



Quality of Service (QoS) in VoIP networks

Mangle, Queue Tree

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Who is SDC B.V. ?

- Distributor in ICT components
- Founded in 1992, located in Doetinchem(NL)
- Family company
- Supportive with solutions and service
- ICT partnerchannel in the Netherlands



Marketing&sales



Logistics



Academy



Professional services



Quality of Service in VoIP networks

- Quality of Service(QoS) introduction
- Circuit switched vs Packet switched networks
- QoS criteria for Voice over IP
- Problems in VoIP environments
- QoS traffic control
- Hierarchical Token Bucket (HTB)
- Qualify VoIP traffic with Mangle
- HTB implementation Queue Tree

Quality of Service(QoS)

Quality of Service(QoS) is a general term that describes the measurement of the overall performance of a service such as telephony, video, file transfer or other network services.

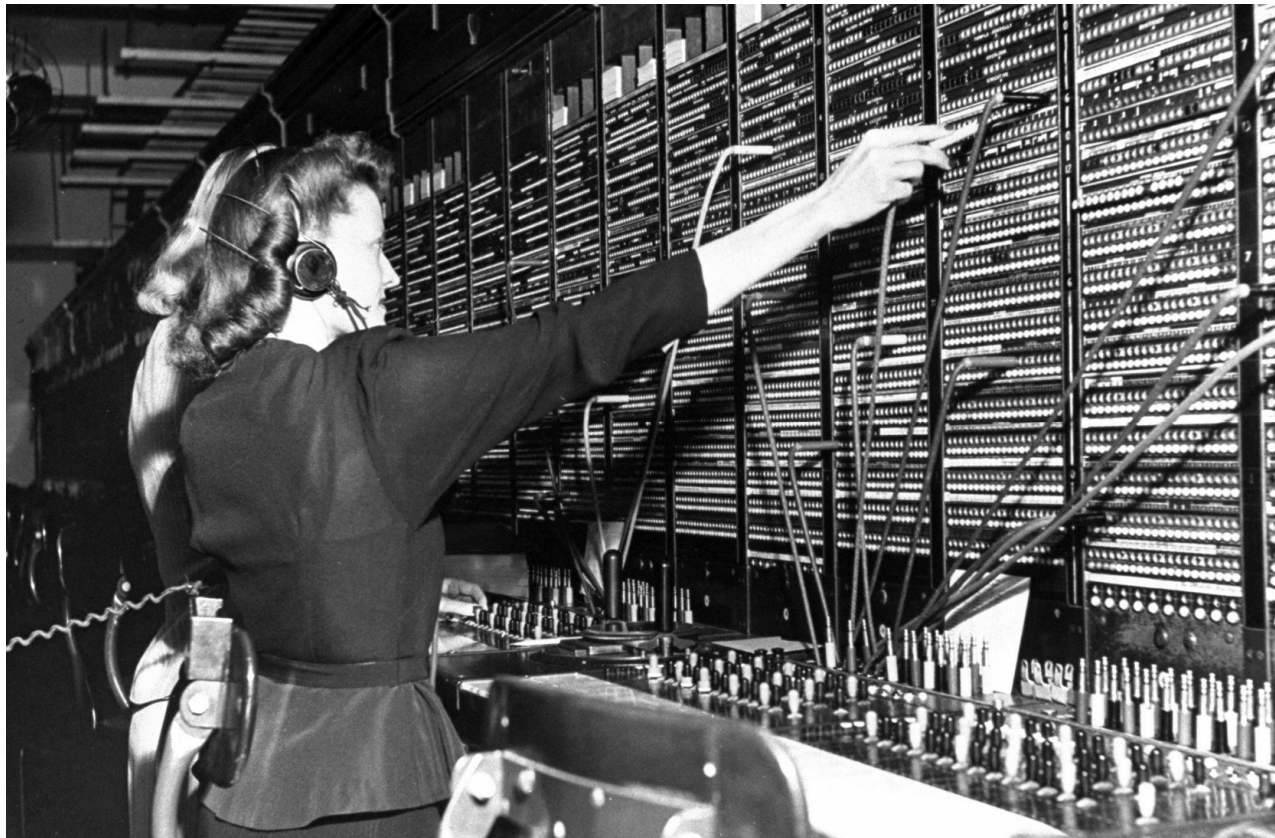
In the field of **computer networking**, quality of service refers to **traffic prioritization** and **resource reservation** control mechanisms. QoS is needed in order to **guarantee** a certain level of **performance** to a data flow

Why the need for Quality of Service(QoS)

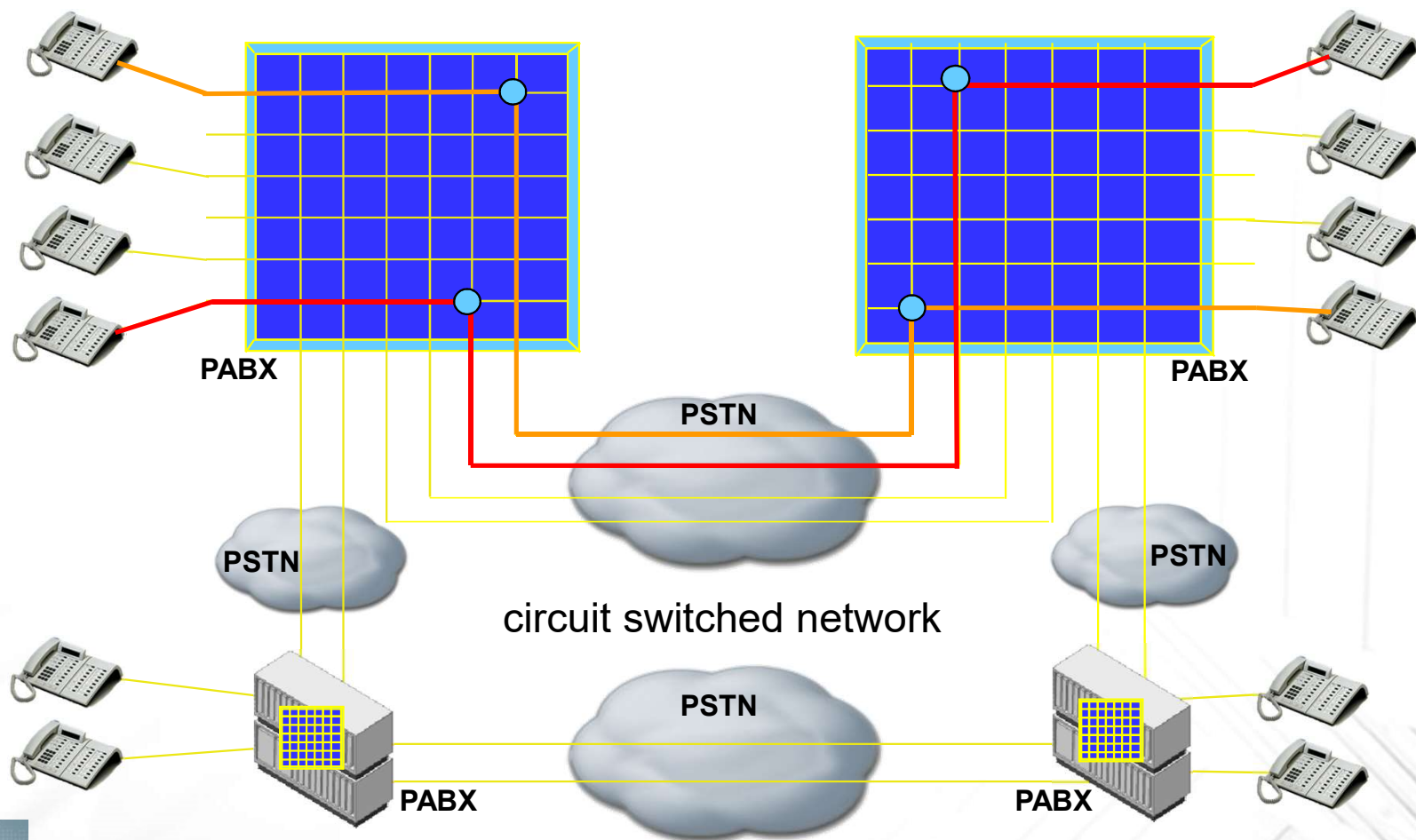
For instance Quality of Service is necessary to guarantee the bandwidth of an IP-phone. A phone call does not need much bandwidth but must always be instantly available for phone calls!



