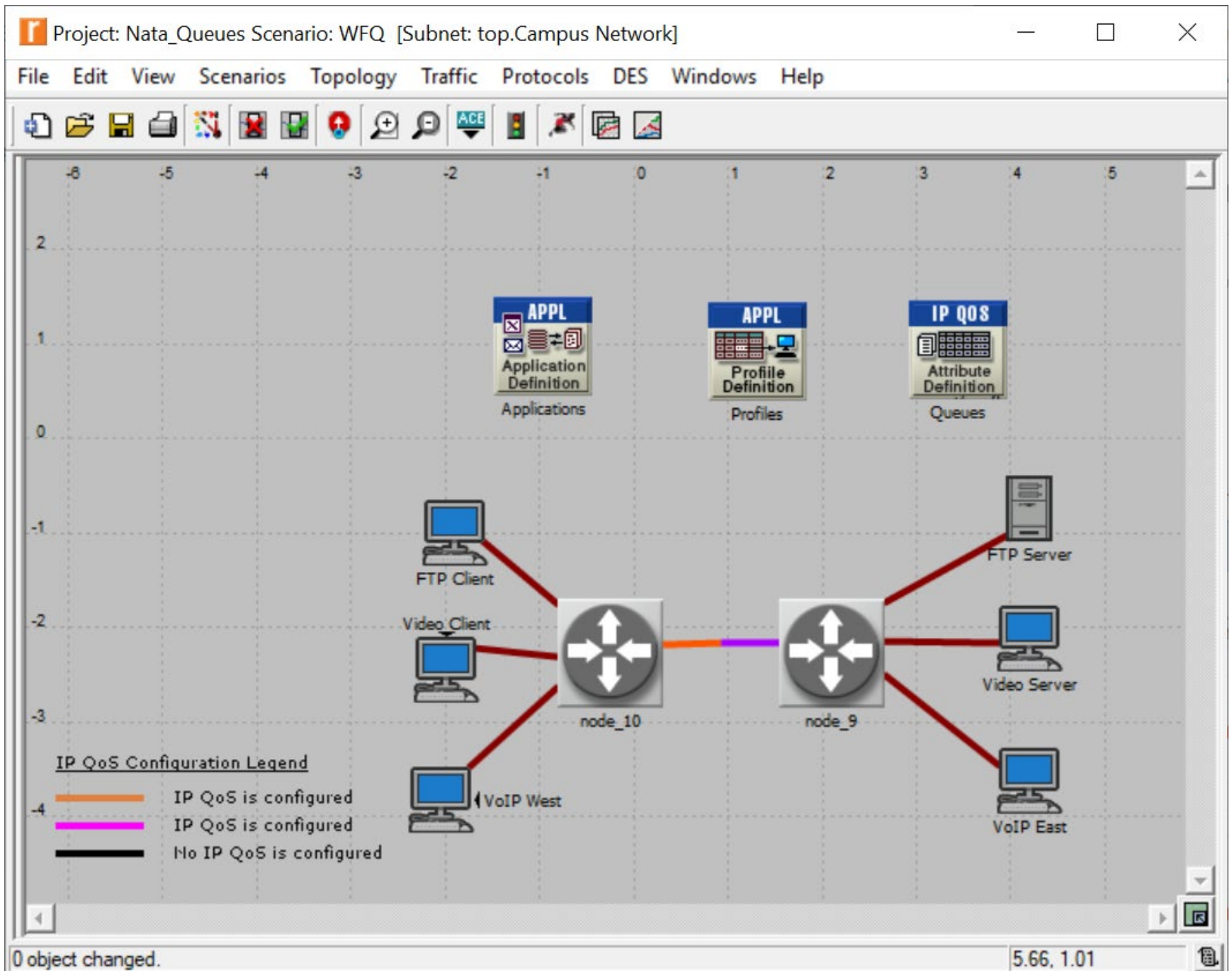


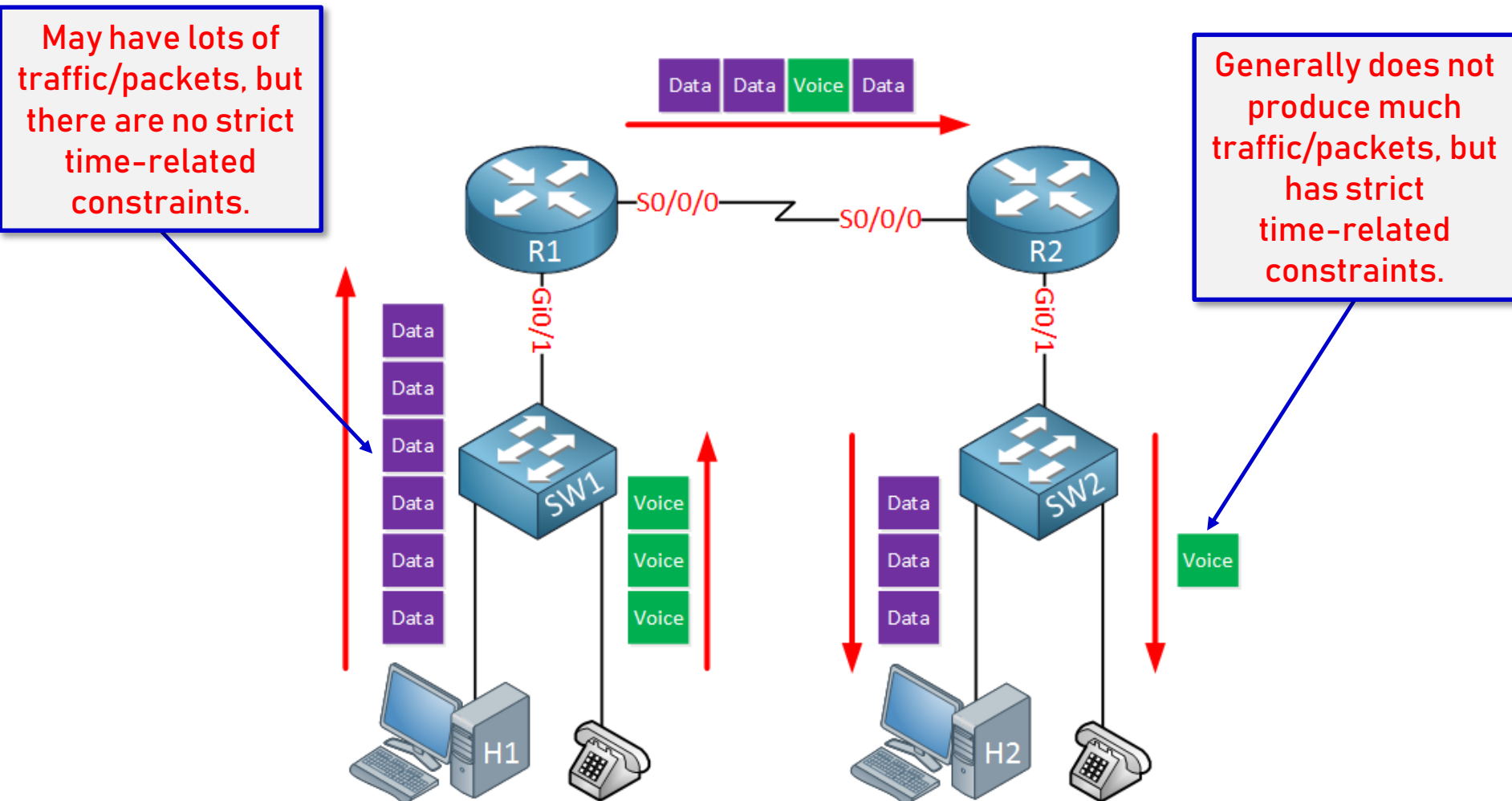
# **Network Layer:**

## **Queueing Disciplines**

### **(LAB 1 Related Material)**

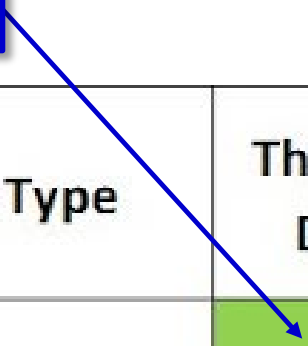


# characteristics & requirements of **data transfer $\neq$ streamed video $\neq$ VoIP**



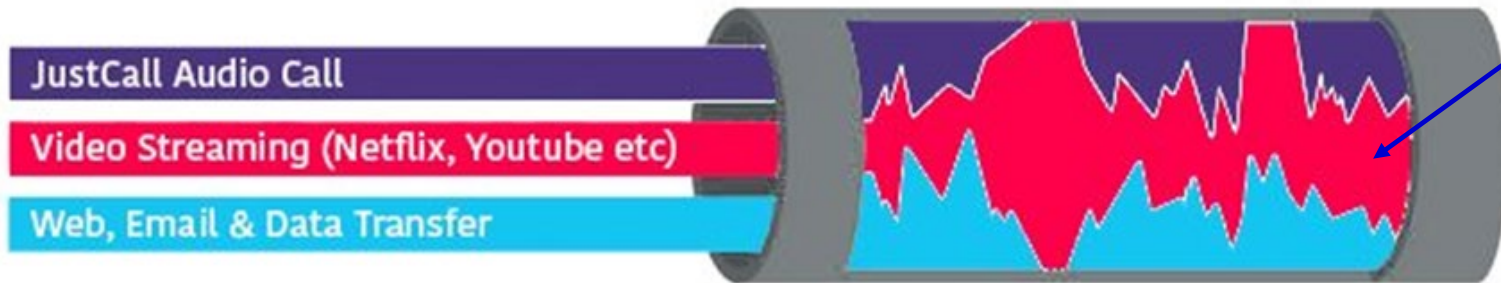
# Quality of Service (QoS) requirements of different applications.

Should throughput be high or low??



Application Type	Throughput Demand	Latency Tolerance	Jitter Tolerance	Loss Tolerance
Email	Low	High	High	High
Web browsing	Low	High	High	High
File transfer (FTP)	Low - High	High	High	High
