



Mastering VoIP in RouterOS

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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

The screenshot shows the Mikrotik WinBox Firewall interface. The 'Connections' tab is active, displaying a table of active connections. A red circle highlights the 'Connection Type' column, which shows 'sip' for several entries. A detailed view of a selected connection is shown in a pop-up window.

	Src. Address	Dst. Address	Protocol	Connection Type
A	172.16.21.98:51805	201.86.87.36:5060	17 (udp)	sip
A	172.16.21.111:56800	193.110.8.151:5060	17 (udp)	sip
U	193.110.8.151:5060	192.168.1.124:5060	17 (udp)	sip
A	192.168.1.1			
U	172.16.21.9			
	172.16.21.9			
	172.16.21.9			
	172.16.21.9			
	172.16.21.9			
A	172.16.21.9			
A	172.16.21.9			
A	172.16.21.9			
A	172.16.21.9			
A	172.16.21.9			
A	172.16.21.9			

Connection <172.16.21.98:51805->201.86.87.36:5060>

Src. Address: 172.16.21.98:51805

Dst. Address: 201.86.87.36:5060

Reply Src. Address: 201.86.87.36:5060

Reply Dst. Address: 192.168.1.124:51805

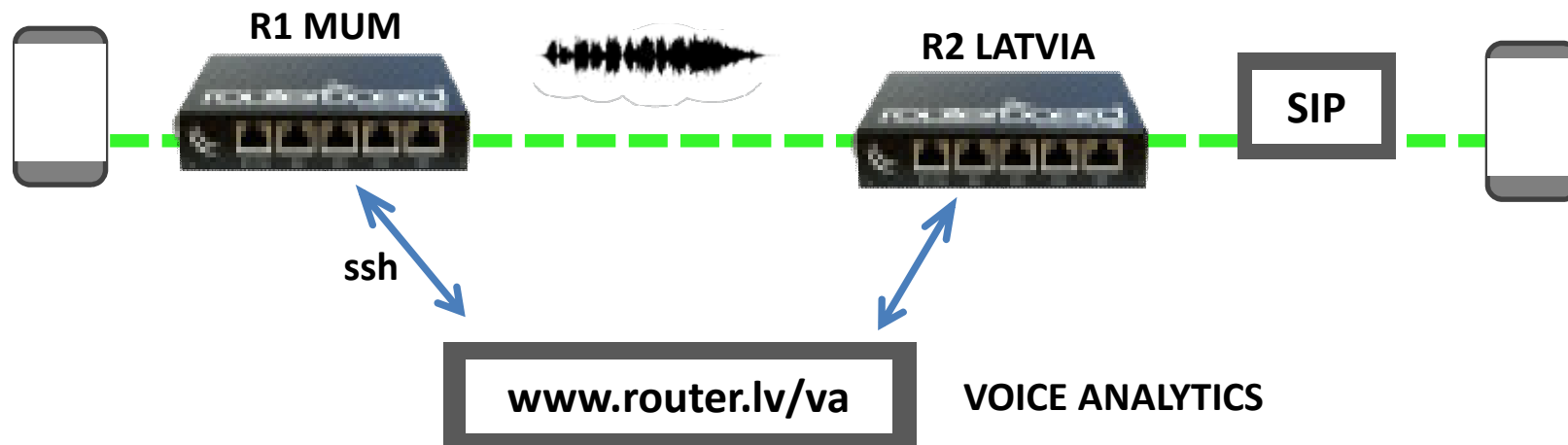
Protocol: 17 (udp)

Connection Type: sip

Phone a friend



VoIP protocol: SIP, transport protocol UDP



SIP call analysis



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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

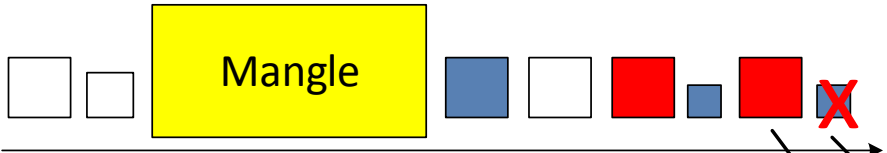
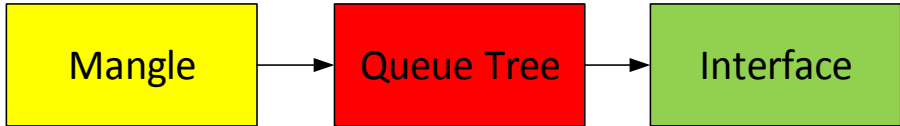
... **RTP DATA** ...

