/ip firewall mangle

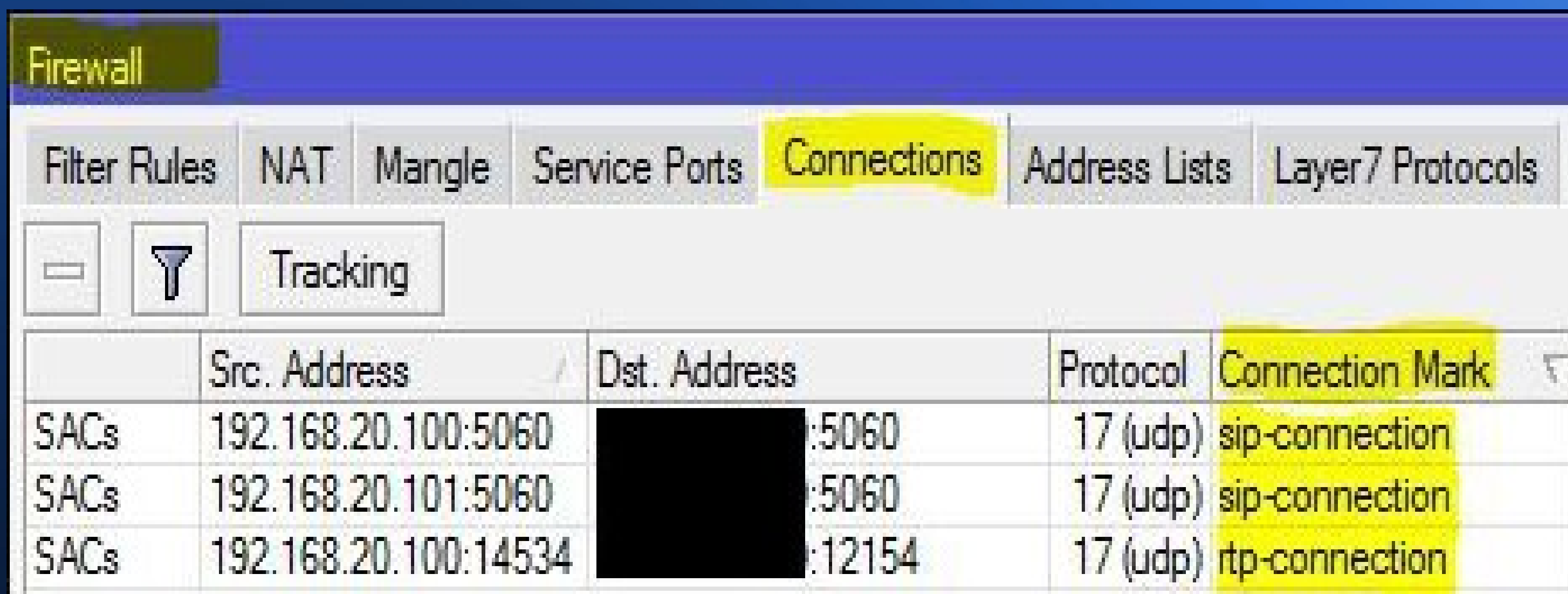
add action=mark-packet chain=forward connection-mark=rtp-connection new-packet-mark=RTP

Mark the RTP Packet



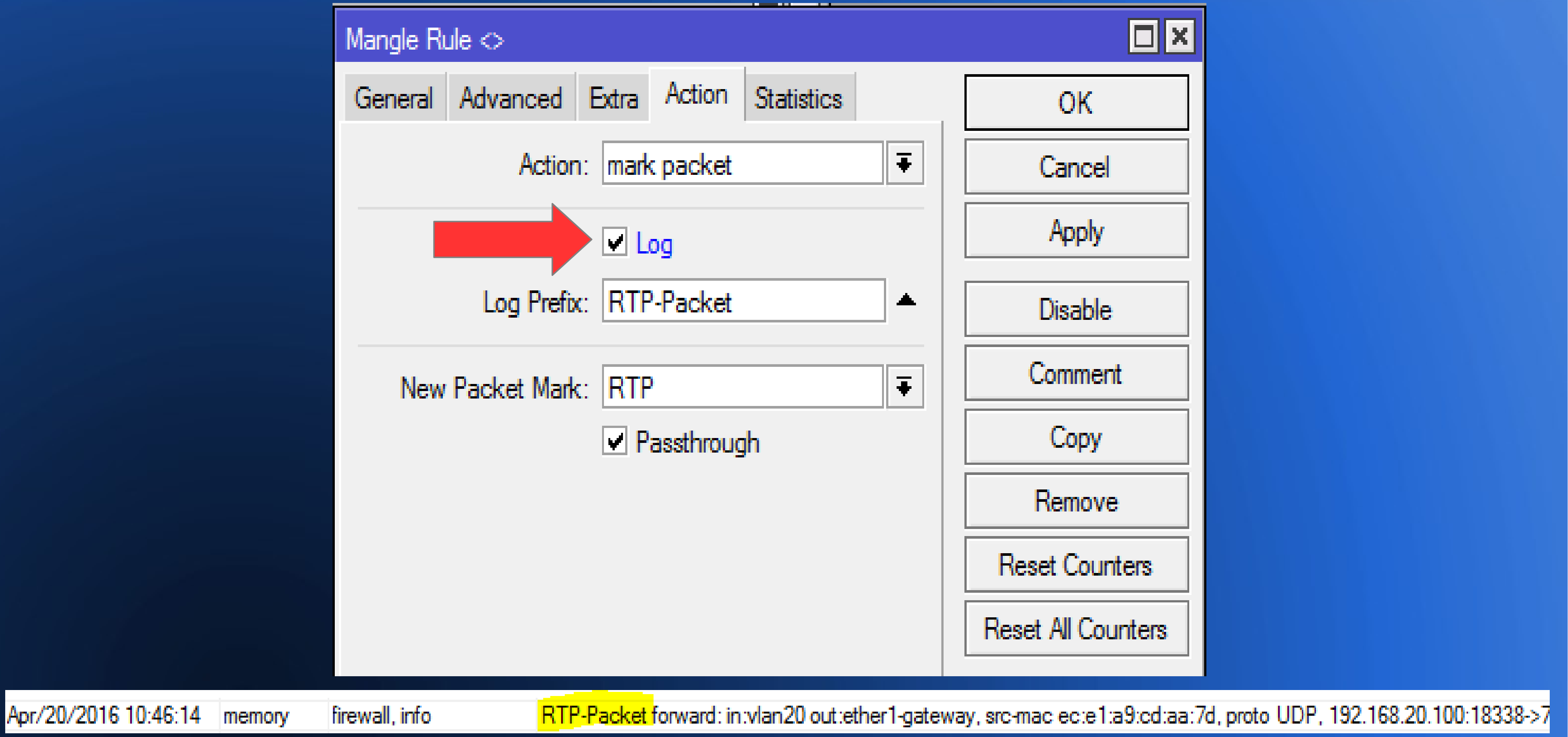
#		Action	Chain	Dst. Address	Protocol	Dst. Port	Any. Port	Connection Mark	New Packet Mark	New Connection Mark
0		mark connection	forward	1.2.3.4	6 (tcp)	5060				sip-connection
1		mark packet	forward					sip-connection	SIP	
2		mark connection	forward	1.2.3.4	17 (udp)		10000-20000			rtp-connection
3		mark packet	forward					rtp-connection	RTP	

How do we know its working?



	Src. Address	Dst. Address	Protocol	Connection Mark
SACs	192.168.20.100:5060	[redacted]:5060	17 (udp)	sip-connection
SACs	192.168.20.101:5060	[redacted]:5060	17 (udp)	sip-connection
SACs	192.168.20.100:14534	[redacted]:12154	17 (udp)	rtp-connection