Chat (IM)	Low	Medium	Medium	Medium
Video streaming (e.g. surveillance)	Medium - High	Medium	Medium	Medium
Video on demand (e.g. YouTube / NetFlix)	High	Medium	Medium	Low
Voice over IP / WiFi	Low	Low	Low	Low
Videoconferencing (e.g. Skype, FaceTime)	Medium - High	Low	Low	Low

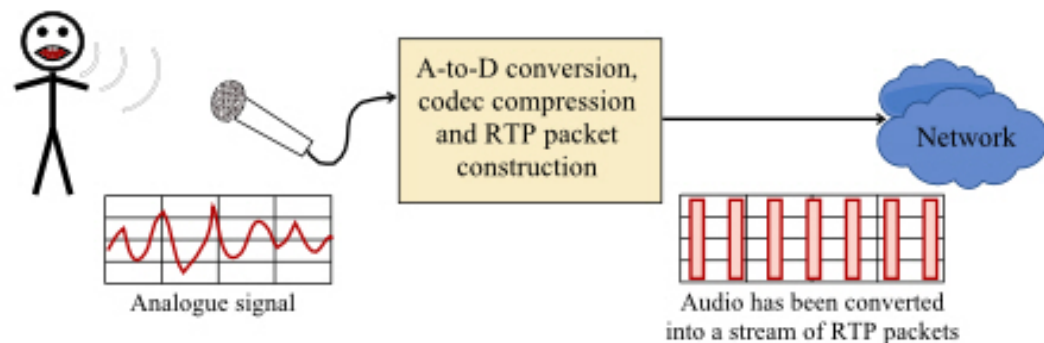
Sharing of bandwidth/router capacity when there are NO special QoS provisions in place.



More dominant traffic flows get allocated more bandwidth 'without any regard' for other traffic flows.

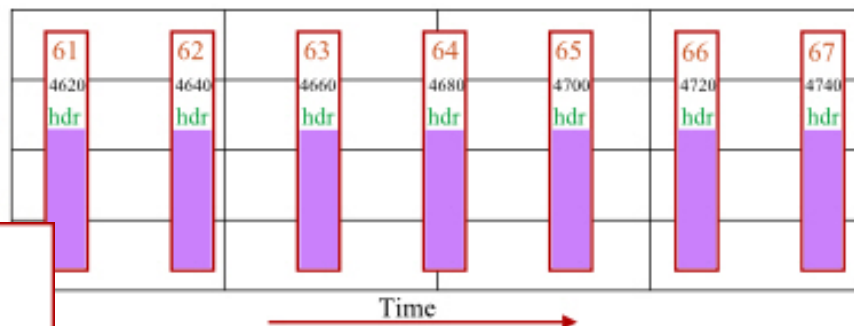
Sharing of bandwidth/router capacity when there are special QoS provisions in place.