Telephony history



Circuit switched network (64kbit PCM telephony)

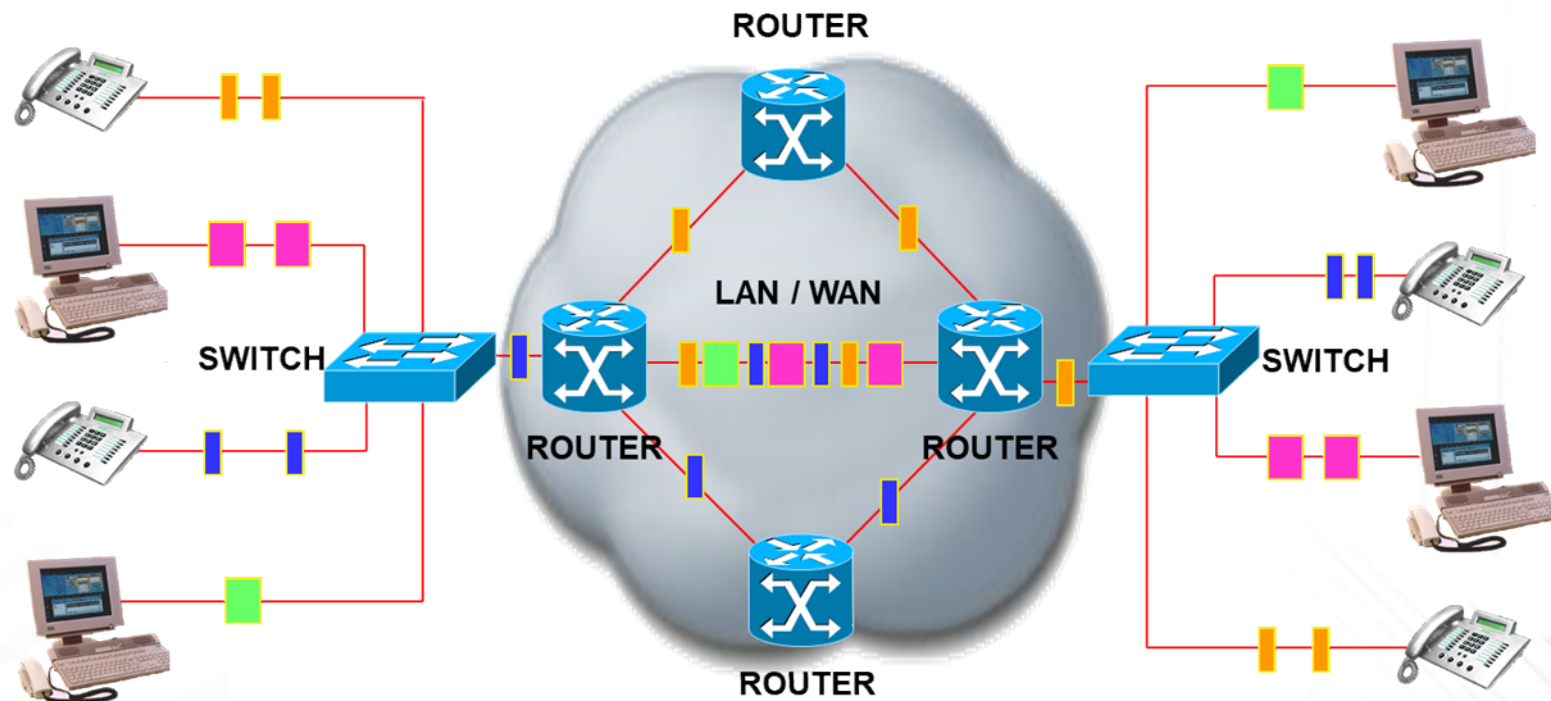


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Circuit switched network

- A **direct(fysical) connection** is made between phones. For setting up the circuit(call setup) a **seperate signalling network** is nessesary.
- A circuit switched network has a fixed(short) latency and for every session a fixed bandwidth. Circuit switched networks by standard have **QoS**.
- Unused bandwith within a circuit is unavailable for other services. Circuit switched networks are **unefficient** in this regard.

Packet switched network



Packet switched network

- IP phones exchange voice data over a **shared network**
- Since there is **no real connection**(circuit) the voice packets are **coded(IP)** so they can be routed through the network to the destination
- To **guarantee QoS** for telephony traffic identification, shaping en prioritizing has to be performed end to end.
- Services like speech, video and data share the same fysical network enabling **more efficient** use of network **bandwith**.

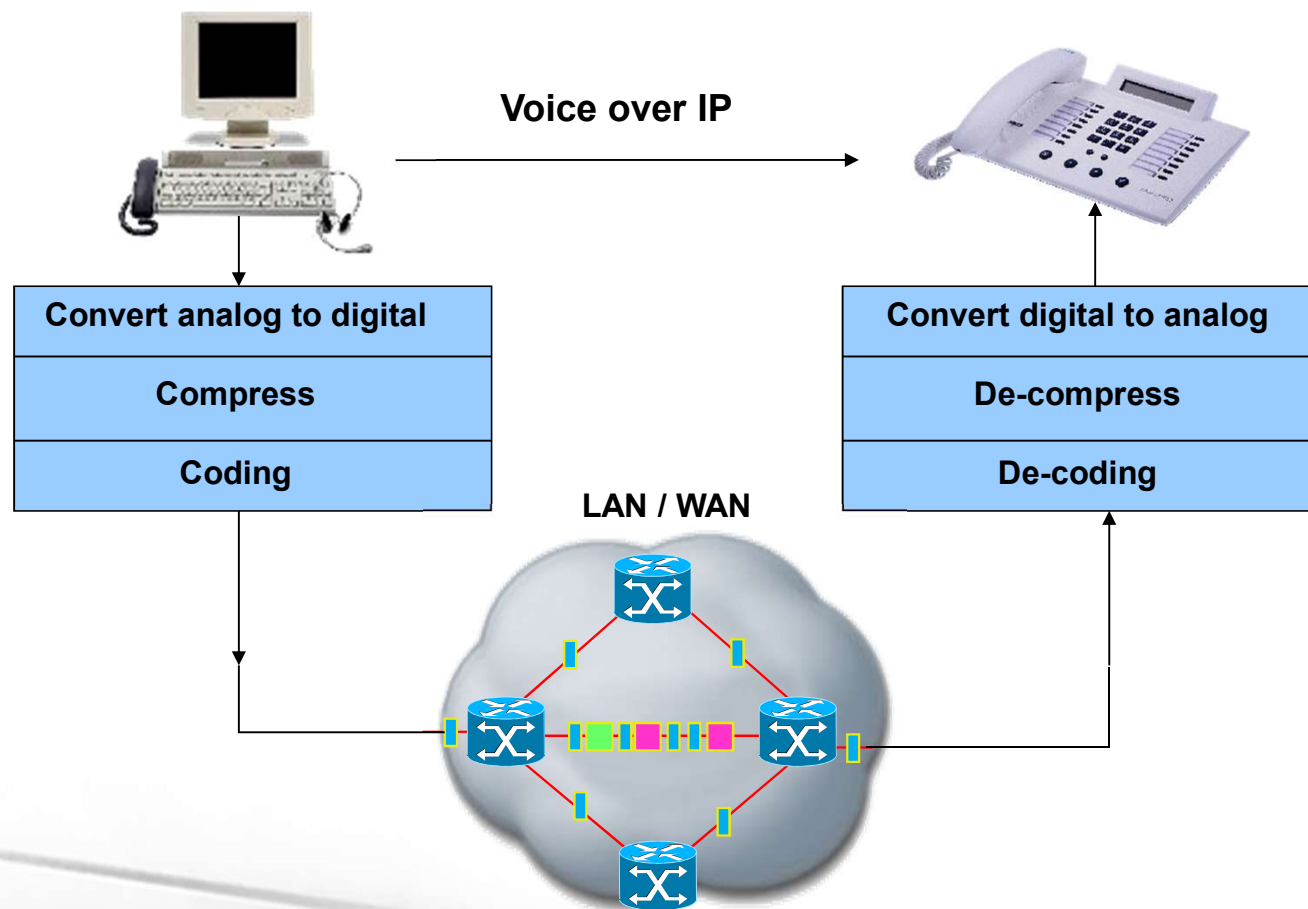
QoS criteria for Voice over IP

- Sufficient bandwidth
- Availability of bandwidth(at all times)
- Low packet loss
- Low round trip delay(low latency)
- Minimal Jitter

Problems in VoIP environments

- The quality of a VoIP call is heavily dependent on the network environment.
- **Latency, Jitter, & Packet Loss** can cause quality of experience issues for **Voice over IP (VoIP)** phone calls
- **Packet loss** is very common in IP networks, but certain networks such as WiFi can be particularly prone to high levels of packet loss.

Bandwith usage



Bandwidth usage

Bandwidth usages in SIP for						
code	Bandwidth for speech	+	Bandwidth for "IP" - Overhead	=	one cannel	bi-directional connection
G.711	64 kBit/s		ca. 19 kBit/s		ca. 83 kBit/s	ca. 166 kBit/s
G.723.1	6,3 kBit/s		ca. 19 kBit/s		ca. 25 kBit/s	ca. 50 kBit/s

RTP(Realtime Transport Protocol)

SIP uses RTP for the end to end transport of audio and video data(**payload**). RTP adds extra information to the UDP header and in some cases can replace the UDP header.

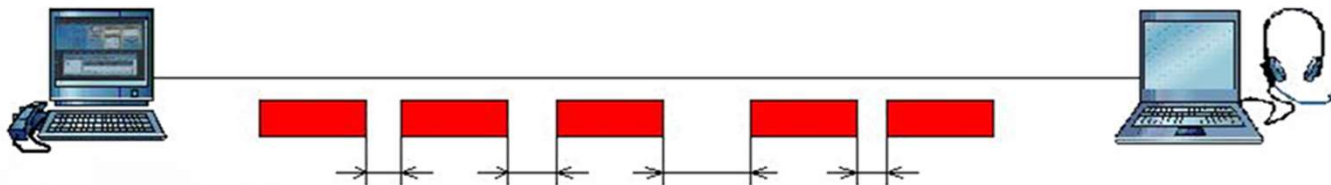
- Sequence number
- Timestamp
- Payload type

RTCP(Realtime Transport Control Protocol)

RTP functionality is expanded by Real Time Control Protocol (RTCP). The goal of RTCP is to provide feedback to the VoIP endpoints about the quality of the data transmission. RTCP regularly sends reports with receipt statistics.

These statistics contain packet loss and jitter.

