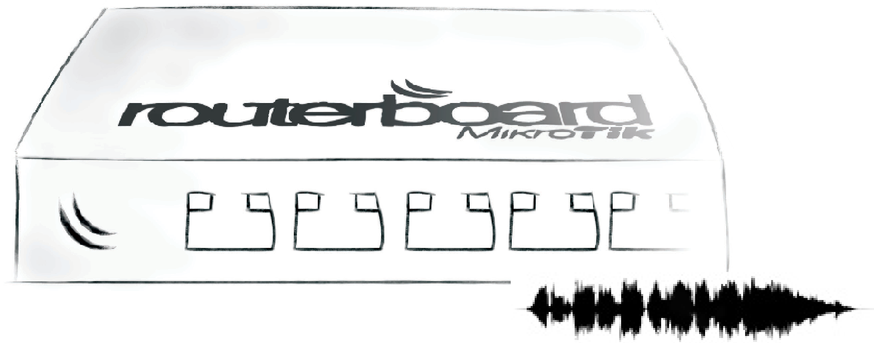


Mastering VoIP in RouterOS

MUM Europe, Zagreb 2013.03.14

Presenter – Andis Arins

- MikroTik Consultant at www.router.lv
- MikroTik Certified trainer for all courses
- Member of the board in Latvian Internet Association

andis[at]router.lv

www.linkedin.com/in/andisarins

What is VoIP

1) analog -> digital
2) transmit over network
3) digital -> analog

Will It Work ?

Codec

G.711 64Kbps (precise speech transmission, low CPU requirements) 87.2Kbps, 50pps, 160b payload

G.729 8Kbps (good speech quality, CPU costly, requires license) 31.2Kbps, 50pps, 20b payload

GSM 13Kbps (acceptable speech quality, available in many hardware/soft platforms) ~36Kbps, 50pps

VoIP Protocol requirements

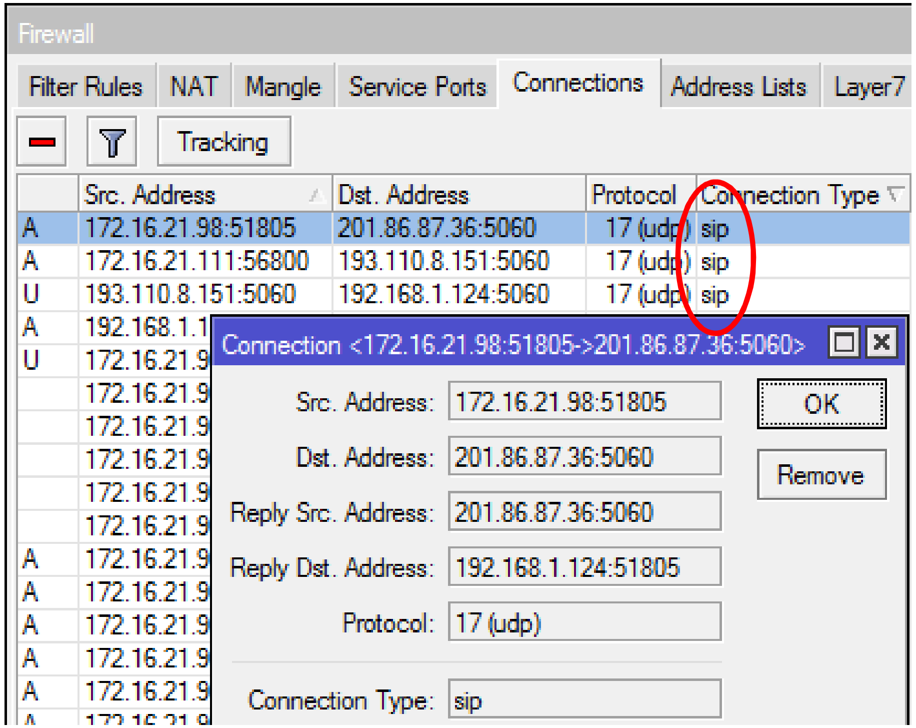
1) Packetloss (evenly < 0-5%)

