

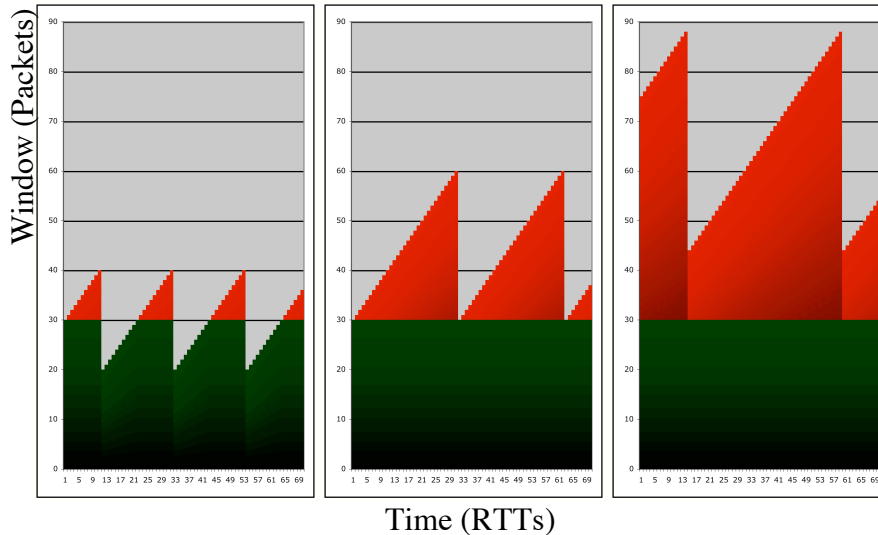
# 17: Queue Management

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## Queuing

- The primary purpose of a queue in an IP router is to smooth out bursty arrivals, so that the network utilization can be high.
- But queues add delay and cause jitter.
  - Delay is the enemy of real-time network traffic.
  - Jitter is turned into delay at the receiver's playout buffer.
  - Understanding and controlling network queues is key to getting good performance from networked multimedia.

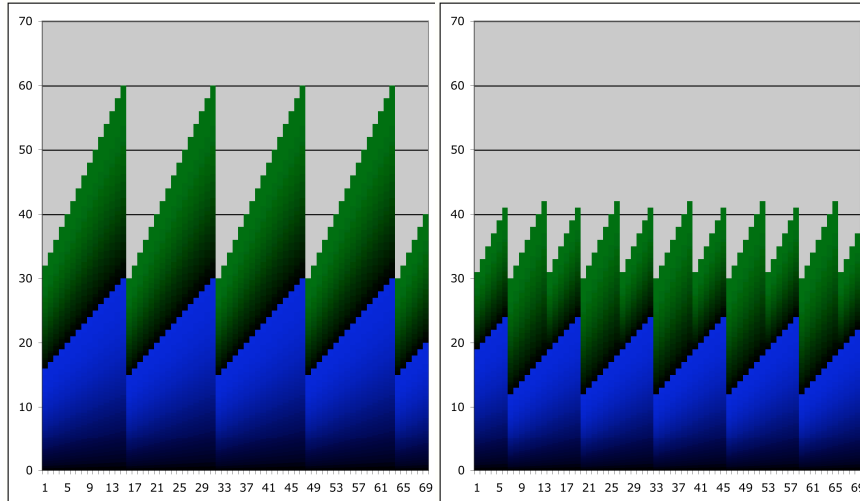
## TCP Throughput and Queue Size



## TCP and Queues

- TCP needs one delay-bandwidth product of buffer space at the bottleneck link for a TCP flow to fill the link and achieve 100% utilization.
- Thus, when everything is configured correctly, the peak delay is twice the underlying network delay.
  - Links are often overbuffered, because the actual RTT is unknown to the link operator.
  - Real-time applications see the difference between peak and min as jitter, and smooth to peak delay.

## Two TCP Flows (Effects of Phase)



## Multiple TCP flows and Queues

- If multiple flows all back-off in phase, the router still needs a delay-bandwidth product of buffering.
- If multiple flows back-off out of phase, high utilization can be maintained with smaller queues.
  - How to keep the flows out of phase?