VoIP endpoints use this information to adapt the jitter buffer and/or the used audiocodec.



Latency

- **Latency** is the time it takes the RTP (media) packets to traverse the network. Too much latency causes callers to speak over the top of each other.
- Callers start to notice the effect of **latency** around 250ms, above ~600ms the experience is unusable. There will always be some latency, the objective is to minimize it and keep total roundtrip time below 250ms.
- Ideally latency should be below 100ms because, while it is noticable at 250ms, other services and issues beyond your control might add delay causing the cumulative total to be over 250ms.

Jitter

- **Jitter** is when the latency through the network is not fixed but varies. Jitter causes that packets don't arrive in the same order they were sent. For small amounts of jitter, this can be resolved in the **jitter**buffer. The length of the jitter buffer introduced causes increased latency.
- Too much jitter cannot be resolved by a reasonable length jitter buffer without introducing too much delay, so instead results in **jitter induced packet loss** causing choppy audio.

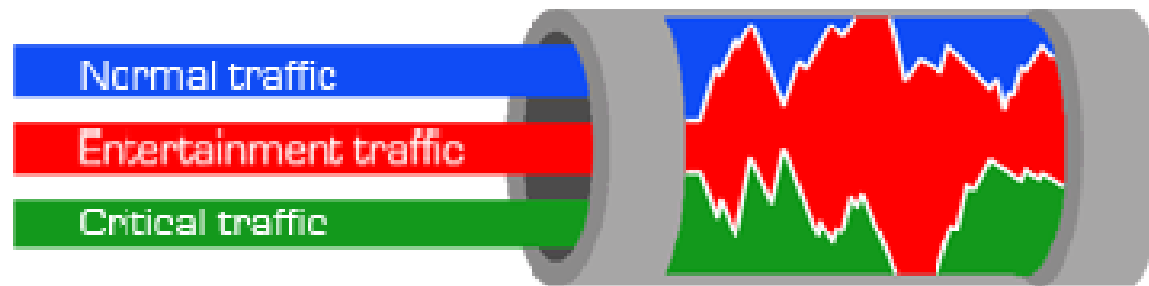
QoS traffic control



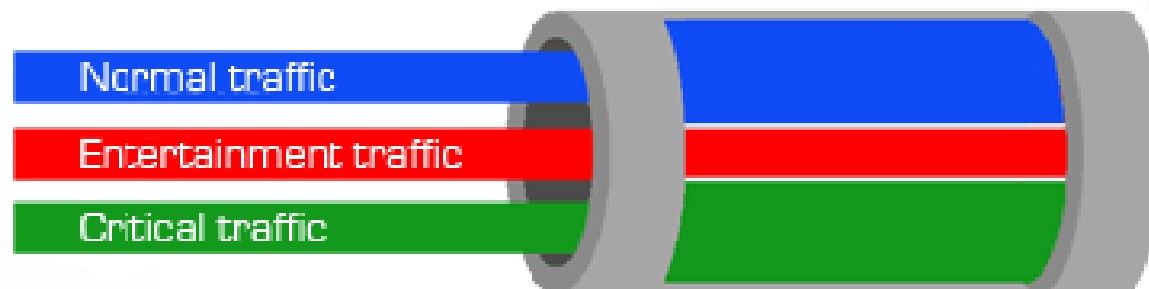
- Traffic shaping
- Prioritizing

QoS traffic shaping.

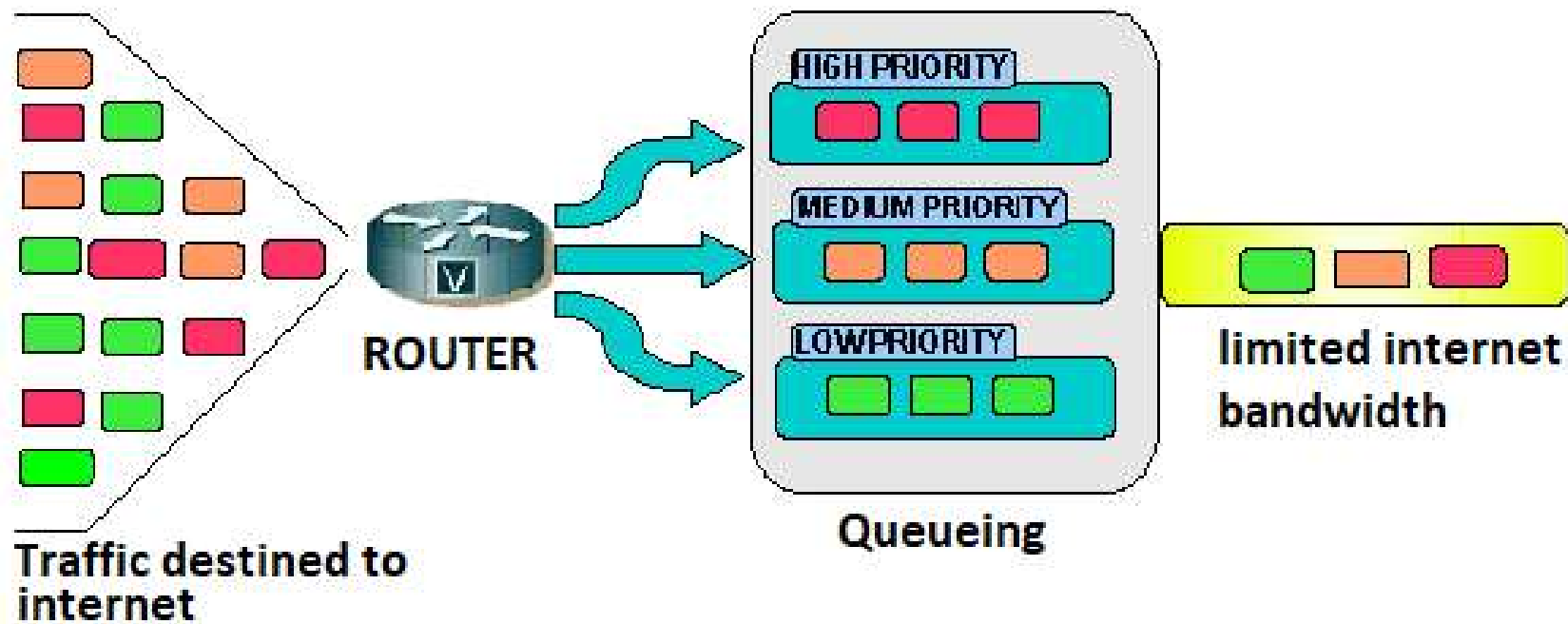
Bandwidth Use without QoS control



Bandwidth Use with QoS control



QoS traffic prioritizing



QoS traffic control

Traffic control is done on the **outbound interface** (we have no direct control over traffic that is being sent to us)

- **Rate limiting** is done by **dropping** some low priority packets so we have capacity for higher priority packets
- We need to know how much bandwidth is available
- We are not reordering the packets! The packets will leave the router in the **exact sequence** as they are received (provided that we are forwarding them)

Hierarchical Token Bucket (HTB)

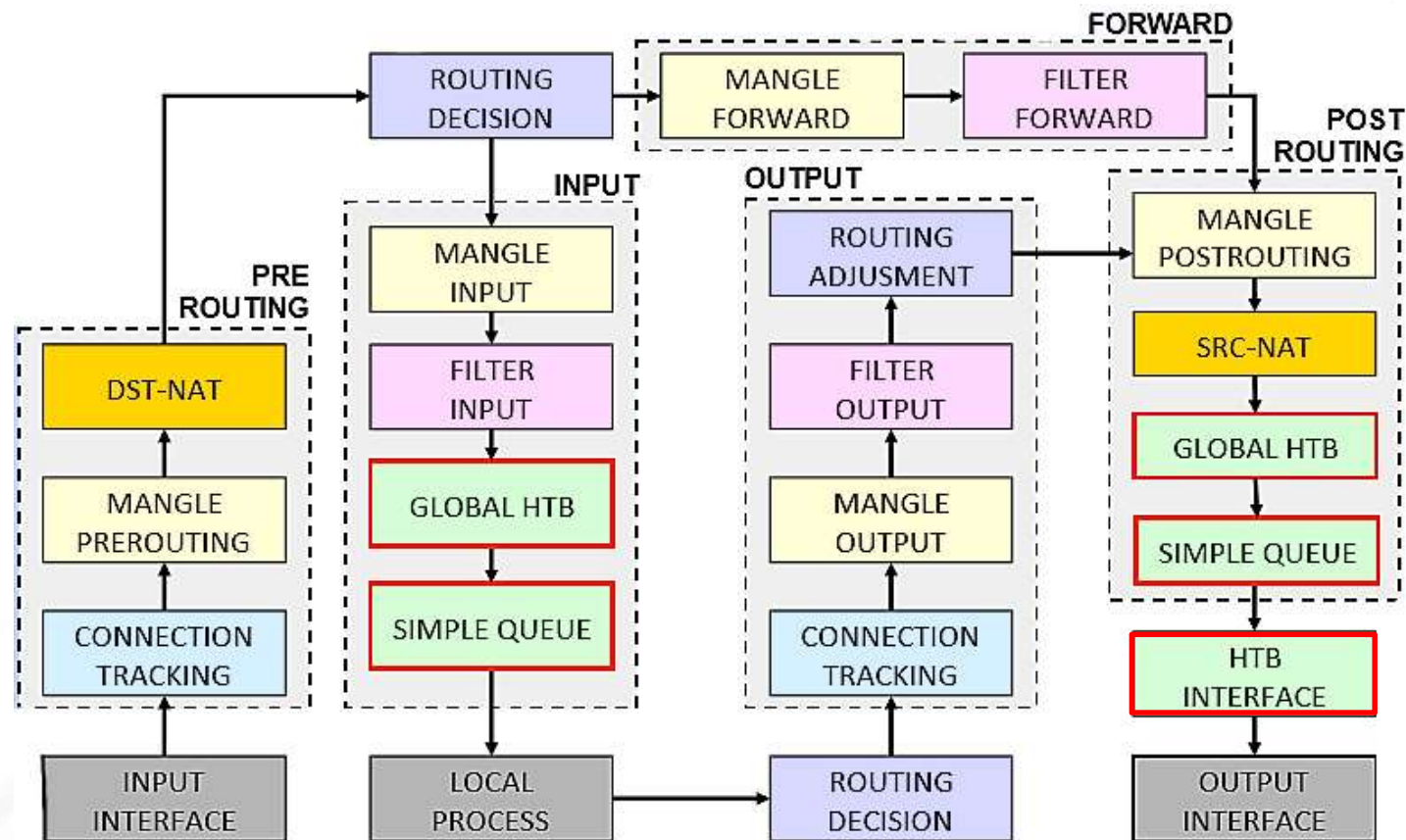
Hierarchical Token Bucket (HTB) is a classful queuing method for handling different kinds of traffic.