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

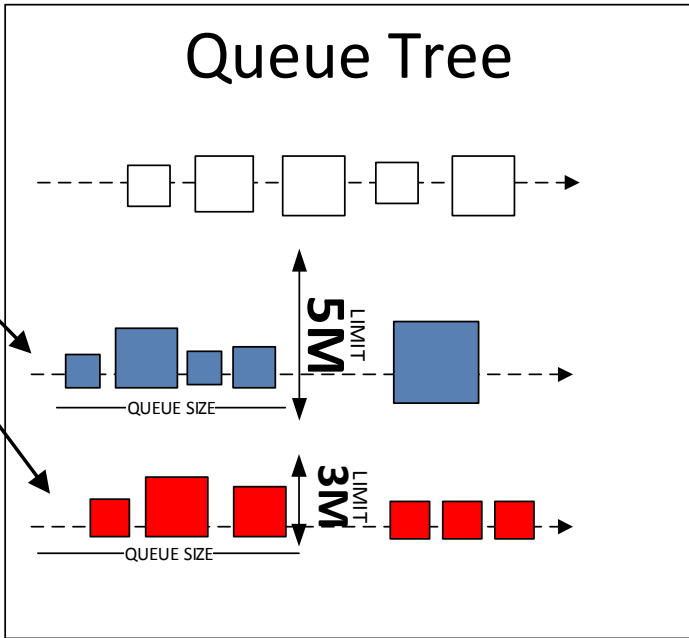
Packet Flow and VoIP packets



200 byte 0.00004s – trip in RB750G



1M 1953 x 64 byte packets
83 x 1500 byte packets
625 x 200 byte packets



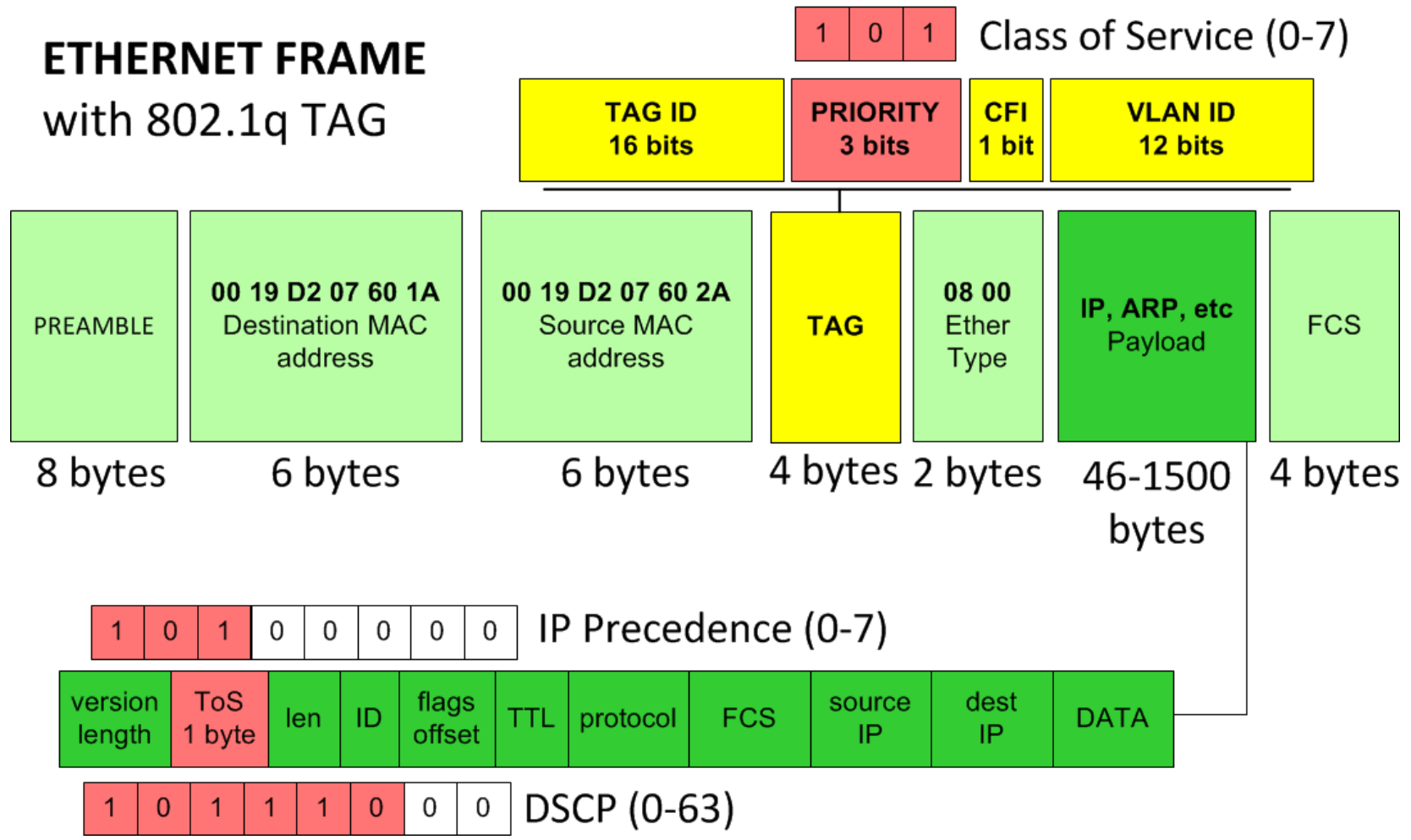
Mark a packet



```
/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 address/port
- 3) connection-type
- 4) DSCP
- 5) layer7

QoS can travel with packet



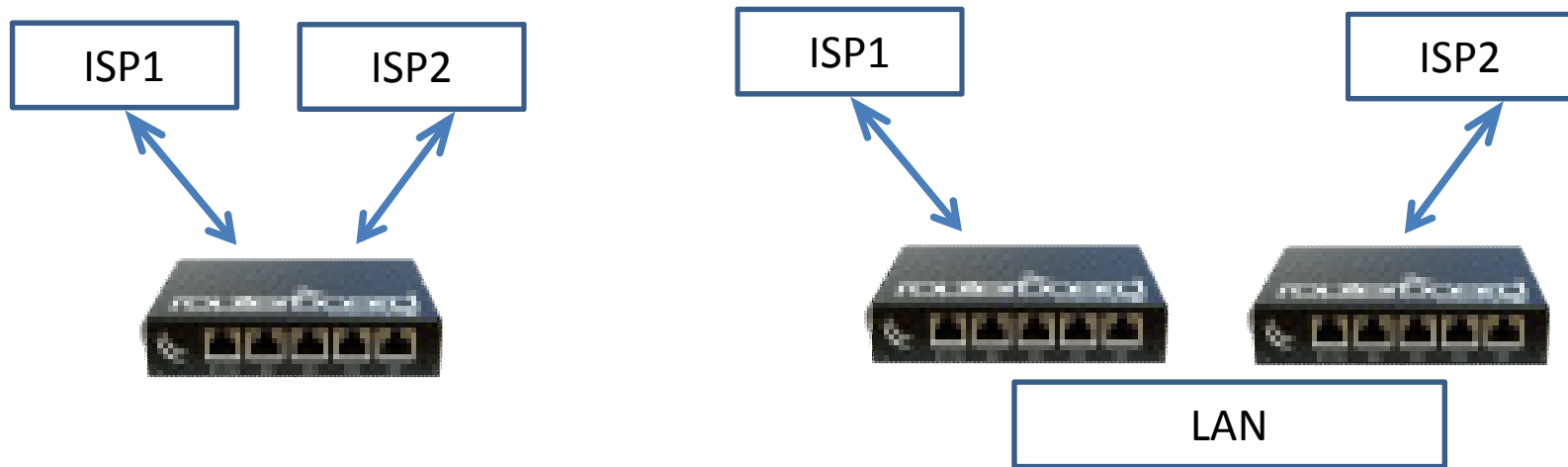
QoS in RouterOS