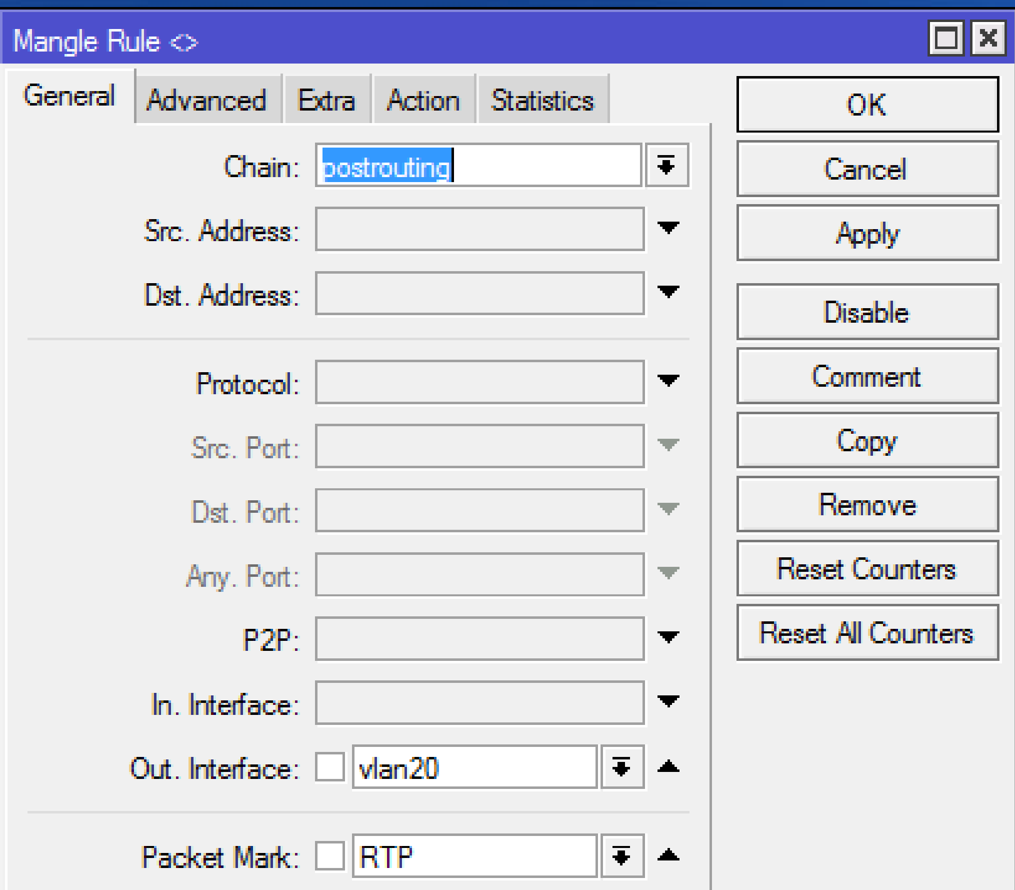
How do we know it's working?



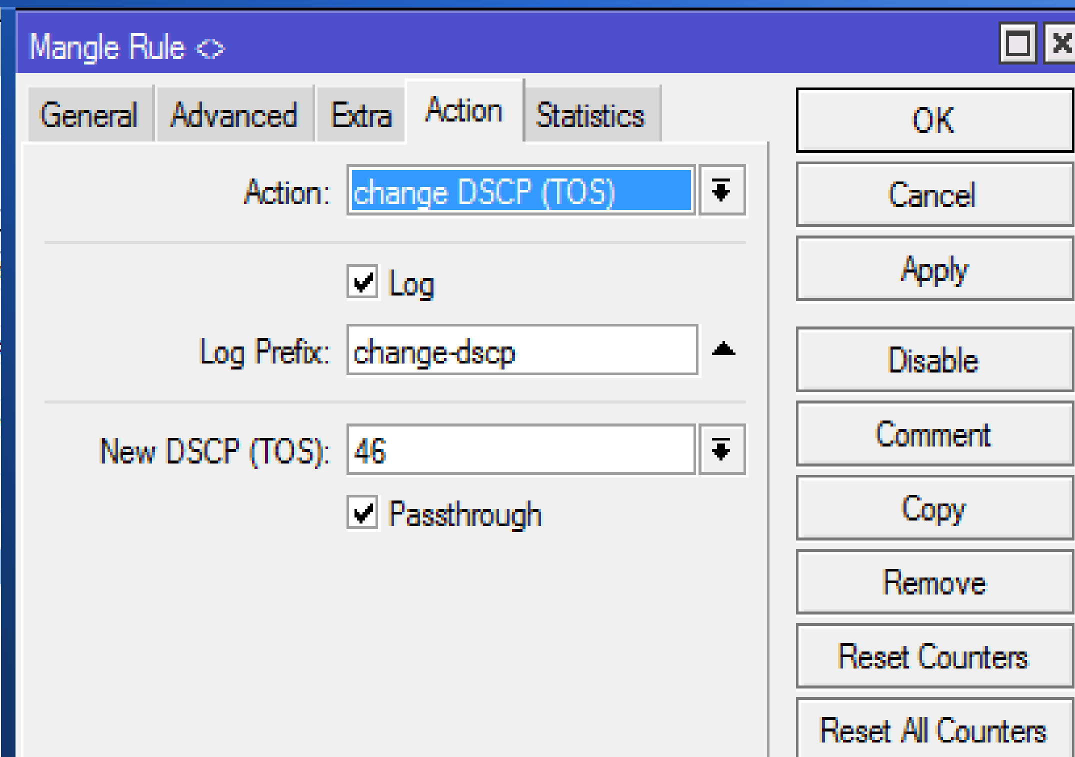
Change DSCP

Differentiated Services Code Point (DSCP) is a field in an IP packet that enables different levels of service to be assigned to network traffic.

Change DSCP



Mangle Rule <>
General
Advanced
Extra
Action
Statistics
Chain:
postrouting
Src. Address:
Dst. Address:
Protocol:
Src. Port:
Dst. Port:
Any. Port:
P2P:
In. Interface:
Out. Interface:
vlan20
Packet Mark:
RTP
OK
Cancel
Apply
Disable
Comment
Copy
Remove
Reset Counters
Reset All Counters



Mangle Rule <>
General
Advanced
Extra
Action
Statistics
Action:
change DSCP (TOS)
Log
Log Prefix:
change-dscp
New DSCP (TOS):
46
Passthrough
OK
Cancel
Apply
Disable
Comment
Copy
Remove
Reset Counters
Reset All Counters

Mikrotik Queues

Mikrotik Queues

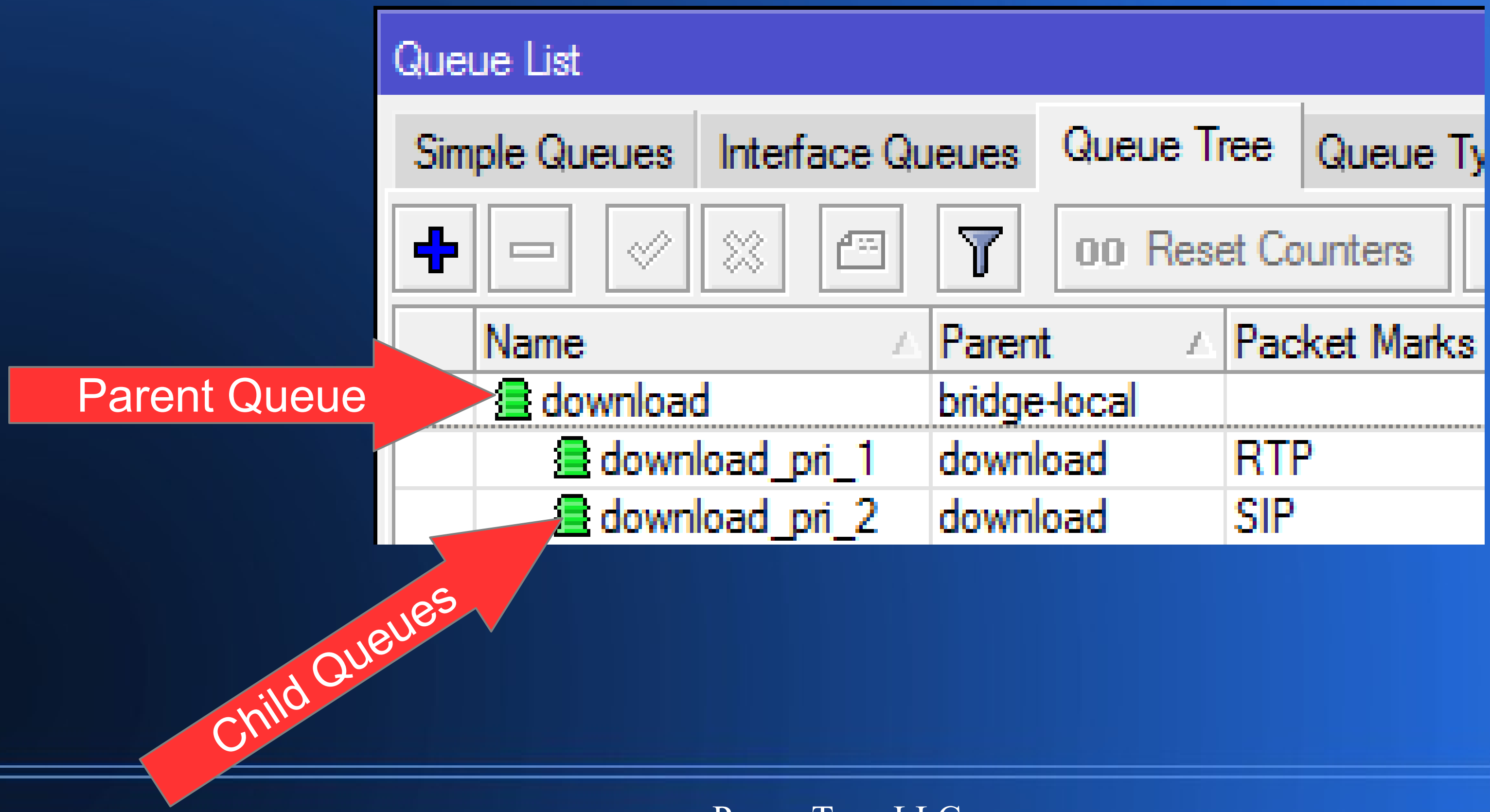
- limit data rate for certain IP addresses, subnets, protocols, ports, and other parameters
- limit peer-to-peer traffic
- prioritize some packet flows over others
- configure traffic bursts for faster web browsing
- apply different limits based on time
- share available traffic among users equally, or depending on the load of the channel

Mikrotik Queues

Parent Queues (inner queues) – distribute bandwidth

Child Queues (leaf queues) – consume bandwidth

Mikrotik Queues



Mikrotik Parent Queues

- Parent Queues only responsibility is to distribute traffic to child queues.
- Parent queue will first satisfy the child queue's “limit-at” traffic then try and reach child “max-limit”.
- Priorities are ignored on Parent Queues.