- All QoS implementation in **RouterOS** is based on Hierarchical Token Bucket.
- HTB provides a way to create a hierarchical queue structure and determine relations between parent and child queues and relations between child queues.

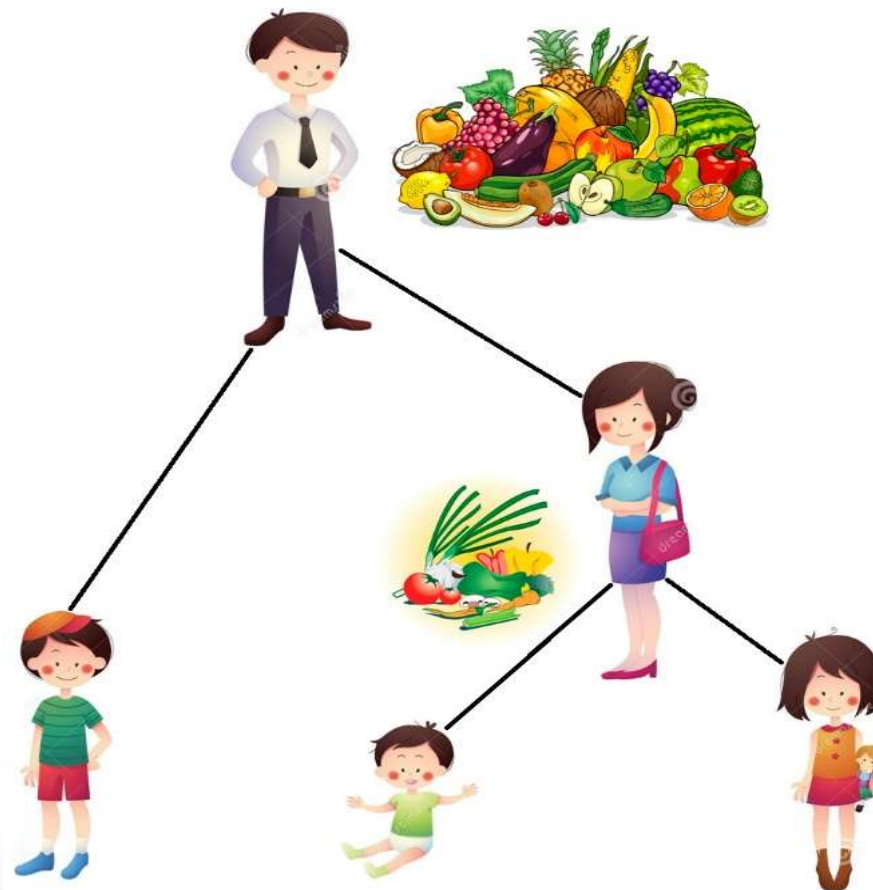
Hierarchical Token Bucket (HTB)

- When a packet travels through the router, it passes **2 HTB trees (global and interface)** and simple queues
- When a packet travels to the router, it passes only **global HTB** and simple queues.
- When a packet travels from the router, it passes **2 HTB trees** and simple queues.

HTB packet flow routerOS V6



HTB queue structure(children and parents)



HTB queue structure (children and parents)

- As soon as queue has at least one child it becomes a parent queue.
- All child queues (it doesn't matter how many levels of parents they have) are on the same bottom level of HTB
- Child queues do the actual traffic consumption, parent queues are only responsible for traffic distribution

HTB rate limiting

HTB has two rate limits:

- **CIR (Committed Information Rate)** – (**limit-at** in RouterOS) worst case scenario, the flow will get this amount of traffic no matter what (assuming there is enough bandwidth)
- **MIR (Maximal Information Rate)** – (**max-limit** in RouterOS) best case scenario, rate that the flow can get up to, if the queue's parent has spare bandwidth available

At first HTB will try to satisfy every child queue's **limit-at** only then it will try to reach max-limit

HTB rate limiting

Maximal rate of the parent should be equal or bigger than sum of committed rates of the children

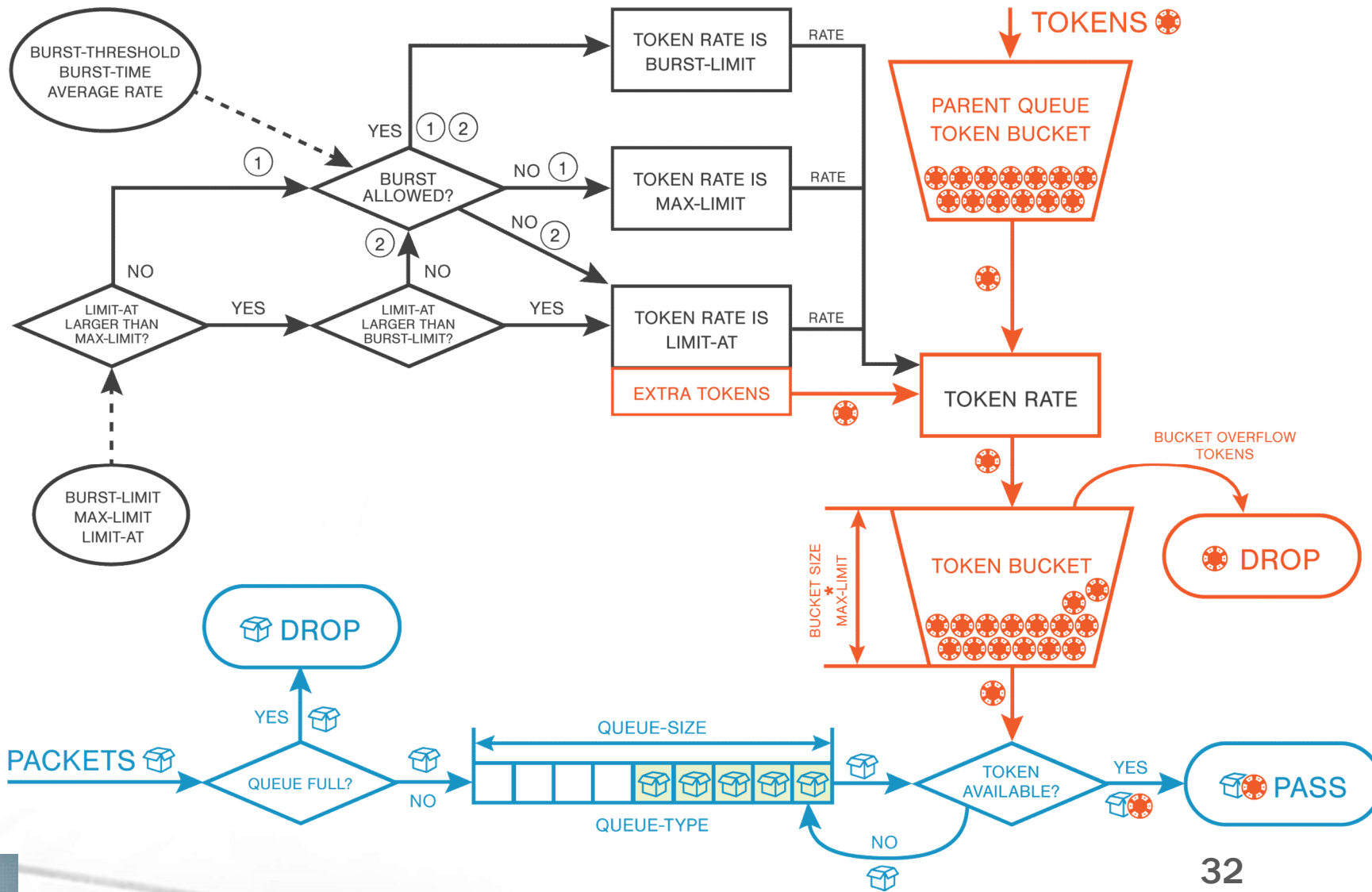
- $MIR(\text{parent}) \geq CIR(\text{child1}) + \dots + CIR(\text{childN})$

Maximal rate of any child should be less or equal to maximal rate of the parent

- $MIR(\text{parent}) \geq MIR(\text{child1})$
- $MIR(\text{parent}) \geq MIR(\text{child2})$
- $MIR(\text{parent}) \geq MIR(\text{childN})$

HTB priority

- Works only for child queues to differentiate them.
- 8 is the lowest priority, 1 is the highest priority.
- Queue with higher priority will get the chance to satisfy its max-limit before other queues.
- Actual traffic prioritization will only work if limits are specified. A queue without limits will not prioritize anything.