2) Latency (< 120-150ms one way)

3) Jitter (buffer < 1-3ms)

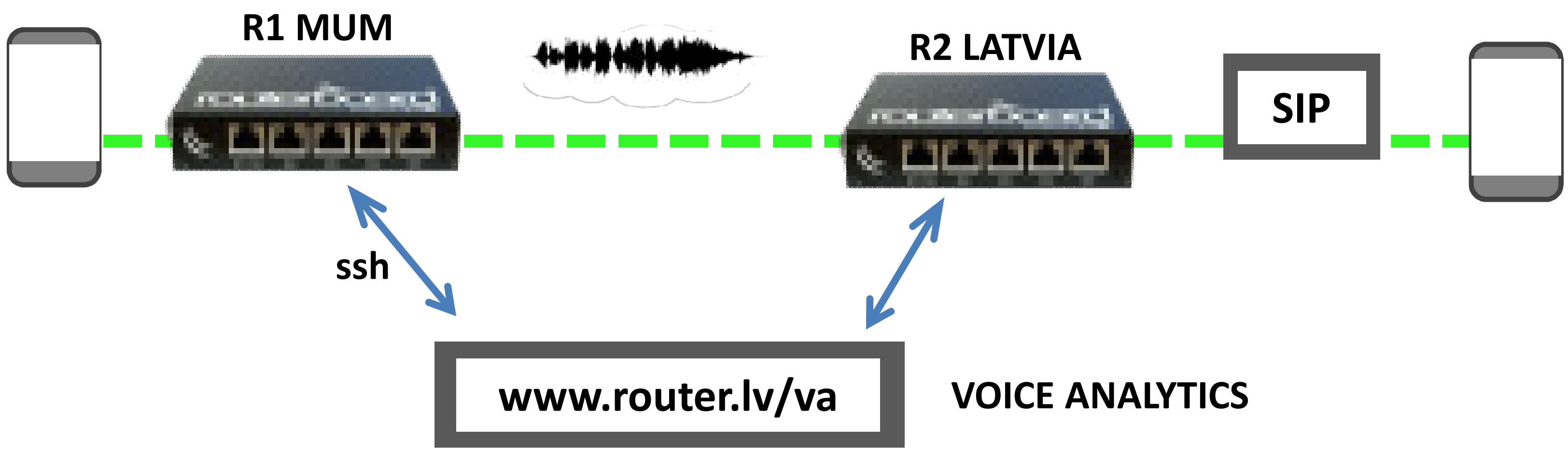
Where to look for VoIP?

1) Connection tracking (conn-type SIP, Q931);
2) Torch tool
3) Firewall
4) Packet sniffer



Phone a friend

VoIP protocol: SIP, transport protocol UDP



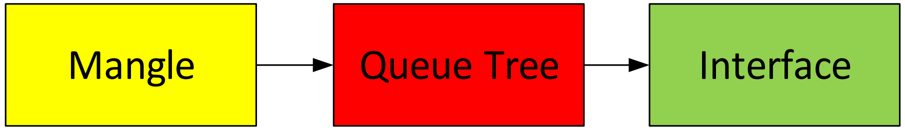
SIP call analysis

A - **INVITE** sip:67317700@193.110.8.151 SIP/2.0
B – SIP/2.0 100 **Trying**
B - SIP/2.0 183 **Session Progress**
B – SIP/2.0 200 **OK**
A - **ACK** sip:67317700@193.110.8.151 SIP/2.0