Mikrotik Queue priorities

- 8 is the lowest priority, 1 is the highest.
- Queue with higher priority will have a chance to satisfy its max-limit value before lower priority queues.
- Actual traffic prioritization will work **only** if limits are specified.

Create A Queue Tree

- Scenario:

My home office

5 phones

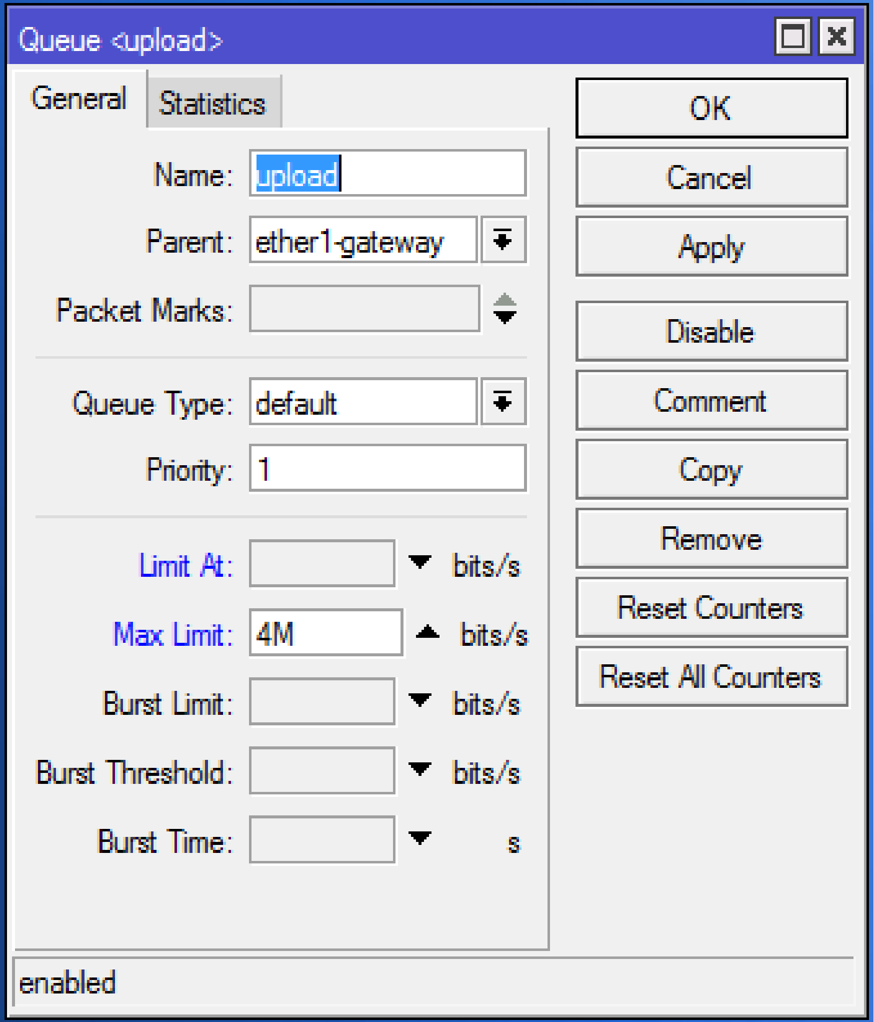
internet bandwidth = 35Mb download
4Mb upload

Create A Queue Tree

- First create a queue

  /queue tree

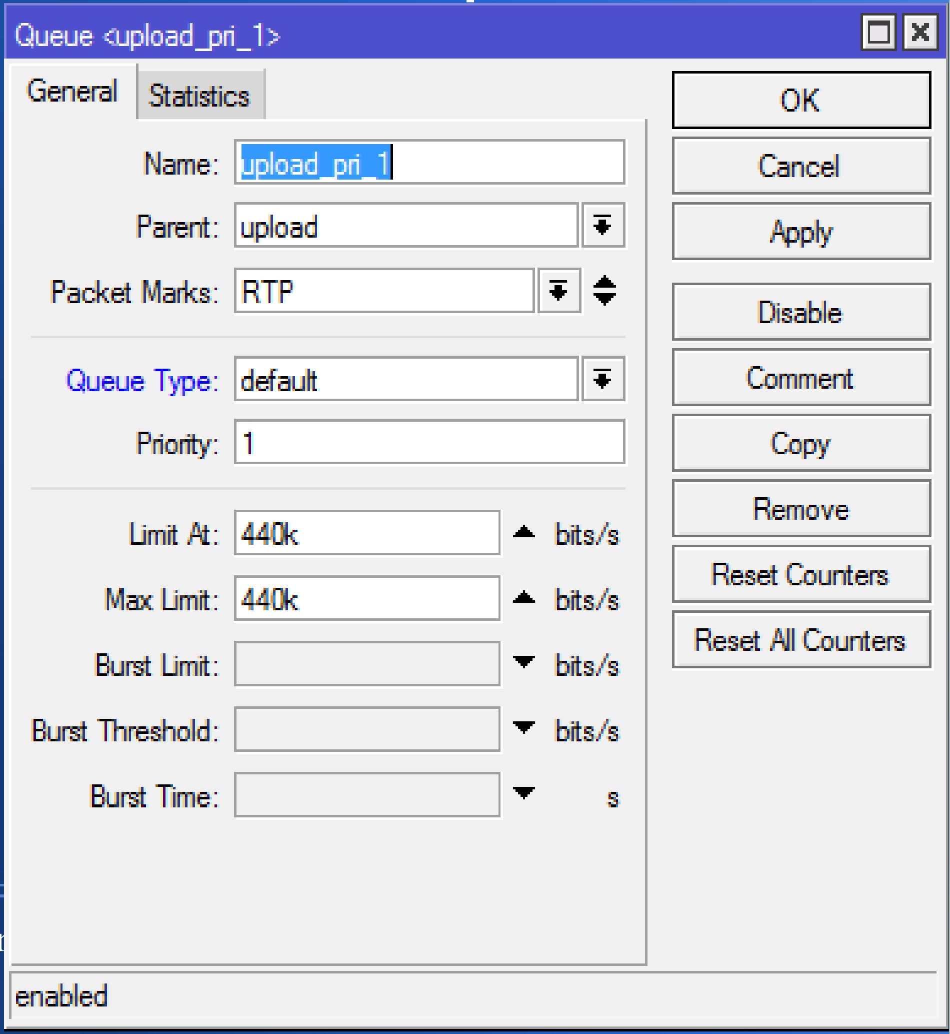
  add limit-at=4M max-limit=4M name=upload parent=ether1-gateway priority=8 queue=default



Create A Child Queue

- Create an RTP queue and select it's “parent” as “upload”