QoS example scenario HTB



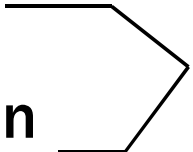
Gigaset N510 IP Pro

- 6 DECT handsets
- 6 SIP registrations
- 4 simultaneous SIP calls

QoS example scenario HTB



Qualify VoIP traffic with Mangle

- Identifying VoIP traffic
 - Marking VoIP connections
 - Marking VoIP packets within connection
- 
- Reduce CPU load

VoIP call consists of 2 kinds of traffic

- Call setup and signalling (UDP 5060)
- RTP Voice payload (UDP ports depended on VoIP system)

Qualify VoIP traffic with Mangle

VoipBuster SIP settings:

- SIP port : 5060
- Registrar : sip.voipbuster.com
- Proxy server : sip.voipbuster.com

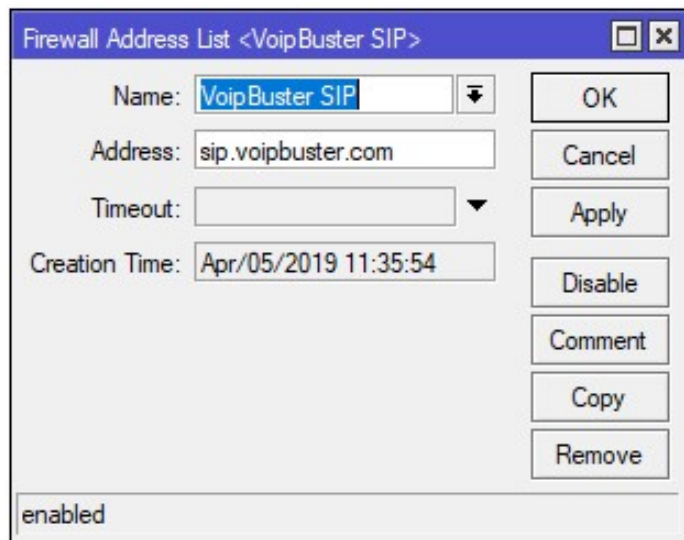
Gigaset N510 IP Pro RTP port numbers:



The screenshot shows a configuration window titled "Listen ports voor VoIP-verbindingen". It contains a section "Willekeurige poorten gebruiken:" with two radio buttons: "Ja" (selected) and "Nee". Below this, there are two rows of port configuration. The first row is for "SIP-poort:" with input fields for "5060" and "5076" separated by a hyphen. The second row is for "RTP-poort:" with input fields for "5004" and "5020" separated by a hyphen.

Listen ports voor VoIP-verbindingen	
Willekeurige poorten gebruiken: ☒ Ja ☐ Nee	
SIP-poort:	5060 - 5076
RTP-poort:	5004 - 5020

Qualify VoIP traffic with Mangle



The screenshot shows the 'Firewall Address List <VoipBuster SIP>' window. It contains the following fields and buttons:

- Name: VoipBuster SIP
- Address: sip.voipbuster.com
- Timeout: (empty)
- Creation Time: Apr/05/2019 11:35:54
- Buttons: OK, Cancel, Apply, Disable, Comment, Copy, Remove
- Status: enabled

Add address list
for discovery SIP
server addresses
Voipbuster

	VoipBuster SIP	sip.voipbuster.com	Apr/05/2019 11:35:54
	;; sip.voipbuster.com		
D	VoipBuster SIP	77.72.169.134	Apr/05/2019 11:35:54
	;; sip.voipbuster.com		
D	VoipBuster SIP	77.72.169.129	Apr/05/2019 11:35:54

Mark VoIP signalling connections

The image displays three overlapping Mikrotik WinBox windows for configuring Mangle Rules:

- Mangle Rule <5060>** (General tab):
 - Chain: **forward**
 - Protocol: ☐ 17 (udp)
 - Dst. Port: **5060**
- Mangle Rule <77.72.169.134:5060>** (General tab):
 - Dst. Address List: ☐ VoipBuster SIP
 - Layer7 Protocol: (empty)
 - Content: (empty)
- Mangle Rule <5060>** (Action tab):
 - Action: **mark connection**
 - ☐ Log
 - Log Prefix: (empty)
 - New Connection Mark: **SIP signalling conn**
 - ☒ Passthrough

Mark VoIP signalling packets

New Mangle Rule

General Advanced Extra Action Statistics

Chain: **forward**

Src. Address:

Dst. Address:

Protocol:

Src. Port:

Dst. Port:

Any. Port:

In. Interface:

Out. Interface:

In. Interface List:

Out. Interface List:

Packet Mark:

Connection Mark: ☐ SIP signalling conn

Routing Mark:

OK Cancel Apply Disable Comment Copy Remove Reset Counters Reset All Counters

New Mangle Rule

General Advanced Extra Action Statistics

Action: **mark packet**

☐ Log

Log Prefix:

New Packet Mark: SIP packet

☒ Passthrough

OK Cancel Apply Disable Comment Copy Remove Reset Counters Reset All Counters

Mark VoIP RTP connections

The image shows two overlapping windows from the Mikrotik WinBox interface. The background window is titled "New Mangle Rule" and has tabs for General, Advanced, Extra, Action, and Statistics. The "General" tab is active, showing the following fields: Chain: forward, Src. Address: (empty), Dst. Address: (empty), Protocol: ☐ udp, Src. Port: ☐ 5004-5020, Dst. Port: (empty), Any. Port: (empty), In. Interface: (empty), and Out. Interface: (empty). The foreground window is titled "Mangle Rule <5004-5020>" and also has tabs for General, Advanced, Extra, Action, and Statistics. The "Action" tab is active, showing: Action: mark connection, ☐ Log, Log Prefix: (empty), New Connection Mark: RTP conn, and ☒ Passthrough. On the right side of the foreground window, there is a vertical stack of buttons: OK, Cancel, Apply, Disable, Comment, Copy, Remove, Reset Counters, and Reset All Counters.

Mark VoIP RTP packets

Mangle Rule <>

General Advanced Extra Action Statistics

Chain: forward

Src. Address:

Dst. Address:

Protocol:

Src. Port:

Dst. Port:

Any. Port:

In. Interface:

Out. Interface:

In. Interface List:

Out. Interface List:

Packet Mark:

Connection Mark: ☐ RTP conn

Routing Mark:

OK

Cancel

Apply

Disable

Comment

Copy

Remove

Reset Counters

Reset All Counters

Mangle Rule <>

General Advanced Extra Action Statistics

Action: mark packet

☐ Log

Log Prefix:

New Packet Mark: RTP packet

☒ Passthrough

OK

Cancel

Apply

Disable

Comment

Copy

Remove

Reset Counters

Reset All Counters

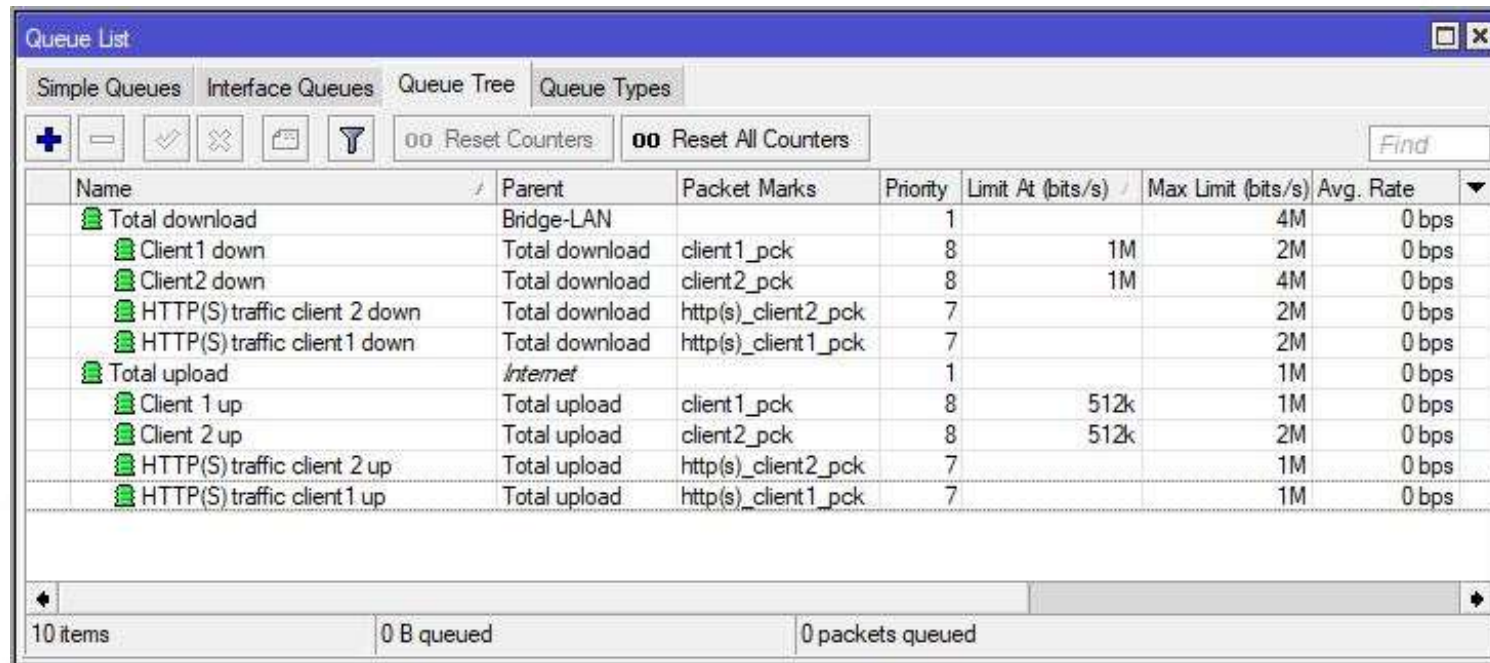
Qualify VoIP traffic with Mangle

Firewall																
Filter Rules		NAT	Mangle	Raw	Service Ports	Connections	Address Lists	Layer7 Protocols								
		00 Reset Counters		00 Reset All Counters		Find										
#	Action	Chain	Src. Address	Dst. Address	Protocol	Src. Port	Dst. Port	In. Inter...	Out. Int...	In. Inter...	Out. Int...	Src. Ad...	Dst. Address List	New Packet Mark	Bytes	Packets
::: Connection mark SIP signalling																
0	mark connection	forward			17 (udp)		5060						VoipBuster SIP		6.5 KiB	26
::: Packet mark SIP signalling																
1	mark packet	forward												SIP signalling pac...	11.5 KiB	35
::: Connection mark RTP																
2	mark connection	forward			17 (udp)	5004-5020									171.5 KiB	878
::: Packet mark RTP																
3	mark packet	forward												RTP packet	343.6 KiB	1 759

Provide the mangle rules with clear comments!

HTB implementation Queue Tree

- Queue Tree is a direct implementation of HTB



Name	Parent	Packet Marks	Priority	Limit At (bits/s)	Max Limit (bits/s)	Avg. Rate
Total download	Bridge-LAN		1		4M	0 bps
Client 1 down	Total download	client1_pck	8	1M	2M	0 bps
Client 2 down	Total download	client2_pck	8	1M	4M	0 bps
HTTP(S) traffic client 2 down	Total download	http(s)_client2_pck	7		2M	0 bps
HTTP(S) traffic client 1 down	Total download	http(s)_client1_pck	7		2M	0 bps
Total upload	Internet		1		1M	0 bps
Client 1 up	Total upload	client1_pck	8	512k	1M	0 bps
Client 2 up	Total upload	client2_pck	8	512k	2M	0 bps
HTTP(S) traffic client 2 up	Total upload	http(s)_client2_pck	7		1M	0 bps
HTTP(S) traffic client 1 up	Total upload	http(s)_client1_pck	7		1M	0 bps

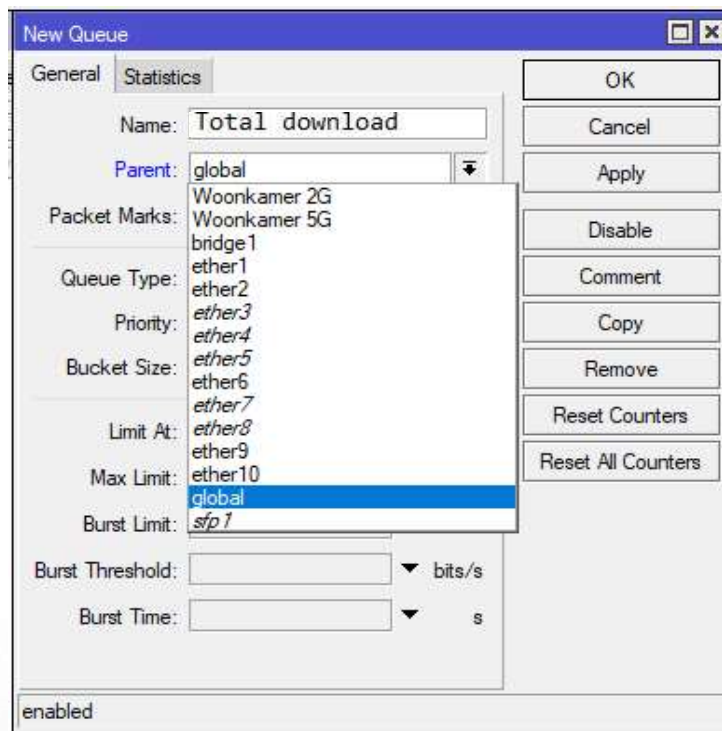
10 items 0 B queued 0 packets queued

HTB implementation Queue Tree

- Queue Tree is one directional only and can be placed in any of the available HTBs
- Queue Tree queues don't have any order – all traffic is processed simultaneously
- All child queues must have packet marks from “/ip firewall mangle” facility assigned to them
- If placed in the same HTB, Simple queue will take all the traffic away from the Queue Tree queue

HTB implementation Queue Tree

The Queue Tree can be placed in 'global' or in 'interface queue'



Each Interface HTB only receives traffic that will be leaving through a particular interface – there is no need for to separate upload and download in mangle

Queue Tree downstream

The image displays four overlapping Mikrotik Queue Tree configuration windows, illustrating a hierarchical setup for downstream traffic. Each window has a 'General' tab and a 'Statistics' tab.

- Queue <Total downstream>**
 - Name: Total downstream
 - Parent: LAN Bridge
 - Packet Marks: (empty)
 - Queue Type: default-small
 - Priority: 8
 - Bucket Size: 0.100
 - Limit At: (empty) bits/s
 - Max Limit: 50M bits/s
 - Burst Limit: (empty) bits/s
 - Burst Threshold: (empty) bits/s
 - Burst Time: (empty) s
 - enabled
- Queue <SIP signalling down>**
 - Name: SIP signalling down
 - Parent: Total downstream
 - Packet Marks: SIP signalling packet
 - Queue Type: default-small
 - Priority: 8
 - Bucket Size: 0.100
 - Limit At: 10k bits/s
 - Max Limit: 10k bits/s
 - Burst Limit: (empty) bits/s
 - Burst Threshold: (empty) bits/s
 - Burst Time: (empty) s
 - enabled
- Queue <RTP down>**
 - Name: RTP down
 - Parent: Total downstream
 - Packet Marks: RTP packet
 - Queue Type: default-small
 - Priority: 7
 - Bucket Size: 0.100
 - Limit At: 400k bits/s
 - Max Limit: 400k bits/s
 - Burst Limit: (empty) bits/s
 - Burst Threshold: (empty) bits/s
 - Burst Time: (empty) s
 - enabled
- Queue <Other traffic down>**
 - Name: Other traffic down
 - Parent: Total downstream
 - Packet Marks: no-mark
 - Queue Type: default-small
 - Priority: 8
 - Bucket Size: 0.100
 - Limit At: (empty) bits/s
 - Max Limit: (empty) bits/s
 - Burst Limit: (empty) bits/s
 - Burst Threshold: (empty) bits/s
 - Burst Time: (empty) s
 - enabled

Buttons on the right of the 'Other traffic down' window: OK, Cancel, Apply, Disable, Comment, Copy, Remove, Reset Counters, Reset All Counters.

Queue Tree upstream

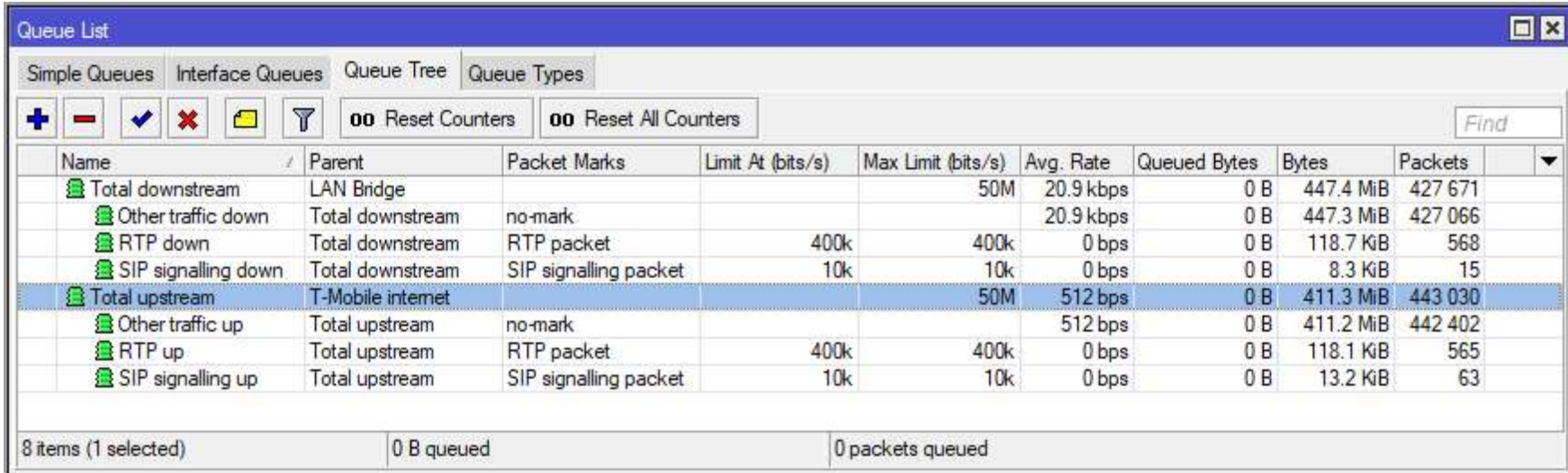
The image displays four overlapping Mikrotik Queue Manager configuration windows, illustrating a queue tree structure for upstream traffic. Each window has a 'General' tab and a 'Statistics' tab.