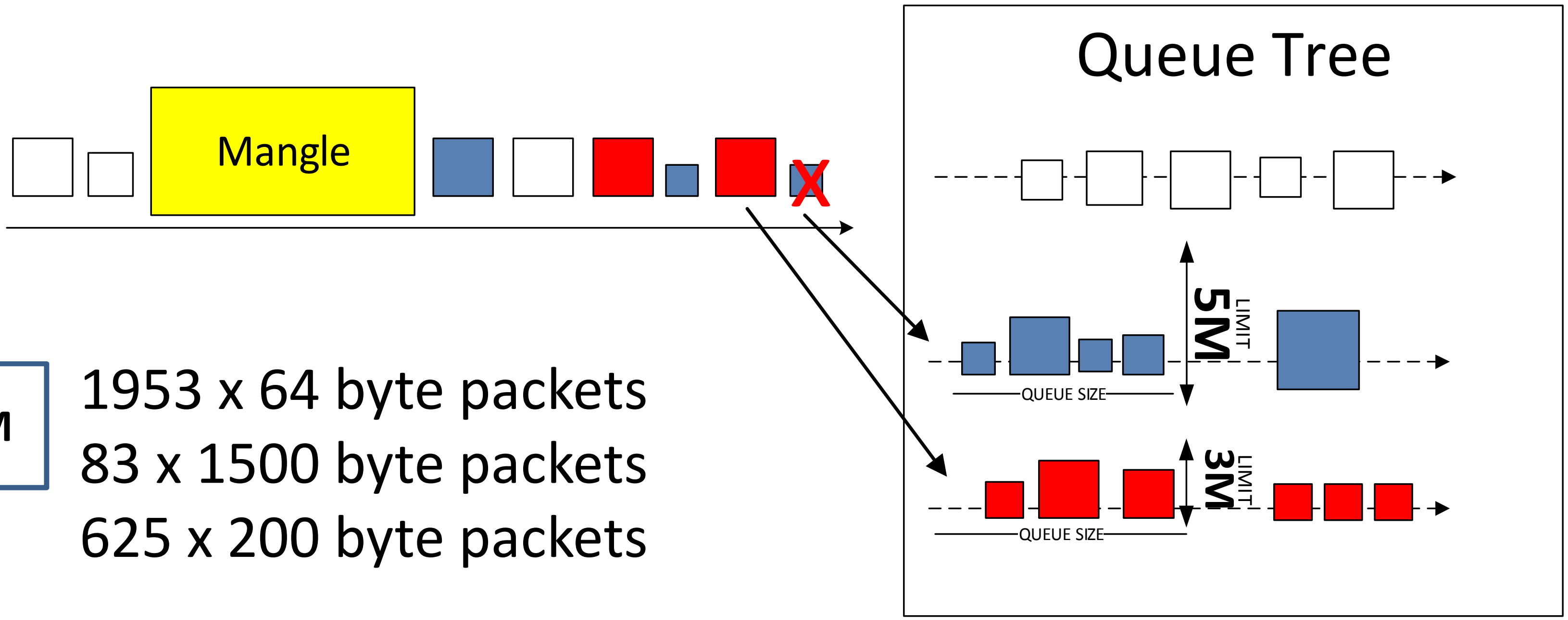
… RTP DATA …

B – **BYE** sip:67796504@64.134.178.65;ob SIP/2.0
A - SIP/2.0 200 **OK**

Packet Flow and VoIP packets



200 byte 0.00004s – trip in RB750G



1M

1953 x 64 byte packets

83 x 1500 byte packets

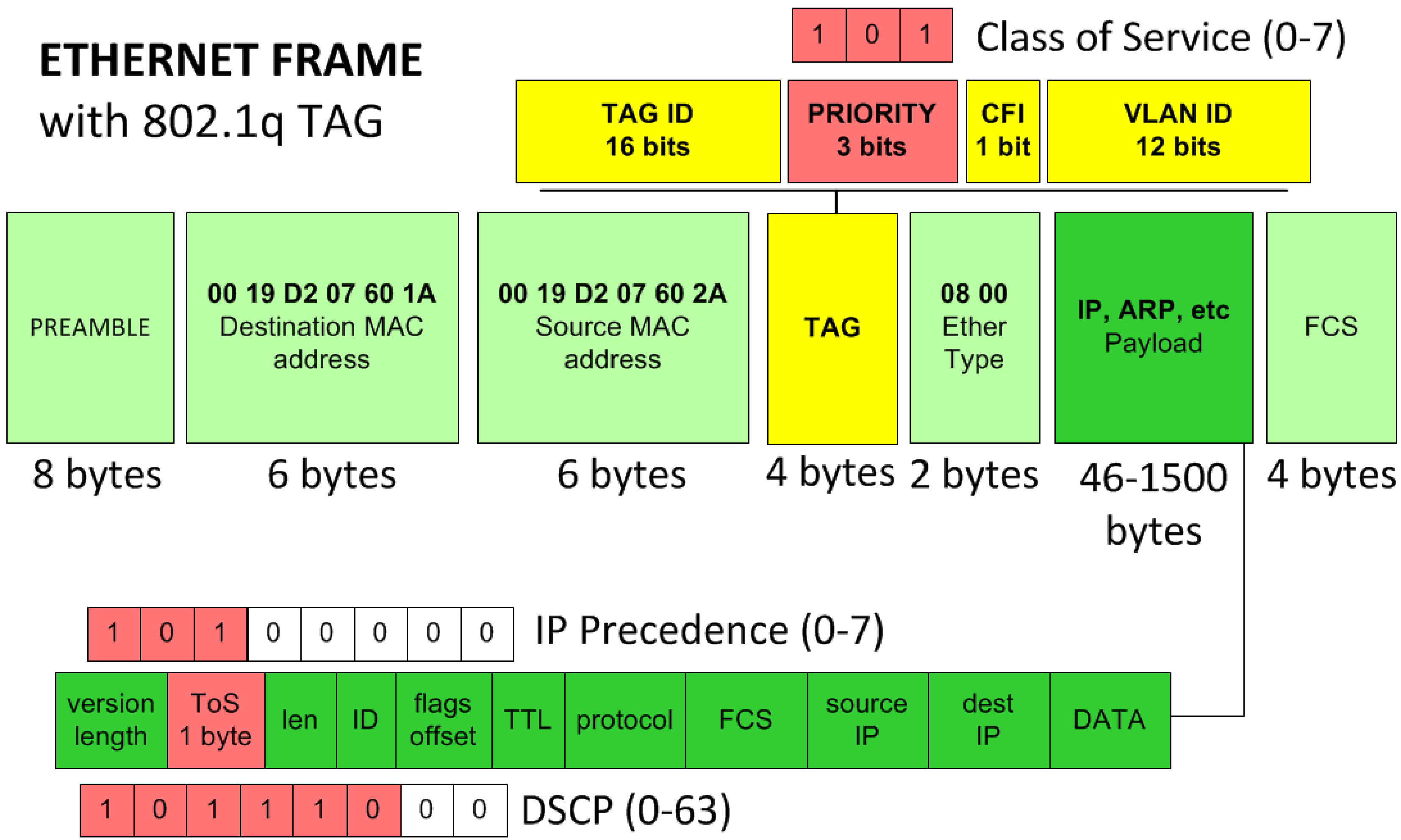
625 x 200 byte packets

Mark a packet


```
/ip firewall mangle
add action=mark-packet chain=prerouting disabled=no dst-address=193.110.8.151 \
    dst-port=5060 new-packet-mark=packet-sip-signaling passthrough=no \
    protocol=udp
add action=mark-packet chain=prerouting disabled=no dst-address=193.110.8.151 \
    new-packet-mark=packet-sip-voice passthrough=no protocol=udp
```

1) src/dst interface
2) src/dst ip addres/port
3) connection-type
4) DSCP
5) layer7