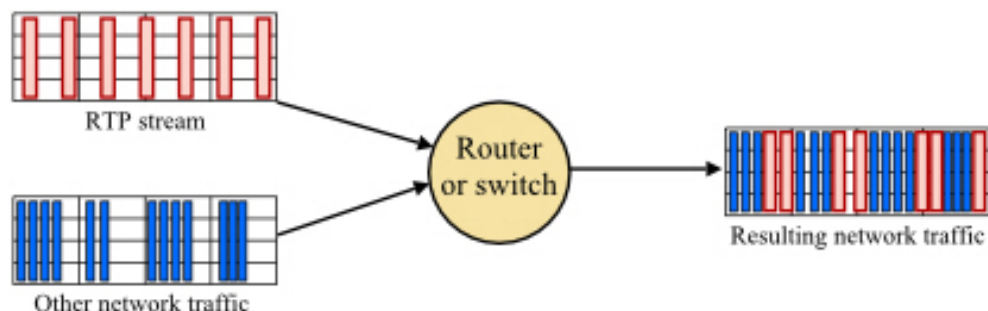


RTP packets generated at the source device

Sequence No  
Time stamp  
Other headers  
Payload (data)

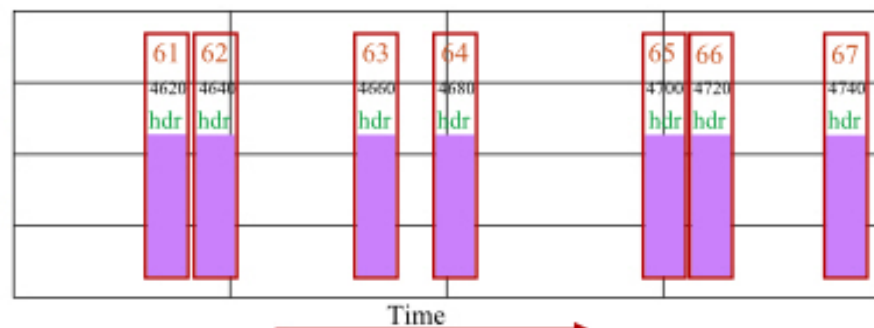


A jitter-free stream of RTP packets



RTP packets have to jostle for position with other network traffic

Sequence No  
Time stamp  
Other headers  
Payload (data)



This sample of the RTP stream has jitter (but has no dropped packets)

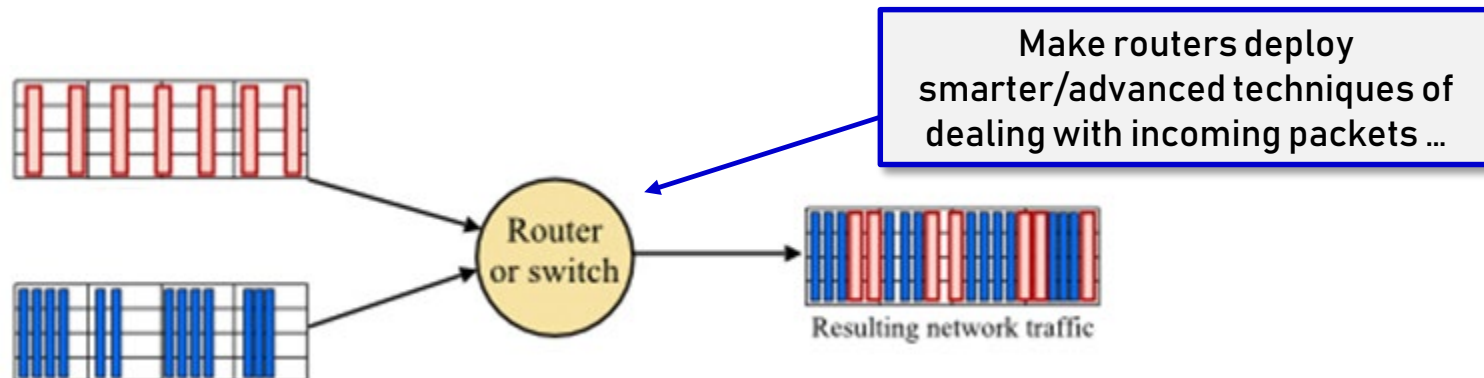
# Quality of Service

## Quality of Service (QoS)

- capability of a network to guarantee a certain level of performance to an application, user or data flow
- parameters used to define / measure QoS
  - (1) **RELIABILITY** - measures packet loss
  - (2) **DELAY** - measures delay in individual packet transmiss.
  - (3) **JITTER** - measures **variation** in packet delay
  - (4) **BANDWIDTH** - measures overall demand on [bps]

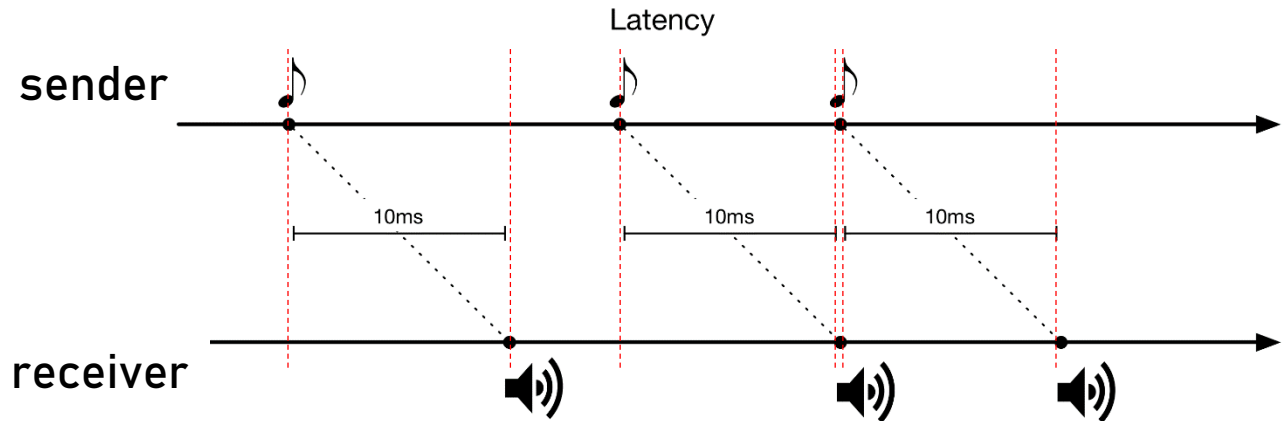
**Different applications required different QoS.**