- Queue <Total upstream>**
 - Name: Total upstream
 - Parent: T-Mobile internet
 - Packet Marks: (empty)
 - Queue Type: default-small
 - Priority: 8
 - Bucket Size: 0.100
 - Limit At: (empty) bits/s
 - Max Limit: 50M bits/s
 - Burst Limit: (empty) bits/s
 - Burst Threshold: (empty) bits/s
 - Burst Time: (empty) s
 - enabled
- Queue <SIP signalling up>**
 - Name: SIP signalling up
 - Parent: Total upstream
 - Packet Marks: SIP signalling packet
 - Queue Type: default-small
 - Priority: 8
 - Bucket Size: 0.100
 - Limit At: 10k bits/s
 - Max Limit: 10k bits/s
 - Burst Limit: (empty) bits/s
 - Burst Threshold: (empty) bits/s
 - Burst Time: (empty) s
 - enabled
- Queue <RTP up>**
 - Name: RTP up
 - Parent: Total upstream
 - Packet Marks: RTP packet
 - Queue Type: default-small
 - Priority: 7
 - Bucket Size: 0.100
 - Limit At: 400k bits/s
 - Max Limit: 400k bits/s
 - Burst Limit: (empty) bits/s
 - Burst Threshold: (empty) bits/s
 - Burst Time: (empty) s
 - enabled
- Queue <Other traffic up>**
 - Name: Other traffic up
 - Parent: Total upstream
 - Packet Marks: no-mark
 - Queue Type: default-small
 - Priority: 8
 - Bucket Size: 0.100
 - Limit At: (empty) bits/s
 - Max Limit: (empty) bits/s
 - Burst Limit: (empty) bits/s
 - Burst Threshold: (empty) bits/s
 - Burst Time: (empty) s
 - enabled

Buttons on the right of the 'Other traffic up' window: OK, Cancel, Apply, Disable, Comment, Copy, Remove, Reset Counters, Reset All Counters.

HTB implementation Queue Tree

Queue Tree structure



The screenshot shows the 'Queue List' window in WinBox. The 'Queue Tree' tab is selected. The table displays the hierarchy of queues, including 'Total downstream' and 'Total upstream' with their respective children. The 'Total upstream' row is highlighted in blue, indicating it is selected. The status bar at the bottom shows '8 items (1 selected)', '0 B queued', and '0 packets queued'.

Name	Parent	Packet Marks	Limit At (bits/s)	Max Limit (bits/s)	Avg. Rate	Queued Bytes	Bytes	Packets
Total downstream	LAN Bridge			50M	20.9 kbps	0 B	447.4 MiB	427 671
Other traffic down	Total downstream	no-mark			20.9 kbps	0 B	447.3 MiB	427 066
RTP down	Total downstream	RTP packet	400k	400k	0 bps	0 B	118.7 KiB	568
SIP signalling down	Total downstream	SIP signalling packet	10k	10k	0 bps	0 B	8.3 KiB	15
Total upstream	T-Mobile internet			50M	512 bps	0 B	411.3 MiB	443 030
Other traffic up	Total upstream	no-mark			512 bps	0 B	411.2 MiB	442 402
RTP up	Total upstream	RTP packet	400k	400k	0 bps	0 B	118.1 KiB	565
SIP signalling up	Total upstream	SIP signalling packet	10k	10k	0 bps	0 B	13.2 KiB	63

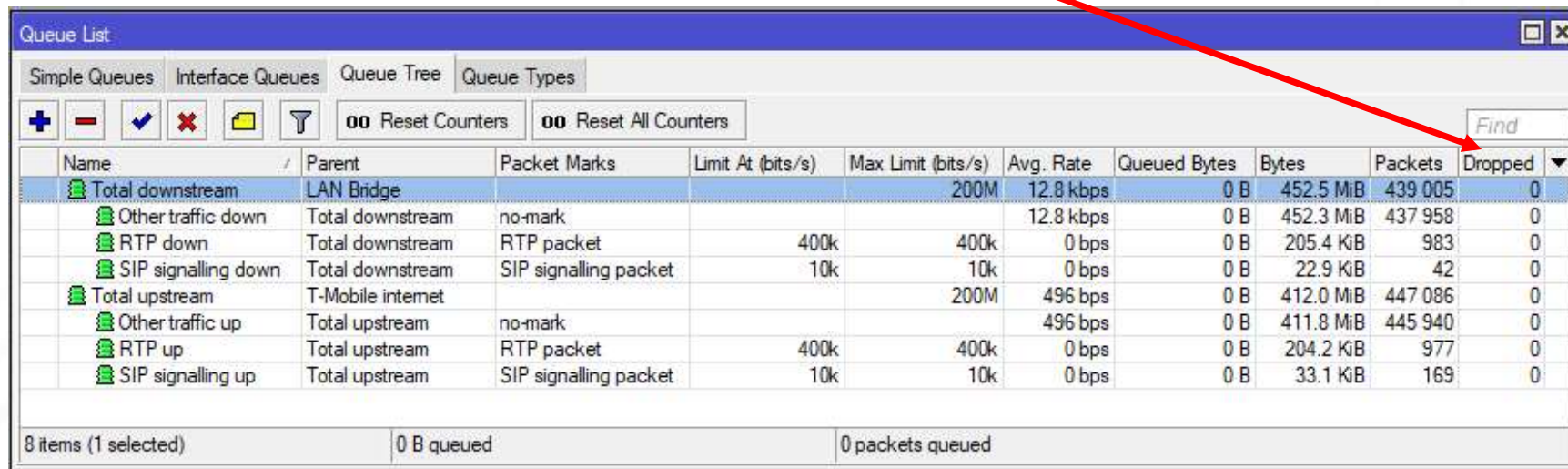
Queue colors in Winbox:

- 0% - 50% available traffic used - green
- 51% - 75% available traffic used - yellow
- 76% - 100% available traffic used - red

Check VoIP traffic

- Checking for dropped packets in Queue Tree

Show Cols 'dropped'



Queue List

Simple Queues Interface Queues Queue Tree Queue Types

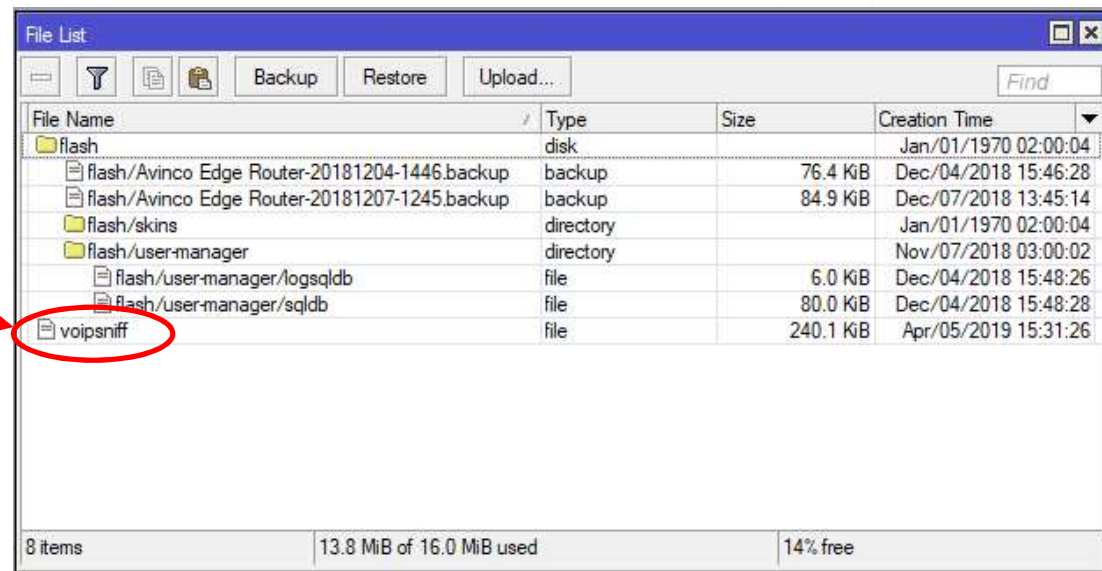
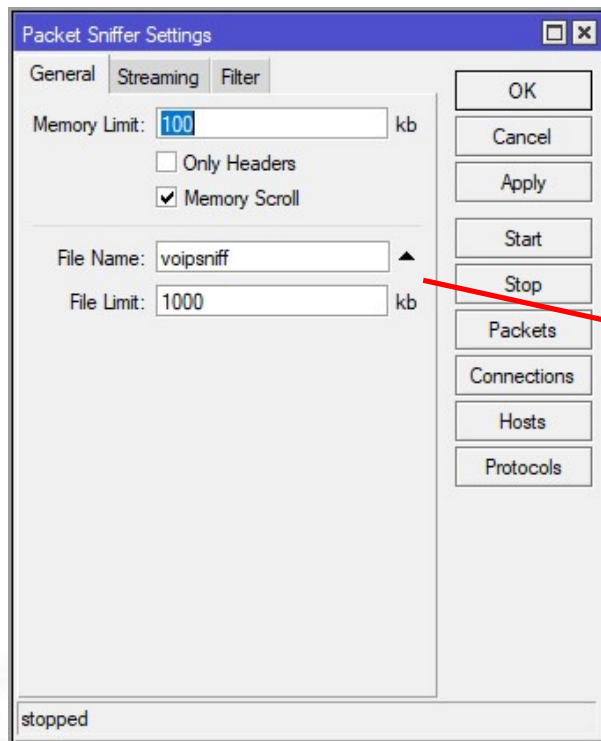
+ - ✓ ✗ 📁 🔍 00 Reset Counters 00 Reset All Counters Find