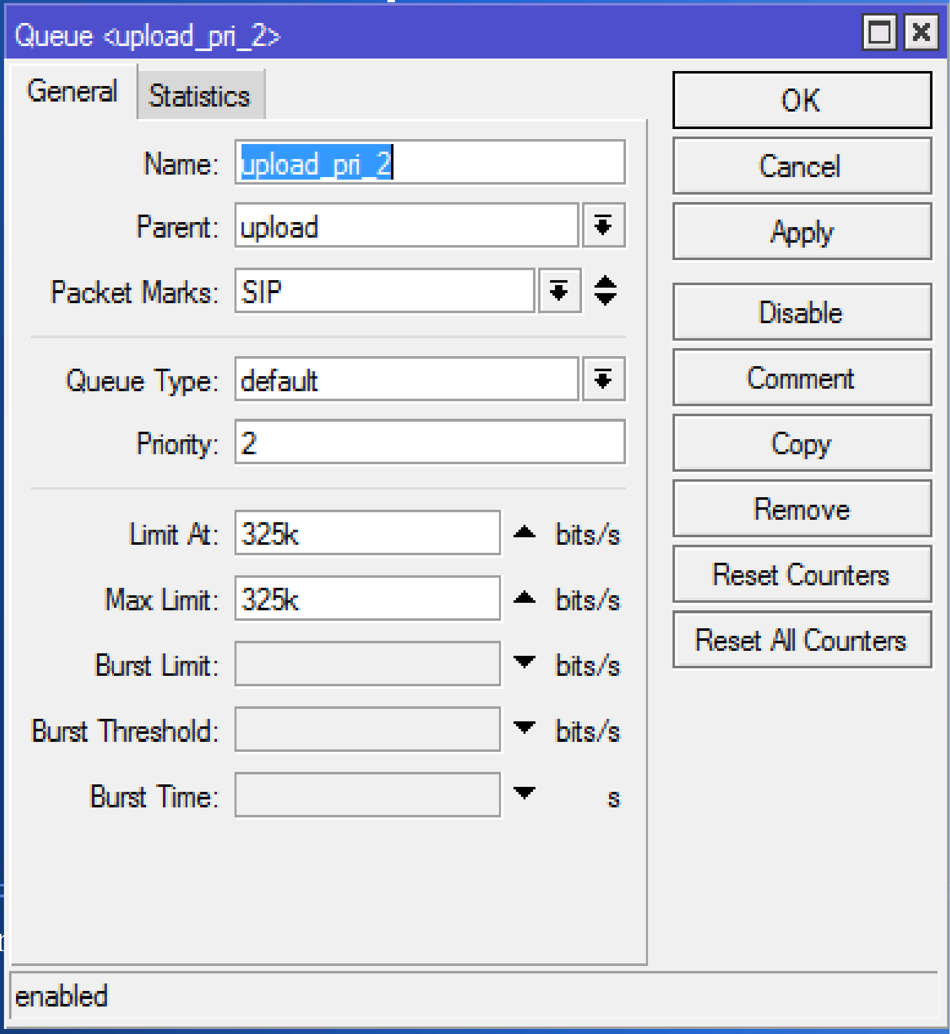
add limit-at=440k
max-limit=440k
name=upload_pri_1
packet-mark=RTP
parent=upload
priority=1
queue=default



Create A Child Queue

- Create a SIP queue and select it's “parent” as “upload”

add limit-at=325k
max-limit=325k
name=upload_pri_2
packet-mark=SIP
parent=upload
priority=2
queue=default



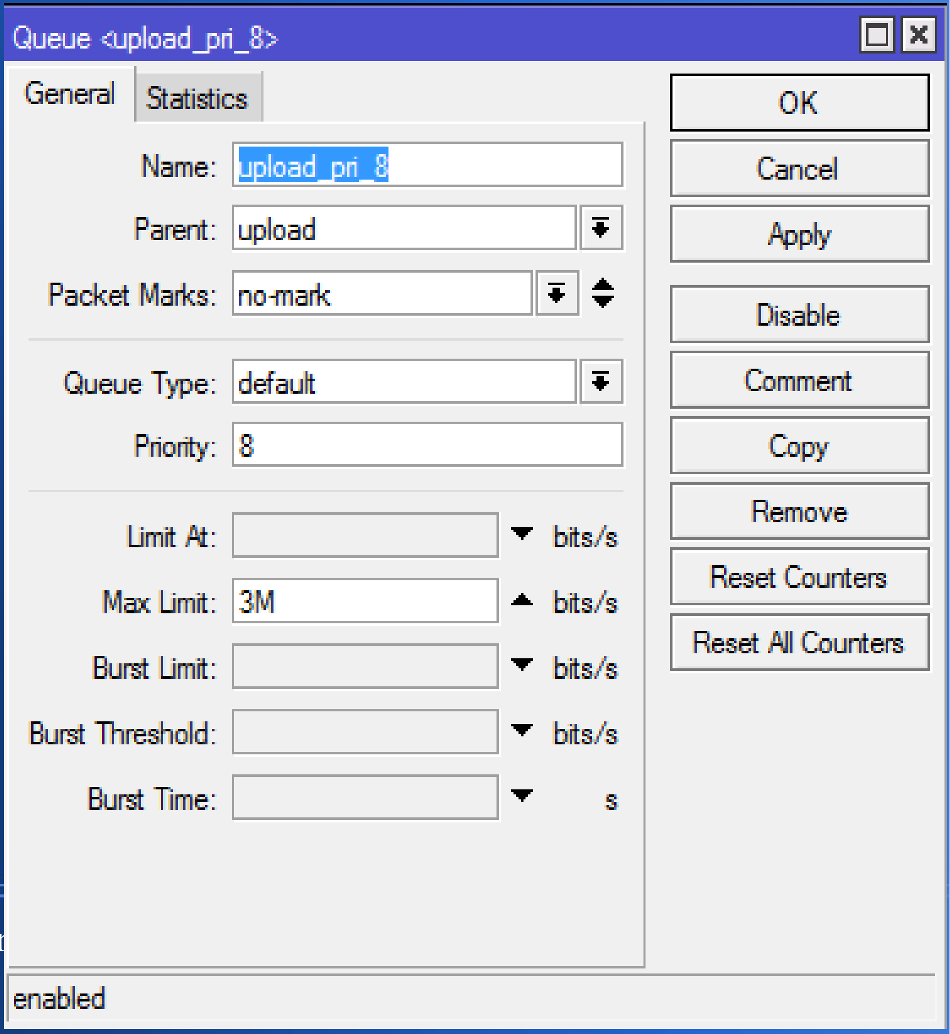
Create A Child Queue

What about packets without any marks?

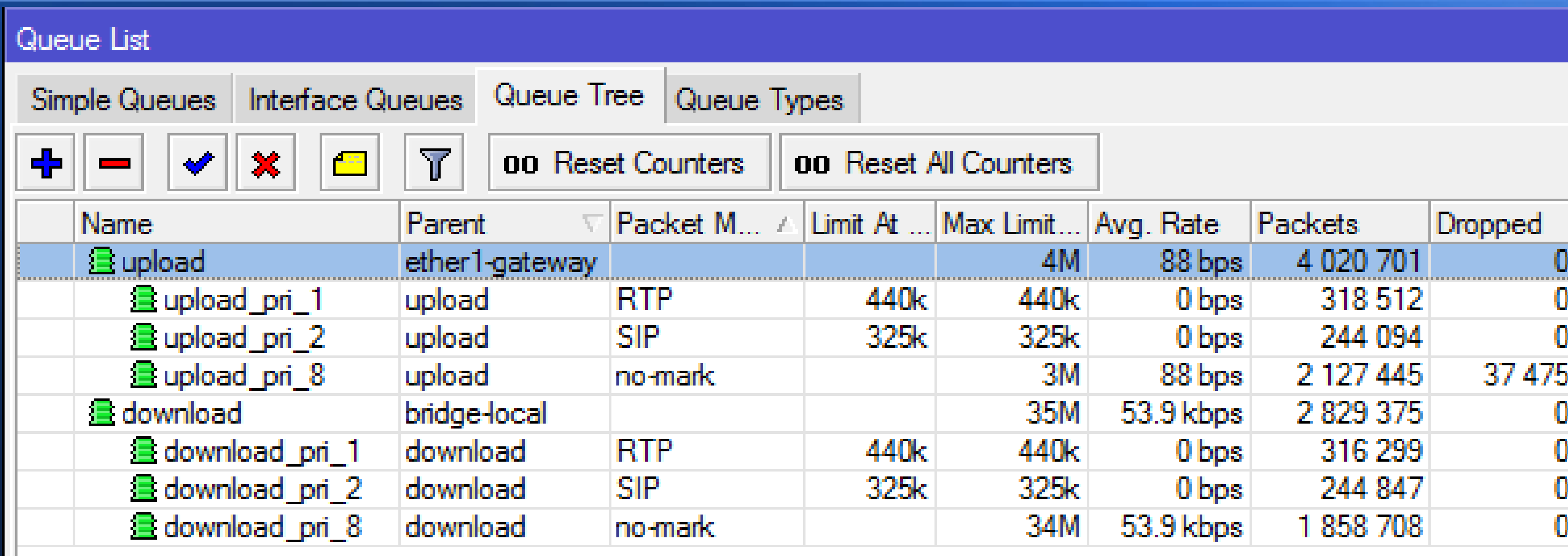
Create A Child Queue

- Create a “no mark” queue and select it's “parent” as “upload”

add limit-at=3M
max-limit=3M
name=upload_pri_2
packet-mark=SIP
parent=upload
priority=2
queue=default