**What could we do to try to provide each application with the right QoS??**

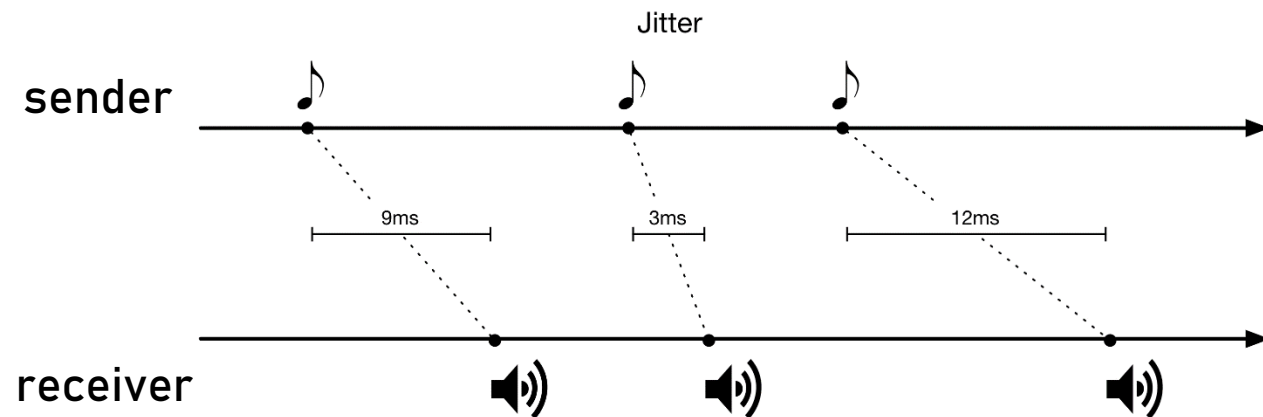


## Example [ delay vs. jitter ]

delay  $\neq 0$   
jitter = 0

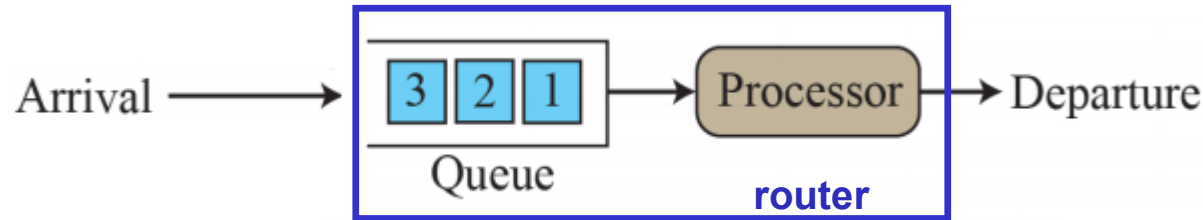


delay  $\neq 0$   
jitter  $\neq 0$





**Example** [ causes of packet delay & jitter: processing + queueing delay ]



**Required processing time**

Packet 1: two time units

Packet 2: three time units

Packet 3: two time units

packet 1:

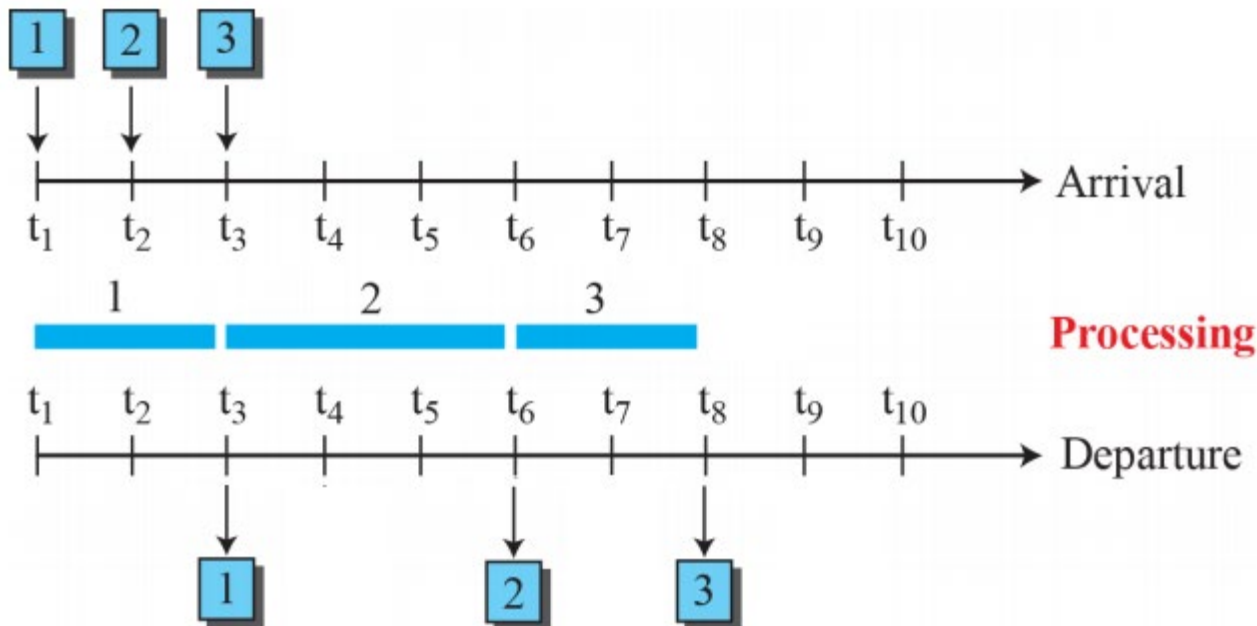
$$d = d_{\text{proc.}} = 2 \text{ [tu]}$$

packet 2:

$$\begin{aligned} d &= d_{\text{que.}} + d_{\text{proc.}} = \\ &= 1 + 3 = 4 \text{ [tu]} \end{aligned}$$