```
/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
```

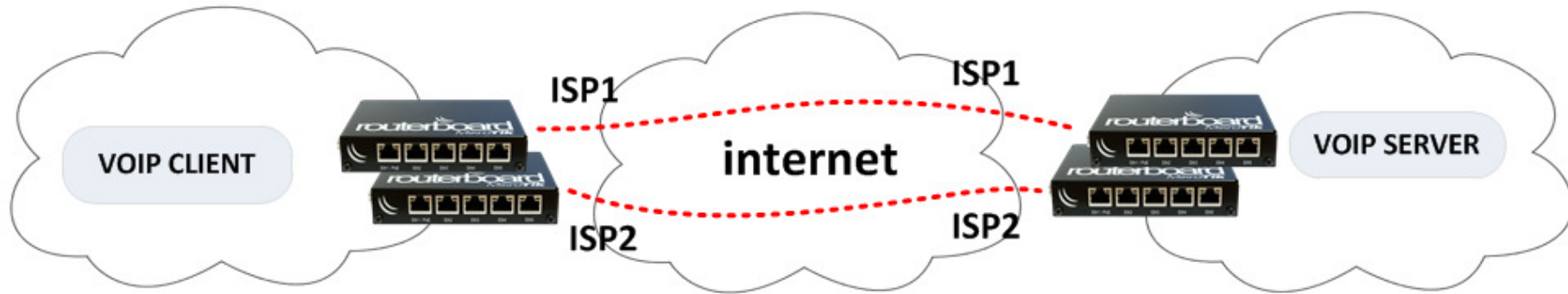
tip - use only 80% of promised

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!