Queue Tree GUI view



Name	Parent	Packet M...	Limit At ...	Max Limit...	Avg. Rate	Packets	Dropped
upload	ether1-gateway			4M	88 bps	4 020 701	0
upload_pri_1	upload	RTP	440k	440k	0 bps	318 512	0
upload_pri_2	upload	SIP	325k	325k	0 bps	244 094	0
upload_pri_8	upload	no-mark		3M	88 bps	2 127 445	37 475
download	bridge-local			35M	53.9 kbps	2 829 375	0
download_pri_1	download	RTP	440k	440k	0 bps	316 299	0
download_pri_2	download	SIP	325k	325k	0 bps	244 847	0
download_pri_8	download	no-mark		34M	53.9 kbps	1 858 708	0

Queue Tree GUI view

Queue List

Simple Queues | Interface Queues | Queue Tree | Queue Types

00 Reset Counters | 00 Reset All Counters

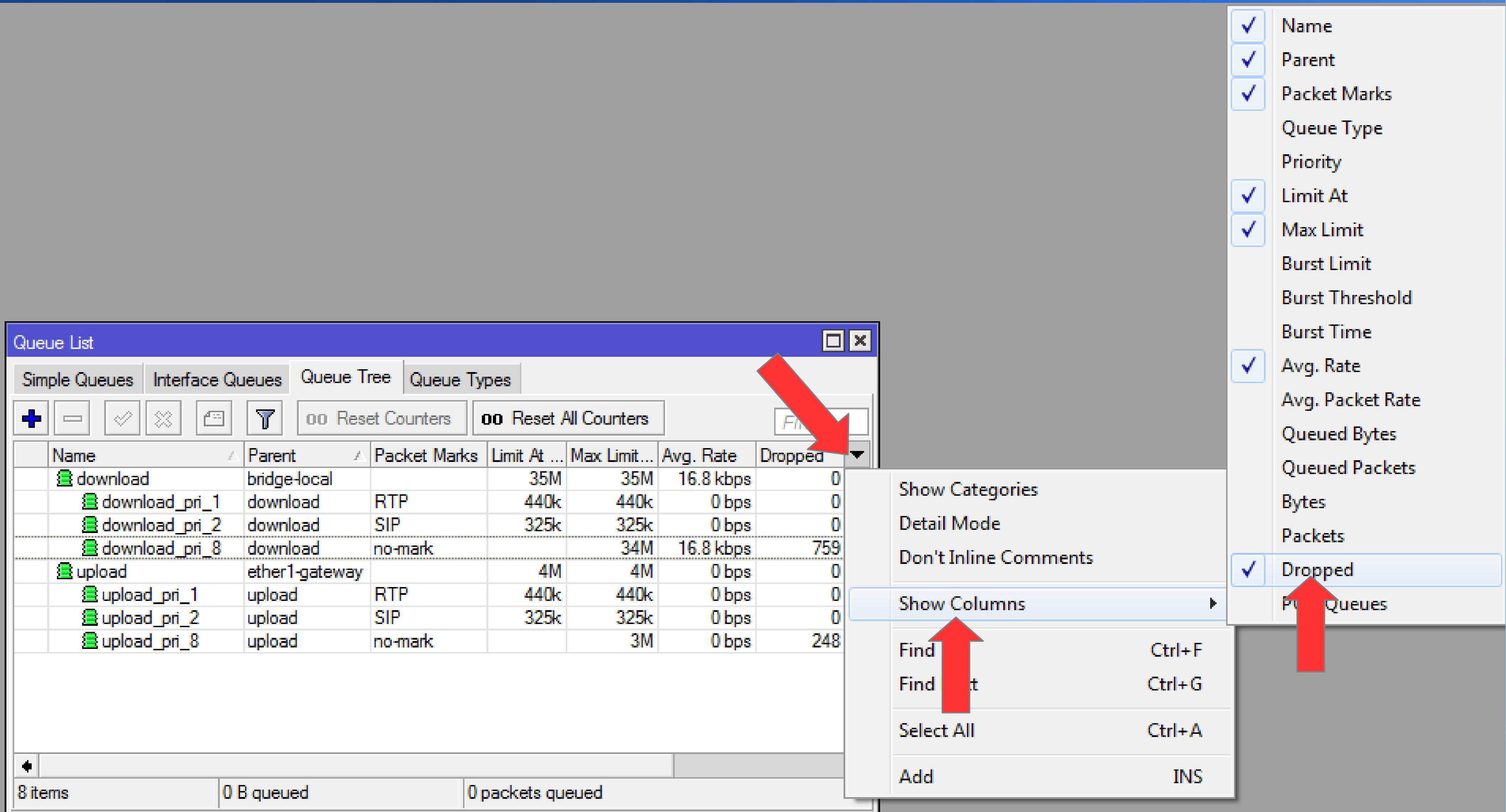
	Name	Parent	Packet M...	Limit At ...	Max Limit...	Avg. Rate	Packets	Dropped
	download	bridge-local			35M	59.5 kbps	2 831 854	0
	download_pri_1	download	RTP	440k	440k	0 bps	316 299	0
	download_pri_2	download	SIP	325k	325k	6.7 kbps	245 095	0
	download_pri_3	download	TCP-ACK	1M	1M	128 bps	201	0
	download_pri_4	download	DNS	1k	128k	0 bps	0	0
	download_pri_8	download	no-mark		34M	52.6 kbps	1 860 738	0
	upload	ether1-gateway			4M	8.5 kbps	4 021 472	0
	upload_pri_1	upload	RTP	440k	440k	0 bps	318 512	0
	upload_pri_2	upload	SIP	325k	325k	7.8 kbps	244 341	0
	upload_pri_3	upload	TCP-ACK	1M	1500k	480 bps	299	0
	upload_pri_4	upload	DNS	1k	128k	0 bps	0	0
	upload_pri_8	upload	no-mark		3M	192 bps	2 127 670	37 475

Detecting Problems

Detecting problems

- Check for “dropped packets” in queue tree
- Enable the “dropped packets” view

Detecting problems



Detecting problems

- Check for “dropped packets” in queue tree
- Enable the “dropped packets” view

Queue List

Simple Queues | Interface Queues | Queue Tree | Queue Types

00 Reset Counters | 00 Reset All Counters

Name	Parent	Packet Marks	Limit At ...	Max Limit...	Avg. Rate	Dropped
download	bridge-local		35M	35M	246.5 kbps	0
download_pri_1	download	RTP	440k	440k	0 bps	0
download_pri_2	download	SIP	325k	325k	0 bps	0
download_pri_8	download	no-mark		34M	246.5 kbps	759
upload	ether1-gateway		4M	4M	3.0 Mbps	0
upload_pri_1	upload	RTP	440k	440k	78.2 kbps	0
upload_pri_2	upload	SIP	325k	325k	0 bps	0
upload_pri_8	upload	no-mark		3M	3.0 Mbps	223

Detecting problems

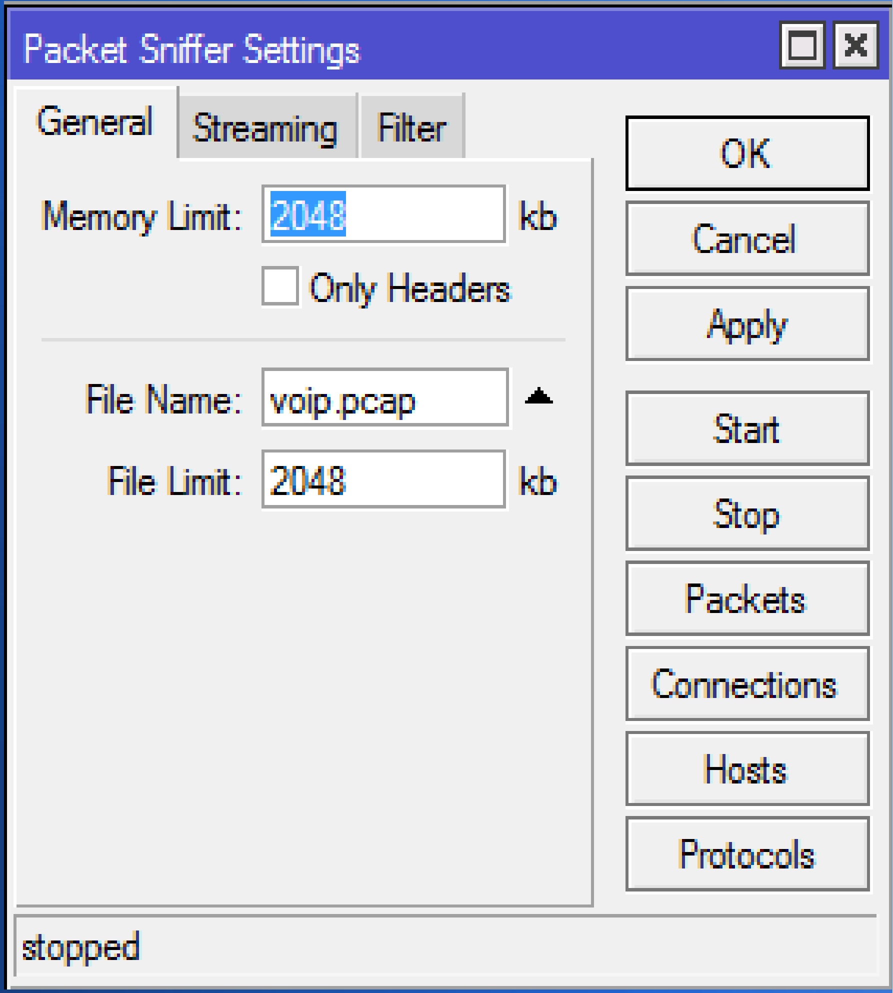
Mikrotik packet capture

Detecting problems

Mikrotik packet capture

From GUI:

Tools – Packet sniffer



Detecting problems

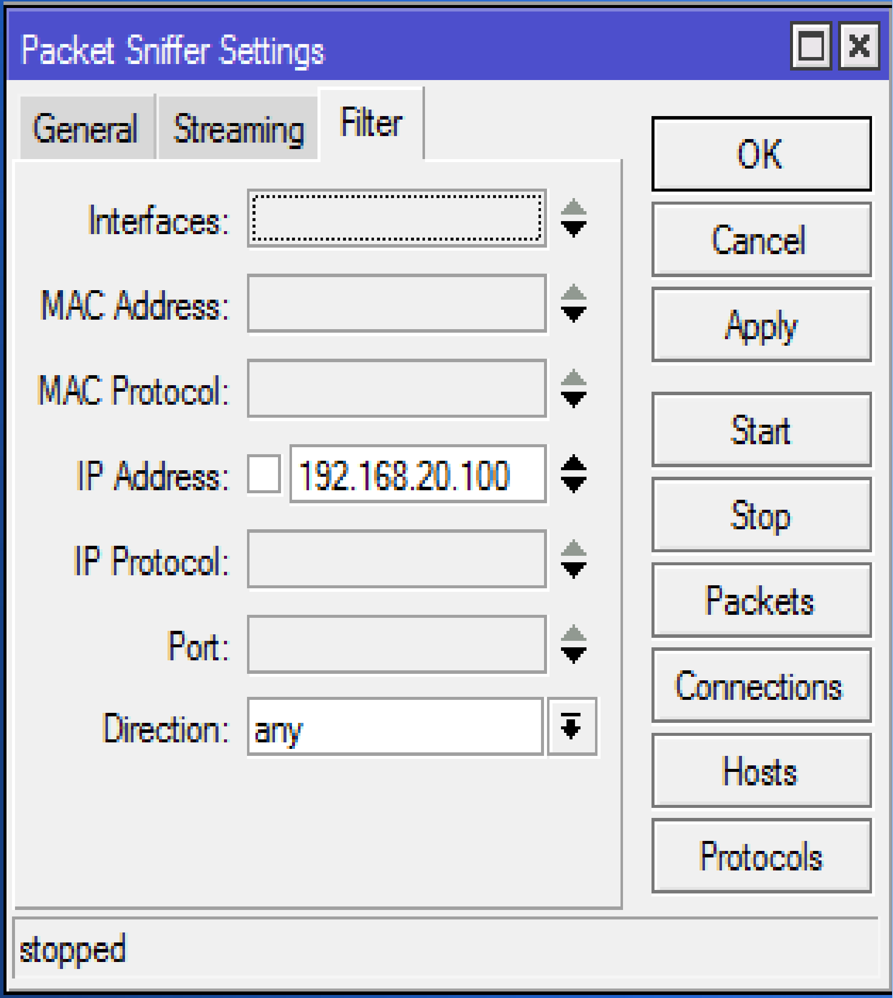
Mikrotik packet capture

From GUI:

Tools – Packet sniffer

Filter

IP



Detecting problems

Cap file voip.pcap will be created in “Files”