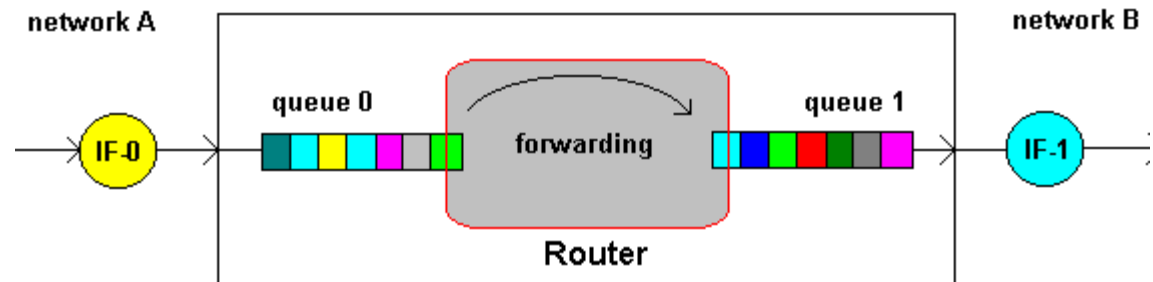
packet 3:

$$\begin{aligned} d &= d_{\text{que.}} + d_{\text{proc.}} = \\ &= 3 + 2 = 5 \text{ [tu]} \end{aligned}$$

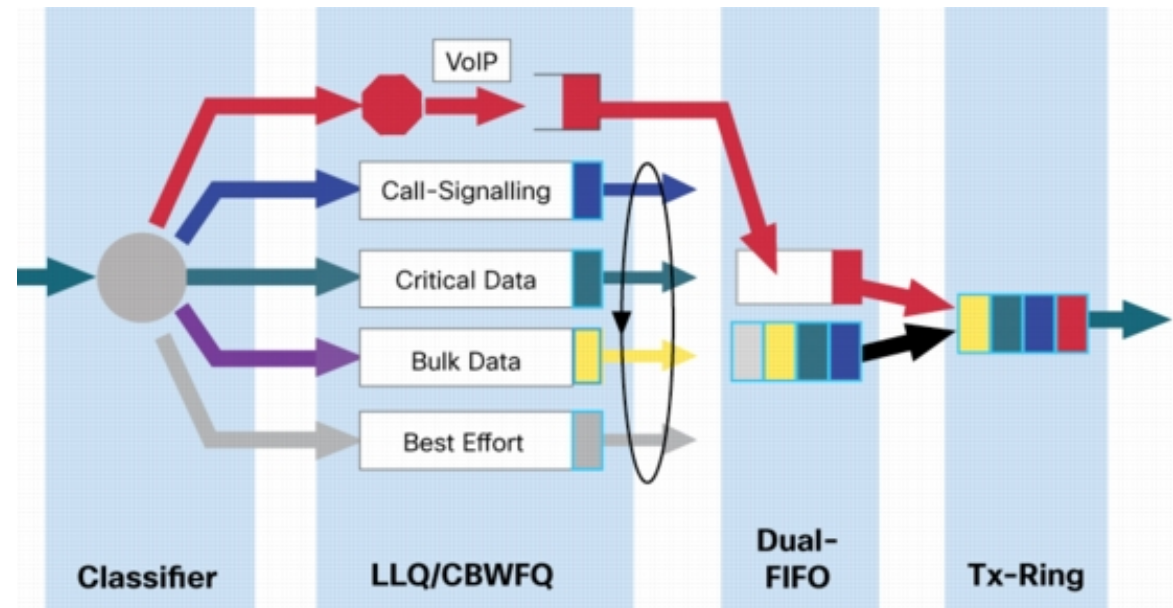


## Example [ packet handling at intermediate routers ]

**simple packet handing/scheduling**  
(all packets 'treated' equally, all applications obtain the same QoS)



**prioritized packet handing/scheduling**  
(packets are separated and 'treated' based on their application-type  
⇒ different QoS can be obtained)



# Packet Scheduling

## Packet Scheduling to Improve QoS

Key element in ensuring priority of multimedia traffic, once packets are properly marked.

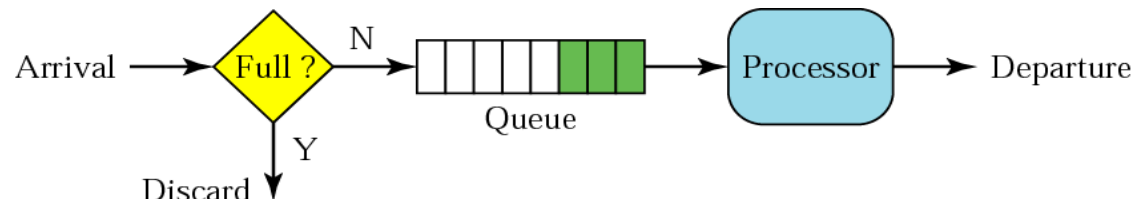
– manner in which queued packets are selected for transmission – **has important role in providing QoS**

• examples:

- (1) **FIFO**
- (2) **Priority Queueing**
- (3) **Weighted Fair Queueing**

**FIFO Scheduling** – packets are selected for transmission in the same order in which they arrive in the queue

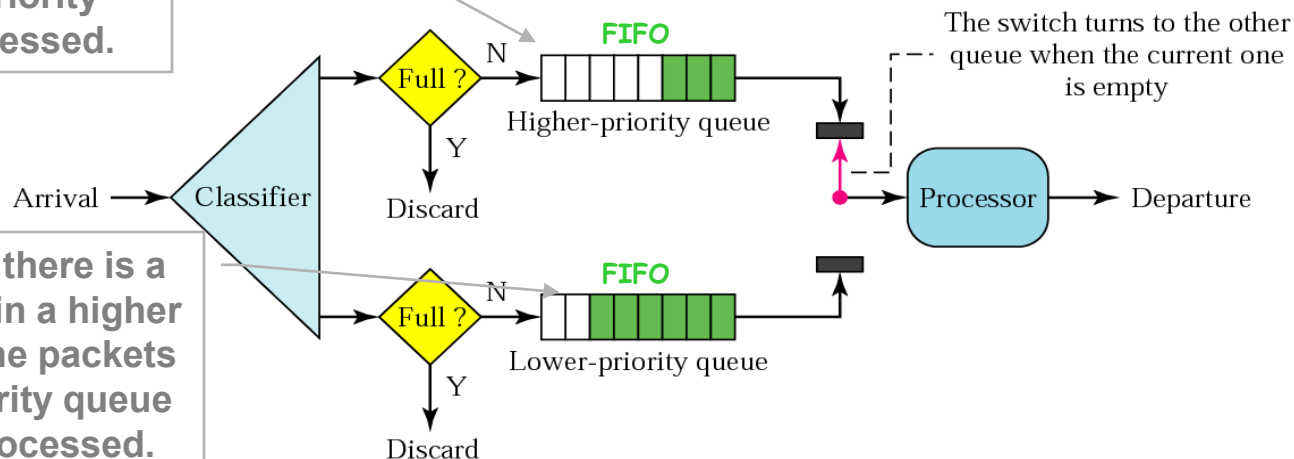
- delays and losses of packets depend on packet interarrival times and packet lengths – no **QoS guaranteed!**
- **packet-discarding policy** – used to determine which packets should be dropped if there is not sufficient buffering space
  - (a) **tail drop**: drop last arriving packets
  - (b) **priority**: drop lower-priority packets
  - (c) **RED** – random early detection