Name	Parent	Packet Marks	Limit At (bits/s)	Max Limit (bits/s)	Avg. Rate	Queued Bytes	Bytes	Packets	Dropped
Total downstream	LAN Bridge			200M	12.8 kbps	0 B	452.5 MiB	439 005	0
Other traffic down	Total downstream	no-mark			12.8 kbps	0 B	452.3 MiB	437 958	0
RTP down	Total downstream	RTP packet	400k	400k	0 bps	0 B	205.4 KiB	983	0
SIP signalling down	Total downstream	SIP signalling packet	10k	10k	0 bps	0 B	22.9 KiB	42	0
Total upstream	T-Mobile internet			200M	496 bps	0 B	412.0 MiB	447 086	0
Other traffic up	Total upstream	no-mark			496 bps	0 B	411.8 MiB	445 940	0
RTP up	Total upstream	RTP packet	400k	400k	0 bps	0 B	204.2 KiB	977	0
SIP signalling up	Total upstream	SIP signalling packet	10k	10k	0 bps	0 B	33.1 KiB	169	0

8 items (1 selected) 0 B queued 0 packets queued

Check VoIP traffic by sniffing

- Activate Packet Sniffer for VoIP traffic



The 'File List' window displays a table of files and directories. The 'voipsniff' file is circled in red. The status bar at the bottom shows 8 items, 13.8 MiB of 16.0 MiB used, and 14% free space.

File Name	Type	Size	Creation Time
flash	disk		Jan/01/1970 02:00:04
flash/Avinco Edge Router-20181204-1446.backup	backup	76.4 KiB	Dec/04/2018 15:46:28
flash/Avinco Edge Router-20181207-1245.backup	backup	84.9 KiB	Dec/07/2018 13:45:14
flash/skins	directory		Jan/01/1970 02:00:04
flash/user-manager	directory		Nov/07/2018 03:00:02
flash/user-manager/logsqldb	file	6.0 KiB	Dec/04/2018 15:48:26
flash/user-manager/sqlqldb	file	80.0 KiB	Dec/04/2018 15:48:28
voipsniff	file	240.1 KiB	Apr/05/2019 15:31:26

Check VoIP traffic by sniffing

The image shows two windows from a network analysis tool. The top window, titled 'voipsniff', has a menu bar with 'File', 'Edit', 'View', 'Go', 'Capture', 'Analyze', 'Statistics', 'Telephony', 'Wireless', 'Tools', and 'Help'. The 'Telephony' menu is open, showing options like 'VoIP Calls', 'ANSI', 'GSM', 'IAX2 Stream Analysis', 'ISUP Messages', 'LTE', 'MTP3', 'Osmux', 'RTP', 'RTSP', 'SCTP', 'SMPP Operations', 'UCP Messages', 'H.225', 'SIP Flows', and 'SIP Statistics'. The 'RTP' option is selected, and a sub-menu is open showing 'RTP Streams' and 'Stream Analysis'. The main pane of 'voipsniff' displays a list of RTP streams with columns 'No.', 'Time', and 'Source'. The bottom window, titled 'Wireshark · RTP Streams · voipsniff', shows a table of RTP streams. The table has columns: 'Source Address', 'Source Port', 'Destination Address', 'Destination Port', 'SSRC', 'Payload', 'Packets', 'Lost', 'Max Delta (ms)', 'Max Jitter', 'Mean Jitter', and 'Status'. The table contains two rows of data. Below the table, it says '2 streams. Right-click for more options.' and there are buttons for 'Close', 'Find Reverse', 'Prepare Filter', 'Export...', 'Copy', 'Analyze', and 'Help'.

No.	Time	Source
119	6.576084	62.41.83.78
120	6.580494	192.168.1.113
121	6.596436	62.41.83.78
122	6.600502	192.168.1.113
123	6.616406	62.41.83.78
124	6.620497	192.168.1.113
125	6.636449	62.41.83.78
126	6.640494	192.168.1.113
127	6.656436	62.41.83.78
128	6.660497	192.168.1.113
129	6.676267	62.41.83.78
130	6.680492	192.168.1.113

Source Address	Source Port	Destination Address	Destination Port	SSRC	Payload	Packets	Lost	Max Delta (ms)	Max Jitter	Mean Jitter	Status
62.41.83.78	32940	192.168.1.113	5004	0x0	g711A	415	0 (0.0%)	22.266	0.534	0.267	
192.168.1.113	5004	62.41.83.78	32940	0xcbbec9fe	g711A	412	0 (0.0%)	20.014	0.005	0.003	

Check VoIP traffic by sniffing

Wireshark · RTP Stream Analysis · voipsniff

62.41.83.78:32940 ↔ 192.168.1.113:5004

Forward

SSRC 0x00000000
Max Delta 22.27 ms @ 101
Max Jitter 0.53 ms
Mean Jitter 0.27 ms
Max Skew -2.60 ms
RTP Packets 415
Expected 415
Lost 0 (0.00 %)
Seq Errs 0
Start at 6.336141 s @ 95
Duration 8.28 s
Clock Drift -222 ms
Freq Drift 7786 Hz (-2.68 %)

Reverse

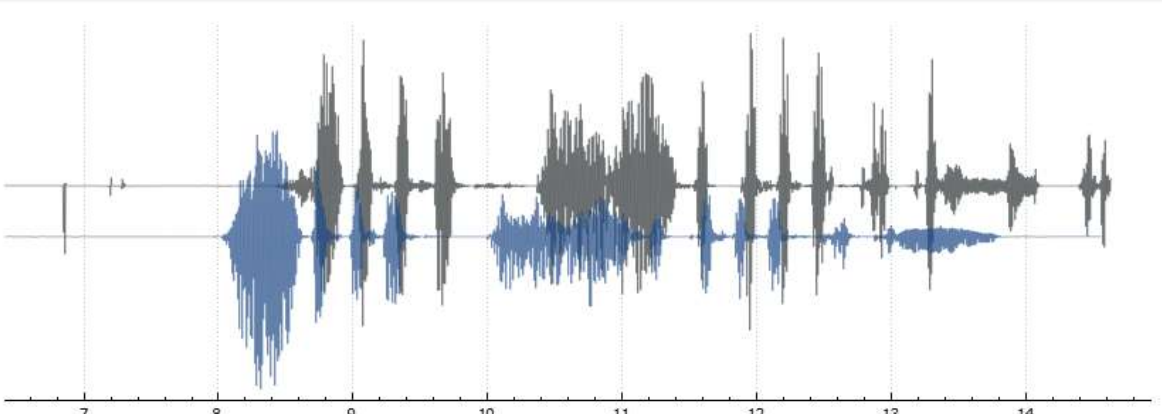
SSRC 0xcbbec9fe
Max Delta 20.01 ms @ 227
Max Jitter 0.00 ms
Mean Jitter 0.00 ms
Max Skew 0.06 ms
RTP Packets 412
Expected 412
Lost 0 (0.00 %)
Seq Errs 0
Start at 6.320512 s @ 94
Duration 8.22 s
Clock Drift -223 ms
Freq Drift 7783 Hz (-2.72 %)

Forward to reverse start diff -0.015629 s @ -1

2 streams found. G: Go to packet, N: Next problem packet

Packet	Sequence	Delta (ms)	Jitter (ms)	Skew	Bandwidth	Marker	Status
95	5	0.00	0.00	0.00	1.60		✓
97	6	20.41	0.03	-0.41	3.20		✓
99	7	19.66	0.05	-0.07	4.80		✓
101	8	22.27	0.18	-2.34	6.40		✓
103	9	17.63	0.32	0.04	8.00		✓
105	10	20.32	0.32	-0.28			
107	11	19.86	0.31	-0.14			
109	12	20.24	0.31	-0.38			
111	13	20.54	0.32	-0.93			
113	14	21.67	0.40	-2.60			
115	15	17.53	0.53	-0.12			
117	16	20.40	0.53	-0.53			
119	17	19.42	0.53	0.06			
121	18	20.35	0.52	-0.30			
123	19	19.97	0.49	-0.27			
125	20	20.04	0.46	-0.31			
127	21	19.99	0.43	-0.30			
129	22	19.83	0.42	-0.13			
131	23	20.10	0.40	-0.22			
133	24	19.97	0.37	-0.19			
135	25	19.82	0.36	-0.01			
137	26	20.11	0.34	-0.12			
139	27	19.96	0.33	-0.08			
141	28	20.18	0.32	-0.26			
143	29	20.10	0.30	-0.36			
145	30	19.96	0.29	-0.32			
147	31	19.95	0.27	-0.27			
149	32	20.83	0.31	-1.10			

Wireshark · RTP Player



Source Address	Source Port	Destination Address	Destination Port	SSRC	Setup Frame	Packets	Time Span (s)	Sample Rate (Hz)	Payloads
62.41.83.78	32940	192.168.1.113	5004	0x00000000	62	415	6.34 - 14.6 (8.28)	8000	g711A
192.168.1.113	5004	62.41.83.78	32940	0xcbbec9fe	62	412	6.32 - 14.5 (8.22)	8000	g711A

Output Device: Default Output Device

Jitter Buffer: 50 Playback Timing: Jitter Buffer ☐ Time of Day

Close Help

Thank you for your attention!