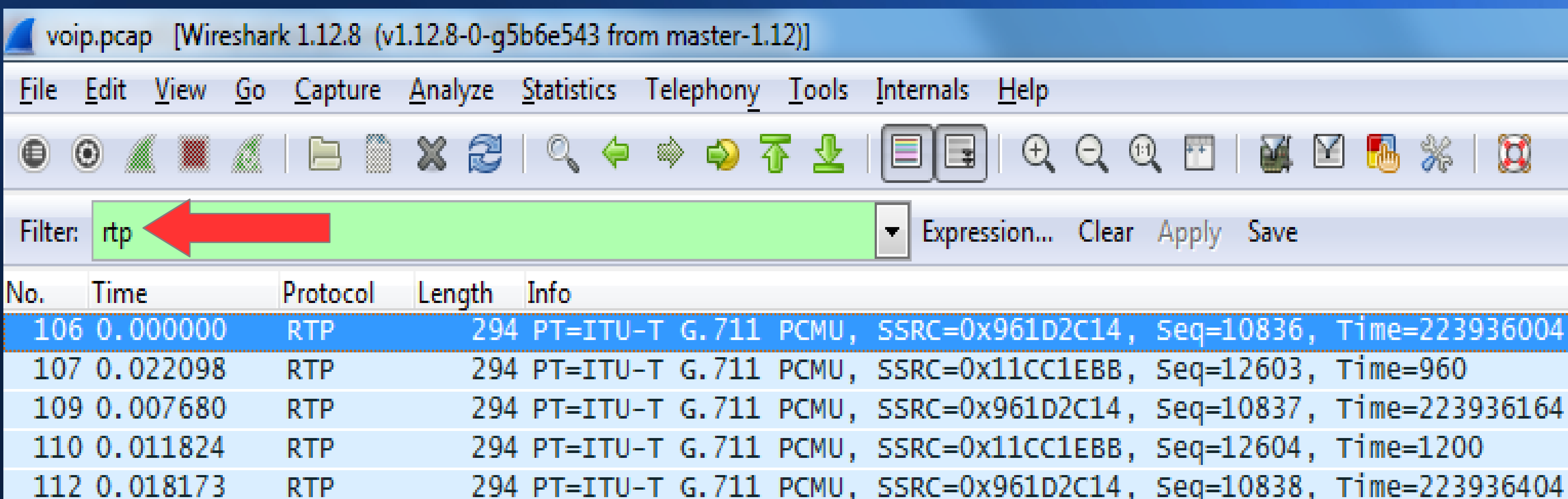
Download it

Then open voip.pcap in wireshark

Detecting problems

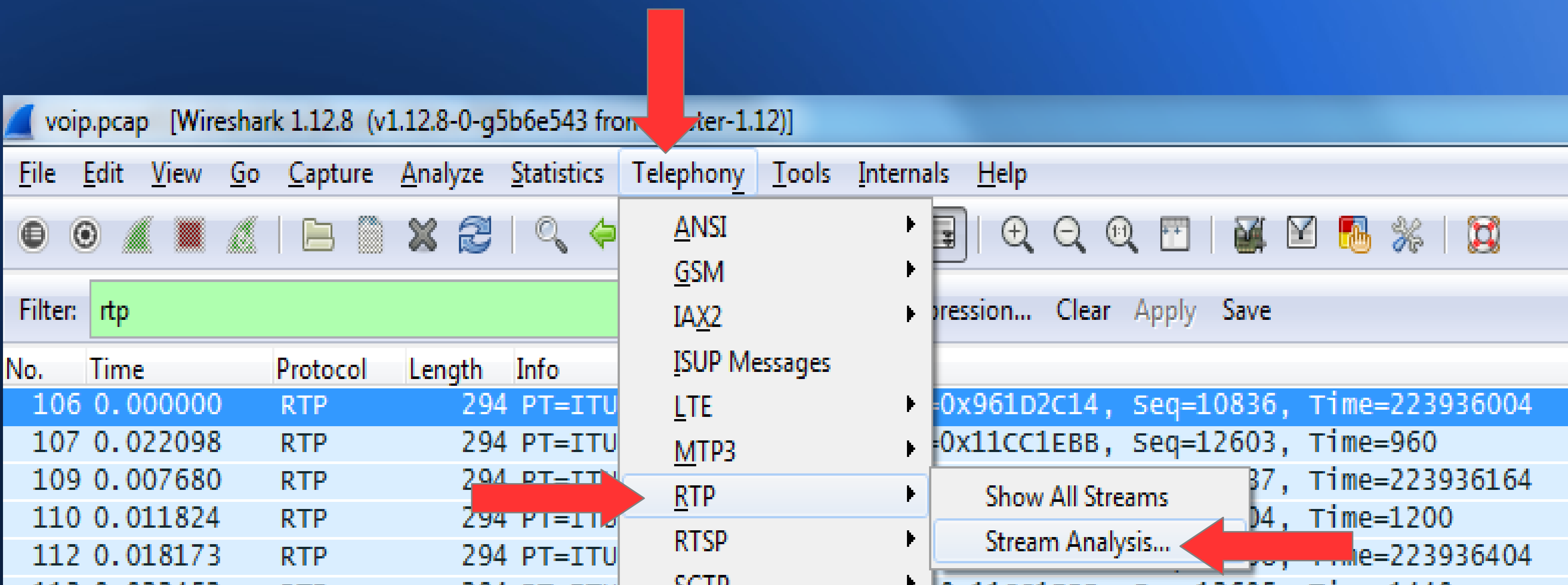
Filer = rtp

Then select one RTP packet

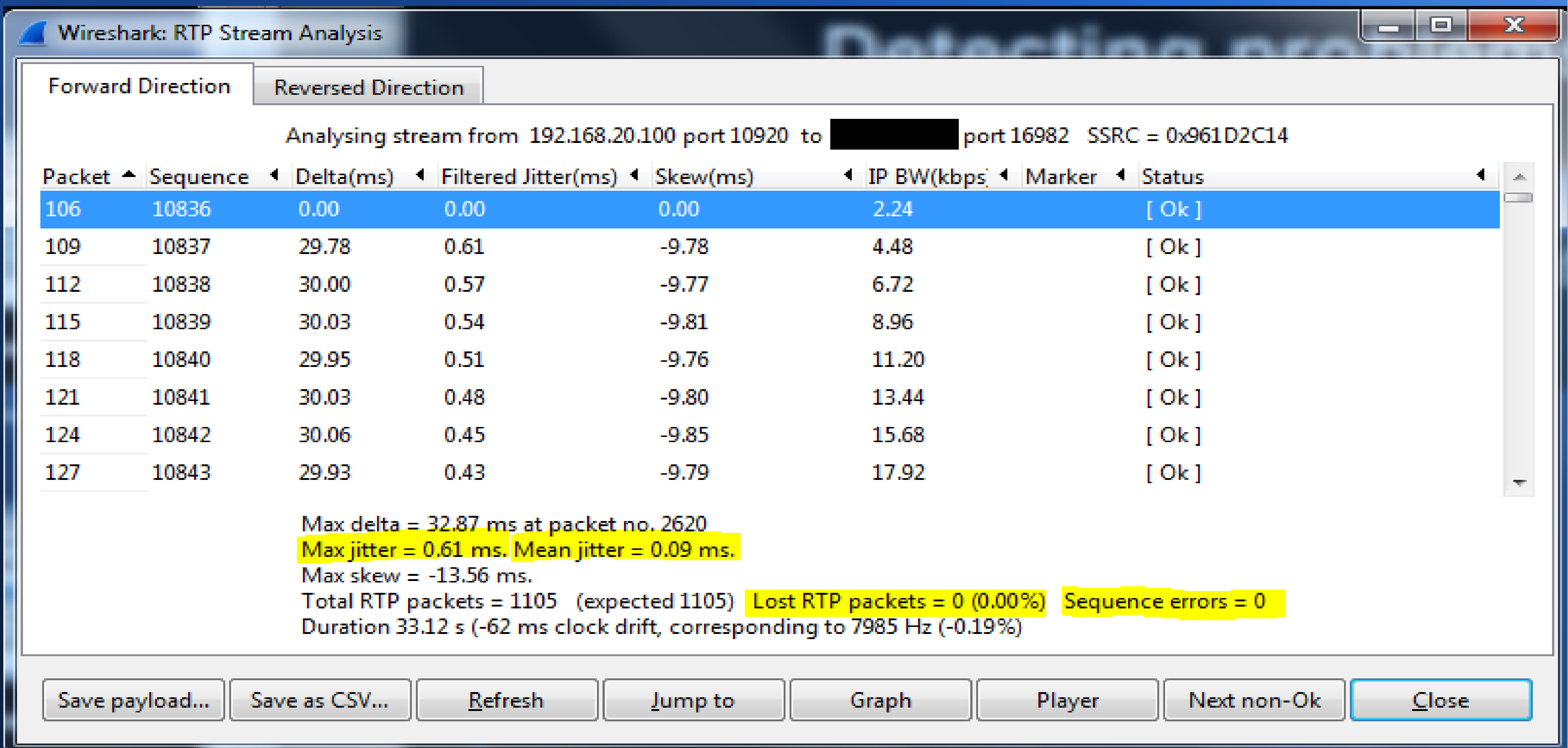


Detecting problems

Select “Telephony” - “RTP” - “Stream Analysis”



Detecting problems



Wireshark: RTP Stream Analysis

Forward Direction | Reversed Direction

Analysing stream from 192.168.20.100 port 10920 to [redacted] port 16982 SSRC = 0x961D2C14

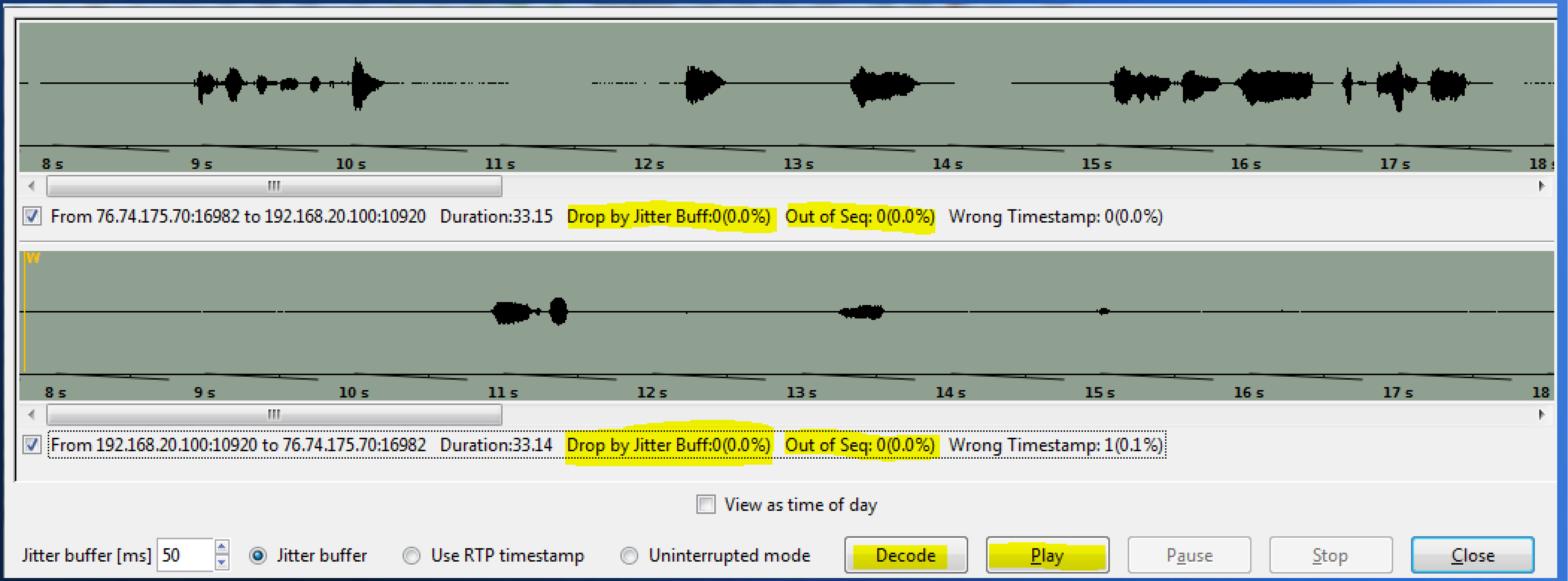
Packet	Sequence	Delta(ms)	Filtered Jitter(ms)	Skew(ms)	IP BW(kbps)	Marker	Status
106	10836	0.00	0.00	0.00	2.24		[Ok]
109	10837	29.78	0.61	-9.78	4.48		[Ok]
112	10838	30.00	0.57	-9.77	6.72		[Ok]
115	10839	30.03	0.54	-9.81	8.96		[Ok]
118	10840	29.95	0.51	-9.76	11.20		[Ok]
121	10841	30.03	0.48	-9.80	13.44		[Ok]
124	10842	30.06	0.45	-9.85	15.68		[Ok]
127	10843	29.93	0.43	-9.79	17.92		[Ok]

Max delta = 32.87 ms at packet no. 2620
Max jitter = 0.61 ms. Mean jitter = 0.09 ms.
Max skew = -13.56 ms.
Total RTP packets = 1105 (expected 1105) Lost RTP packets = 0 (0.00%) Sequence errors = 0
Duration 33.12 s (-62 ms clock drift, corresponding to 7985 Hz (-0.19%)

Save payload... | Save as CSV... | Refresh | Jump to | Graph | Player | Next non-Ok | Close

Detecting problems

Click “Player” - “Decode” - “Play”