## Priority Queueing – packets are classified into priority classes – a separate buffer is maintained for each priority class

- assigned priority class may depend on: ToS field, source or destination IP address, destination port number
- **next packet for transmission is selected from the head of the highest priority queue that is not empty** – each individual queue is served in a FIFO manner
- advantages: better QoS to high(est) priority traffic than FIFO
- disadvantages:
  - (a) no QoS guarantees to lower priority traffic (**starvation**)
  - (b) cannot discriminate between users of the same priority – certain users can hog bandwidth with excessive traffic

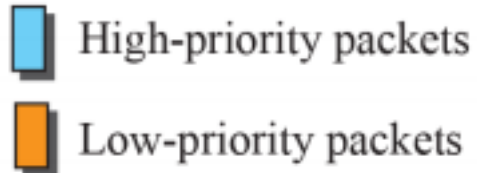
The processor moves to the lower-priority queue only after all packets from higher-priority queue are processed.



**STARVATION:** if there is a continuous flow in a higher priority queue, the packets in the lower priority queue will never be processed.

## Example [ priority queueing ]

### Legend



### required processing time

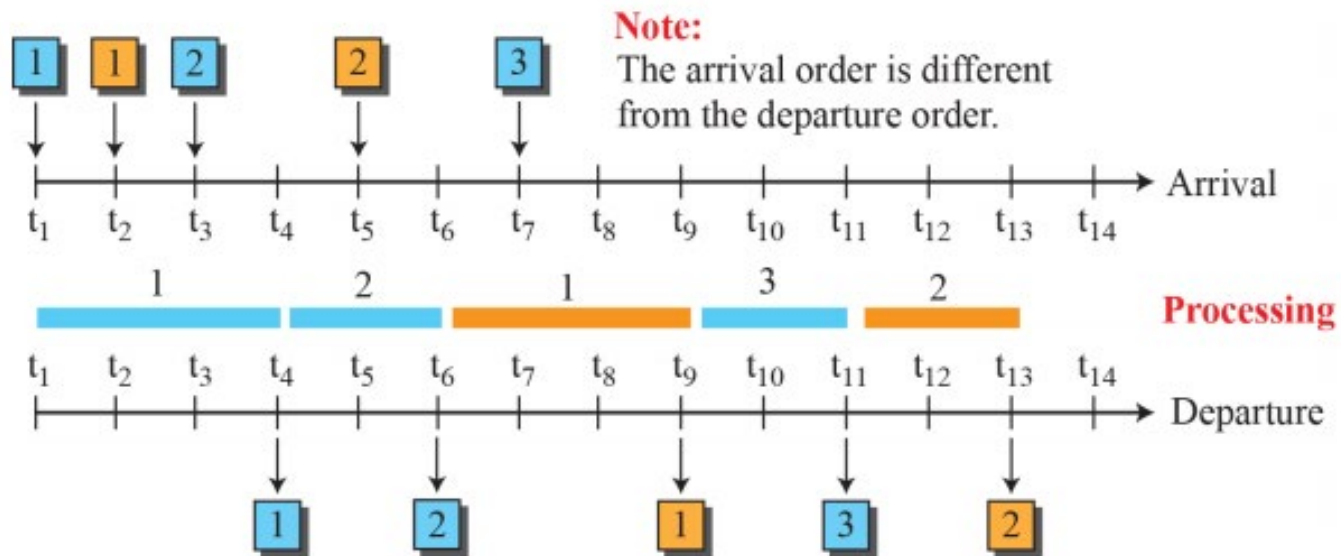
high-priority P1: 3 time units

high-priority P2: 2 time units

high-priority P3: 2 time units

low-priority P1: 3 time units

low-priority P2: 2 time units

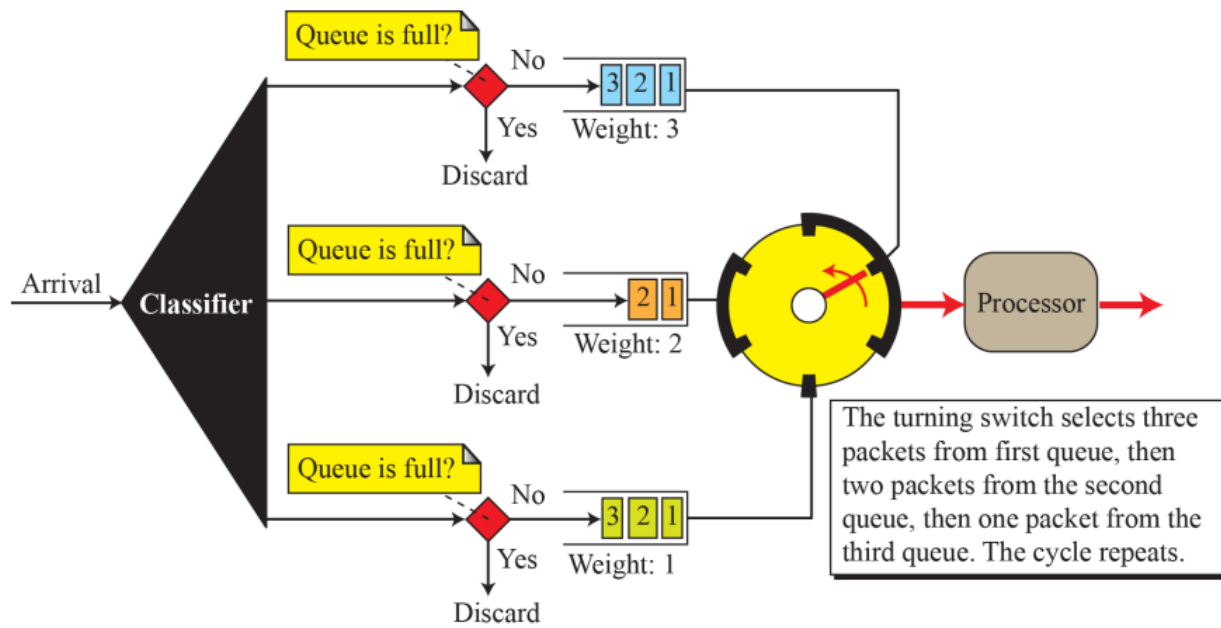


## Weighted Fair Queueing

- packets are classified into priority classes as in priority queueing, however WFQ scheduler serves classes in a circular manner (but not plain round-robin!)
- class(i) receives a weight that determines its relative share of bandwidth ( $w_i$ ): **in any interval of time class(i) is guaranteed to receive at least a fraction of service/bandwidth equal to**