QoS can travel with packet

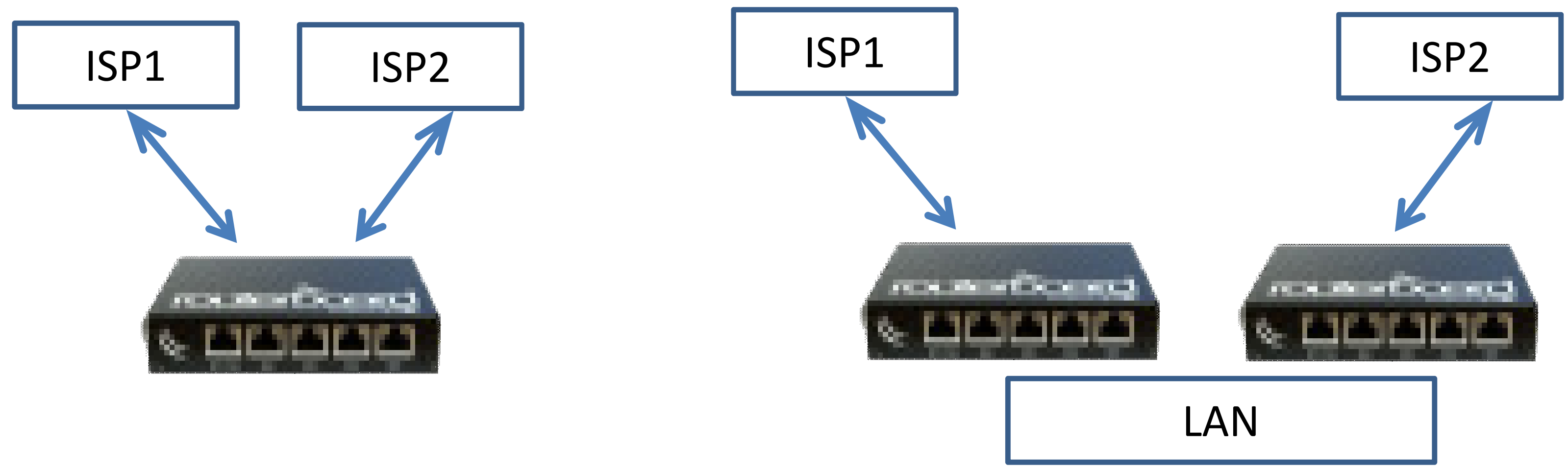


QoS in RouterOS


```
/queue tree
add burst-limit=0 burst-threshold=0 burst-time=0s disabled=no limit-at=0 max-limit=5M name=\
    total-uplink packet-mark="" parent=WAN priority=8
add burst-limit=0 burst-threshold=0 burst-time=0s disabled=no limit-at=2M max-limit=4M name=\
    other packet-mark=no-mark parent=total-uplink priority=8 queue=wireless-default
add burst-limit=0 burst-threshold=0 burst-time=0s disabled=no limit-at=2M max-limit=3M name=\
    voip-upload packet-mark=voip-upload parent=total-uplink priority=1 queue=default
```

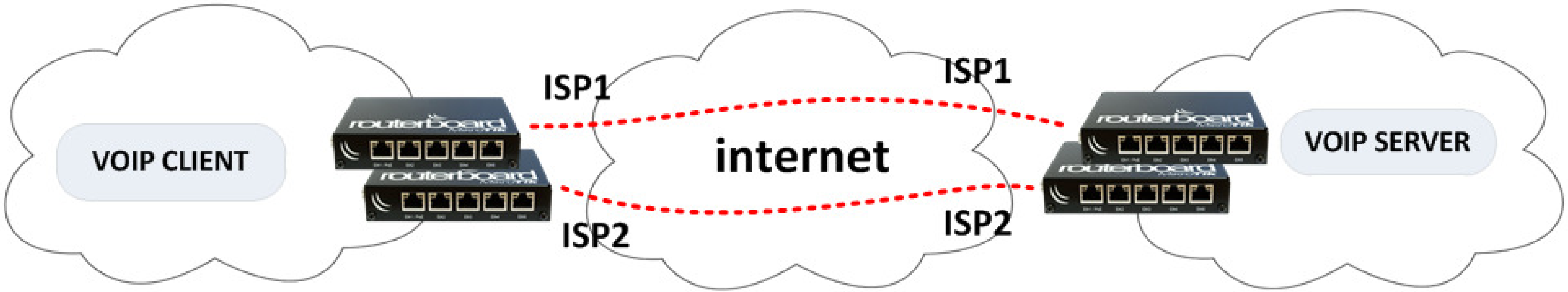
tip - use only 80% of promised

VoIP design



1) Redundant internet (static/BGP)
2) Redundant router (VRRP)
3) Tunneled connection (SSTP, IPSEC)

VoIP design



Full redundancy, several links

Active monitoring with simulated packets

Thank You!