Avoid Problems

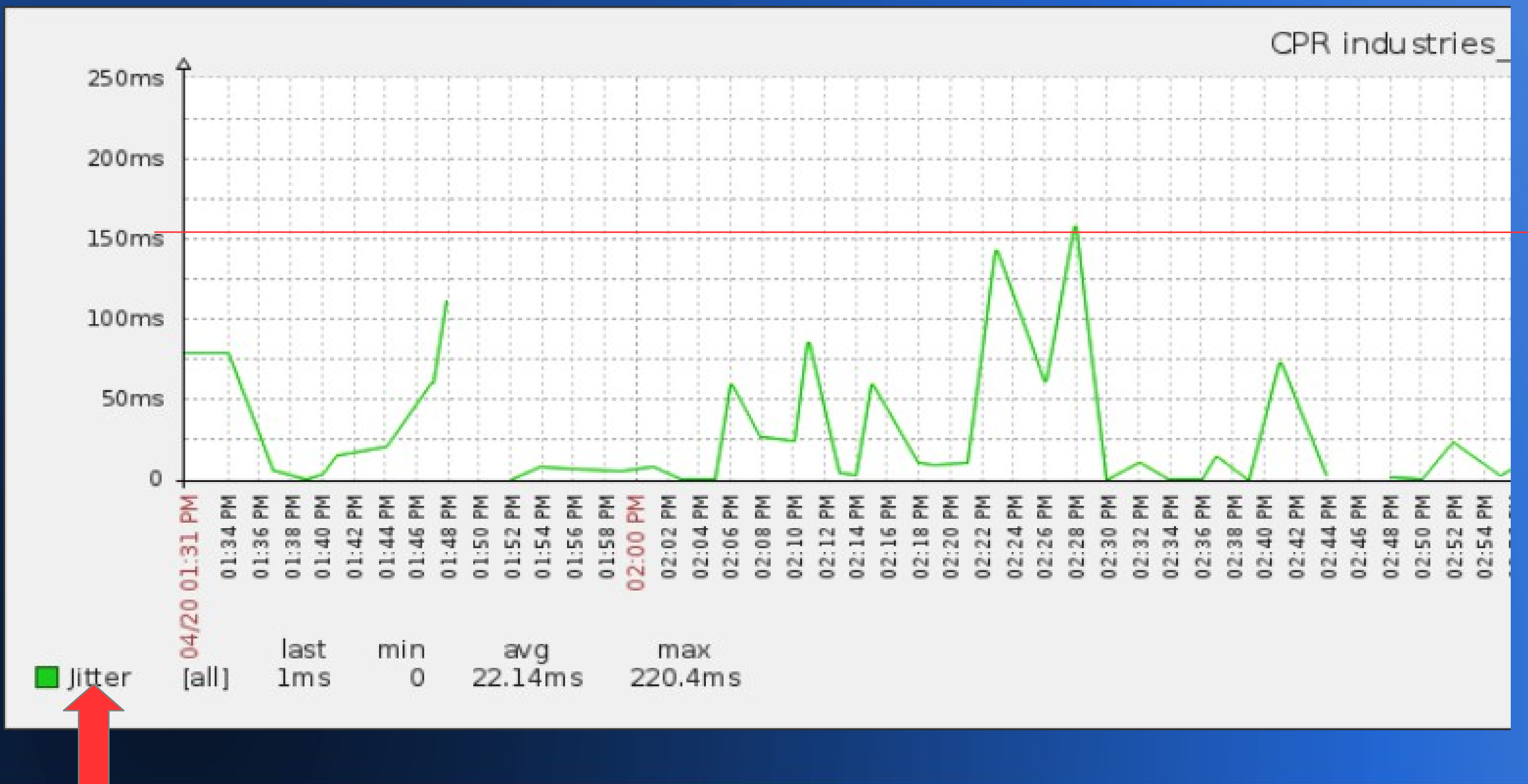
Before on-boarding a customer:

Make sure their internet connection is adequate!

Monitor their WAN

Review the monitoring data!

Avoid Problems

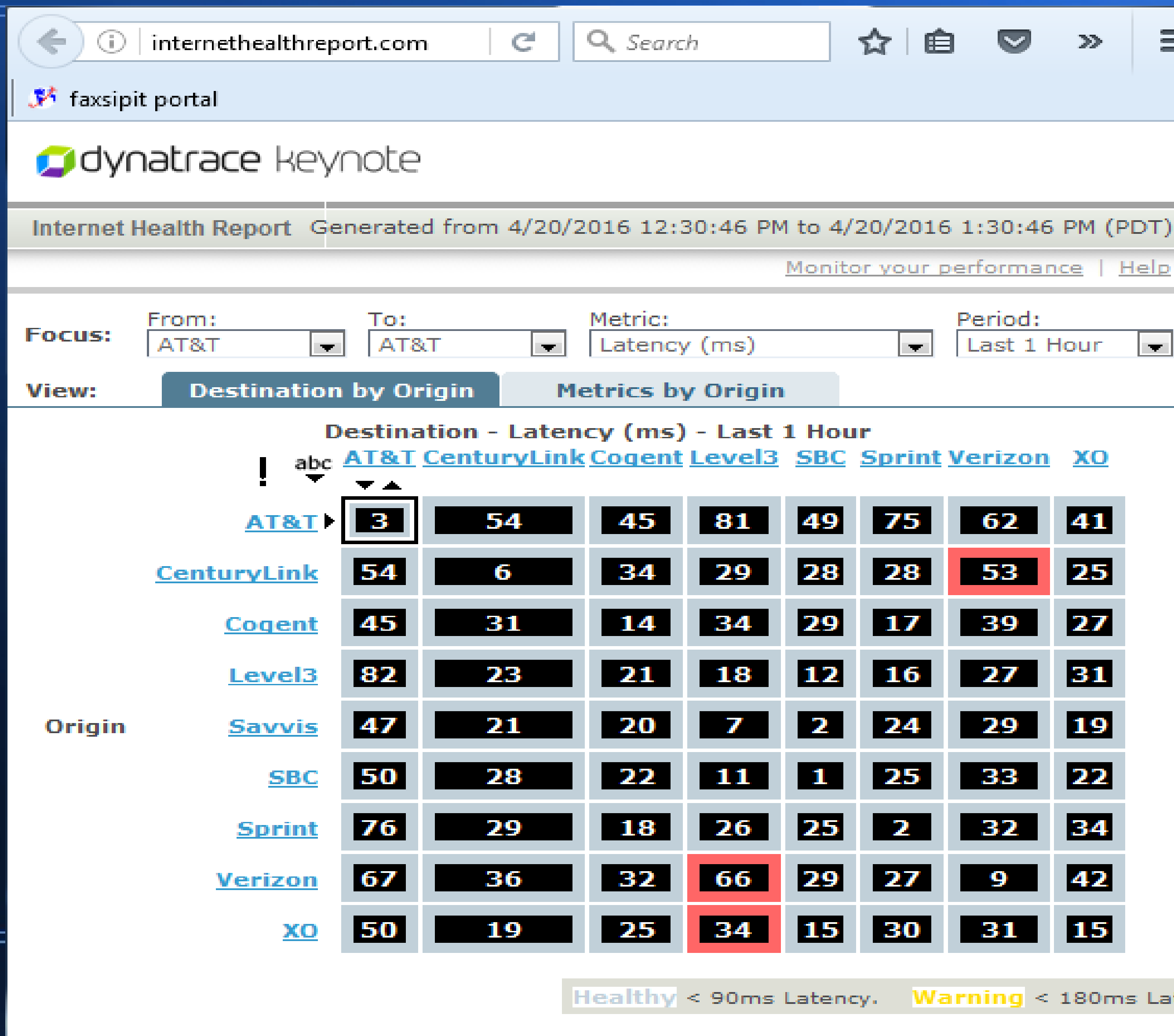


Avoid Problems

- Packet capture every call! www.sipcapture.org

- Also be aware of internet carrier problems
  www.internethealthreport.com

Avoid Problems



internethealthreport.com

Search

faxsipit portal

dynatrace keynote

Internet Health Report Generated from 4/20/2016 12:30:46 PM to 4/20/2016 1:30:46 PM (PDT)

Monitor your performance | Help

Focus: From: AT&T To: AT&T Metric: Latency (ms) Period: Last 1 Hour

View: Destination by Origin | Metrics by Origin

Destination - Latency (ms) - Last 1 Hour

Origin	AT&T	CenturyLink	Cogent	Level3	SBC	Sprint	Verizon	XO
AT&T	3	54	45	81	49	75	62	41
CenturyLink	54	6	34	29	28	28	53	25
Cogent	45	31	14	34	29	17	39	27
Level3	82	23	21	18	12	16	27	31
Savvis	47	21	20	7	2	24	29	19
SBC	50	28	22	11	1	25	33	22
Sprint	76	29	18	26	25	2	32	34
Verizon	67	36	32	66	29	27	9	42
XO	50	19	25	34	15	30	31	15

Healthy < 90ms Latency. Warning < 180ms Lat

Summary

1- Learned about some factors that affect VoIP call quality

2- Learned how to reduce or eliminate call quality issues

3- Leaned how to find issues and diagnose issues

4- Learned how to avoid issues