$$R(i) = R \cdot \frac{w_i}{\sum w_j}$$

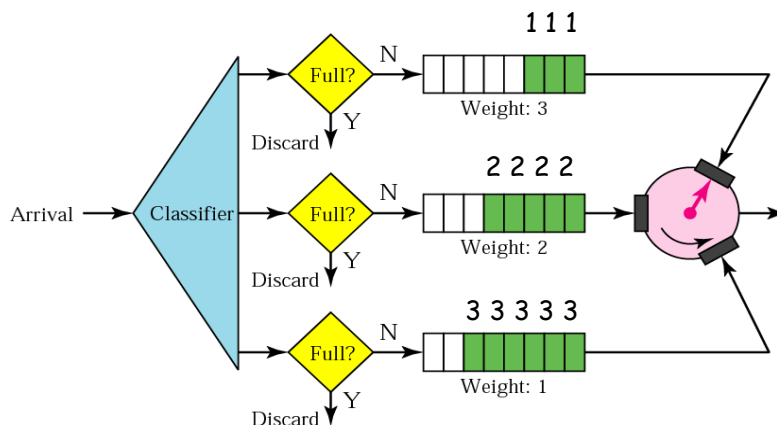
- **advantages**: QoS guarantees to all priority classes



## Example [ WFQ ]

Suppose that the WFQ scheduling policy is applied to a buffer that supports 3 classes, and suppose the weights are: 0.5, 0.25, 0.25 for the three classes.

- Suppose that each class has a large number of packets in the buffer. In what sequence might the three classes be served in order to achieve the WFQ weights?
- Suppose that classes 1 and 2 have a large number of packets in the buffer, and there are no class 3 packets in the buffer. In what sequence might the three classes be served to achieve the WFQ weights?



## Solution:

- Possibility 1: 1 1 2 3 1 1 2 3 1 1 2 3 ...

Possibility 2: 1 2 1 3 1 2 1 3 1 2 1 3 ...

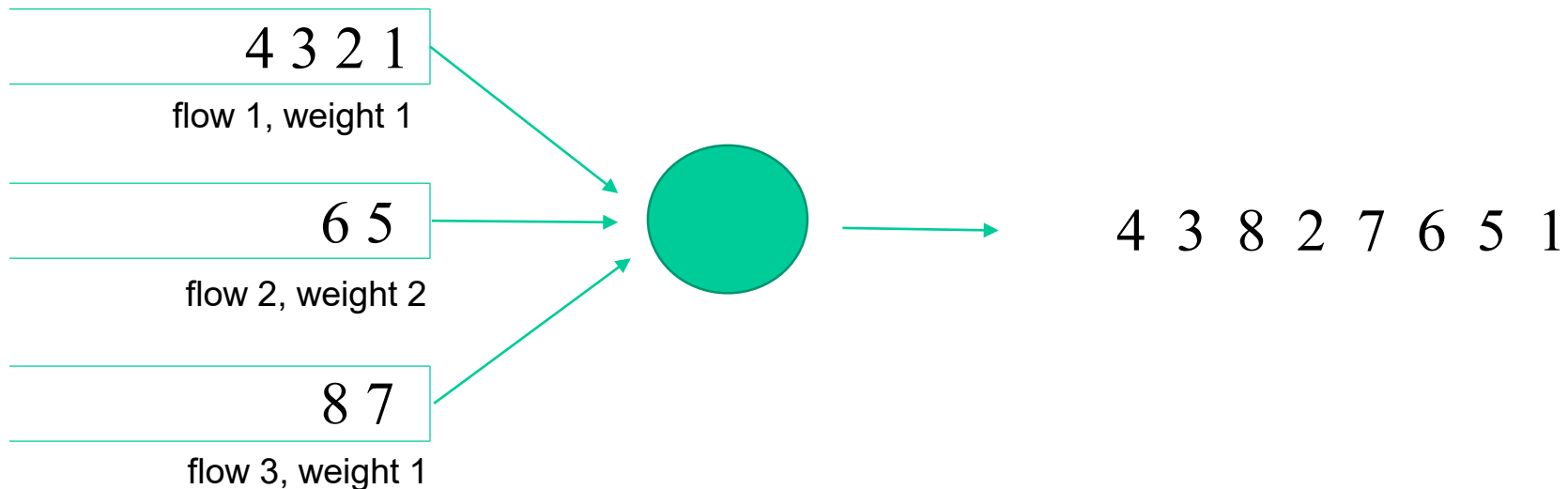
- The new value of weights for class 1 and class 2 has to satisfy:

$$\left. \begin{array}{l} w1_{\text{new}} = 2 * w2_{\text{new}} \\ w1_{\text{new}} + w2_{\text{new}} = 1 \end{array} \right\} \begin{array}{l} w1_{\text{new}} = 2/3 \\ w2_{\text{new}} = 1/3 \end{array}$$

Possible solution: 1 1 2 1 1 2 1 1 2

## Example [WFQ]

Suppose a router has three input flows and one output. It receives the packets listed in the table below all at about the same time in the order listed during a period in which the output port is busy but all queues are otherwise empty. Give the order in which packets are transmitted, assuming: weighted fair queuing with flow #2 having weight 2, and flows #1,3 with weight 1.

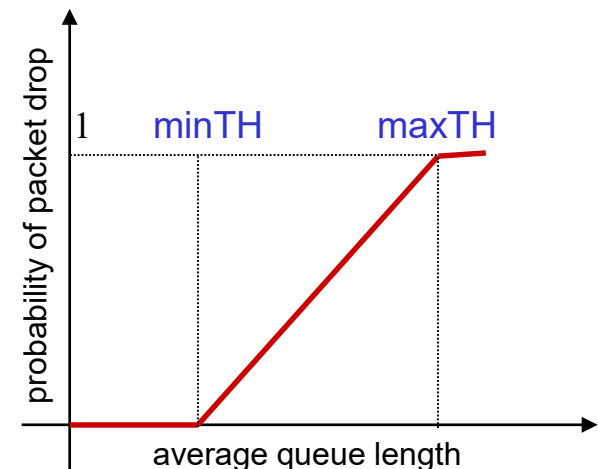
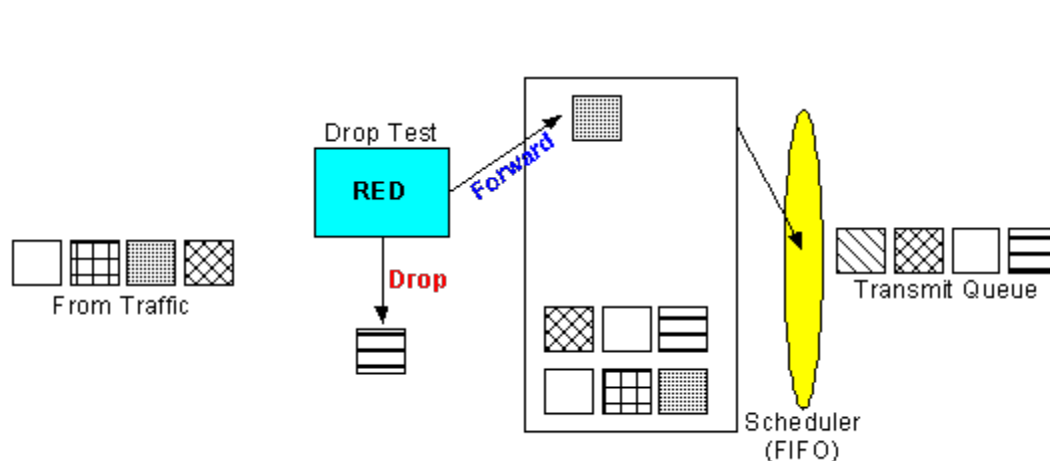




## Random Early Detection (RED) Algorithm

– **proactive packet discard** – router discards one or more **incoming packets** before output buffer gets full  $\Rightarrow$  improved network performance

- **motivation**: TCP connections enter slow start around the same time, and come out of slow start around the same time, causing another cycle of congestion (**‘global synchronization’** phenomenon)
- **solution**: anticipate the onset of congestion and tell one TCP connection at a time to slow down – **“apply brakes gradually to gently reduce traffic”**
- **RED idea**: packets are randomly discarded with increased probability as the queue size grows



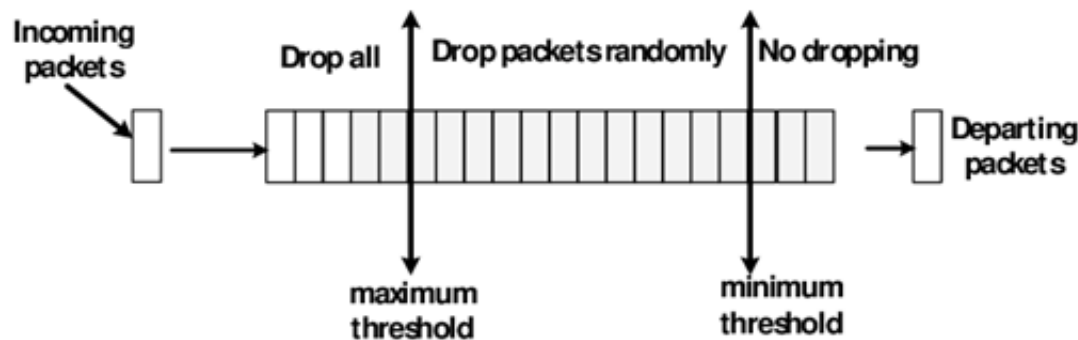
for each packet arrival  
calculate the average queue size **avg**

Exponentially  
weighted average of  
previous queue  
lengths.

if **avg** < **minTH**  
queue packet

else if **minTH** ≤ **avg** < **maxTH**  
calculate probability  $P_a$   
with probability  $P_a$  discard packet  
with probability  $(1-P_a)$  queue packet

else if **avg** ≥ **maxTH**  
discard packet



The method of “**early drop**” is used to notify the source to reduce its transmission rate before the buffer becomes full.

## Queuing and Loss

This animation illustrates queuing delay and packet loss.

Three different senders - indicated by colors - send packets. The packets arrive and queue for service.

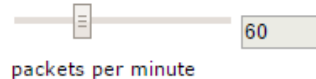
If the emission rate is higher than the transmission rate (both are slotted for better visualisation) a queue overflow will happen and according to the chosen method different packets

### configuration

#### method

- ☒ drop tail
- ☐ drop head

#### emission rate



#### transmission rate



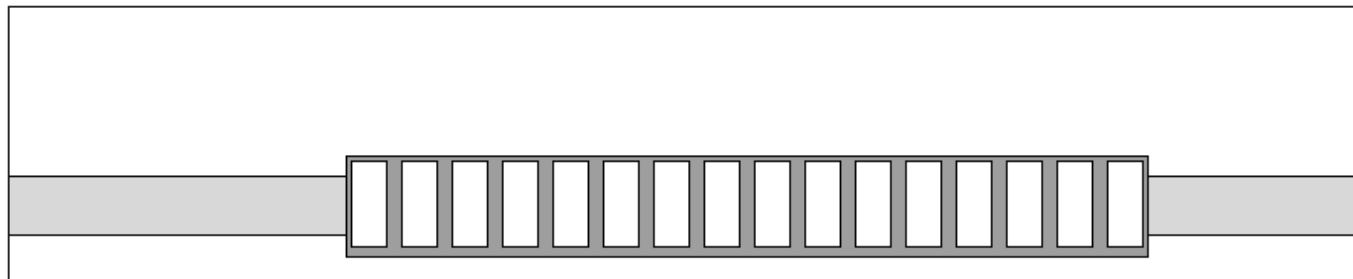
#### automatic emission of packets

start

starts or stops the automatic emission of packets by the upper layer

### legend

- packet from sender 0
- packet from sender 1
- packet from sender 2



<http://www.ccs-labs.org/teaching/rn/animations/queue/index.htm>