## Active Queue Management



### Goals of Active Queue Management

- The primary goal: Controlling average queuing delay, while still maintaining high link utilization.
- Secondary goals:
  - Improving fairness (e.g., by reducing biases against bursty low-bandwidth flows).
  - Reducing unnecessary packet drops.
  - Reducing global synchronization (i.e., for environments with small-scale statistical multiplexing).
  - Accommodating transient congestion (lasting less than a round-trip time).

## Random Early Detection (RED)

- As queue builds up, randomly drop or mark packets with increasing probability (before queue gets full).
- Advantages:
  - Lower average queuing delay.
  - Avoids penalizing streams with large bursts.
  - Desynchronizes co-existing flows.

## RED Algorithm

```
for each packet arrival
  calculate the new average queue size  $q_{avg}$ 
  if  $min_{th} < q_{avg} < max_{th}$ 
    calculate probability  $p_a$ 
    with probability  $p_a$ :
      mark/drop the arriving packet
  else if  $max_{th} < q_{avg}$ 
    drop the arriving packet
```

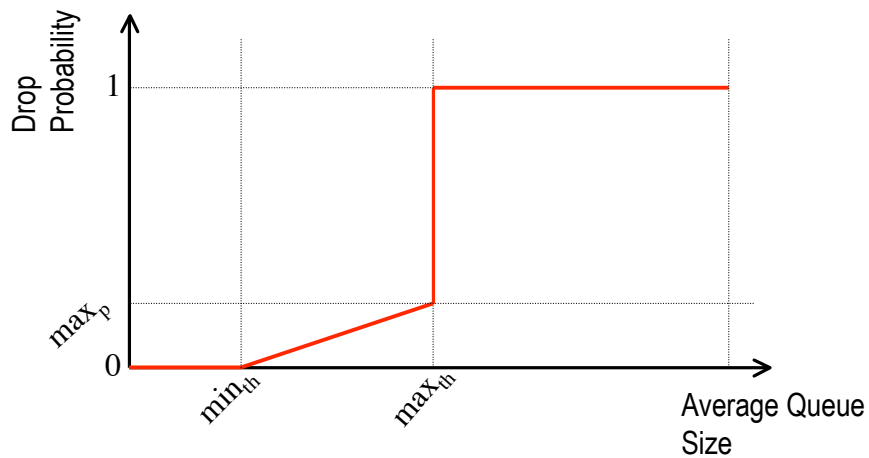
### Variables:

$q_{avg}$  : average queue size  
 $p_a$  : packet marking or  
dropping probability

### Parameters:

$min_{th}$  : minimum threshold for  
queue  
 $max_{th}$  : maximum threshold for  
queue

## RED Drop Probabilities



## The argument for using the *average* queue size in AQM

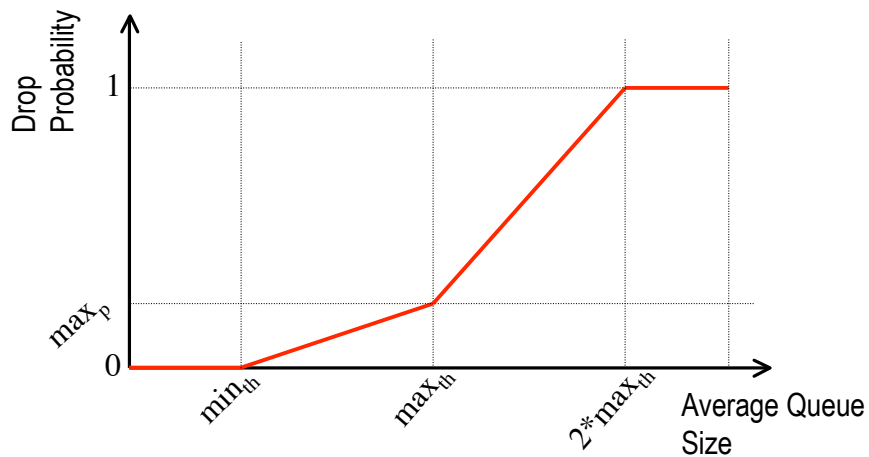
To be robust against transient bursts:

- When there is a transient burst, to drop just enough packets for end-to-end congestion control to come into play.
- To avoid biases against bursty low-bandwidth flows.
- To avoid unnecessary packet drops from the transient burst of a TCP connection slow-starting.

## The problem with RED

- Parameter sensitivity
  - How to set  $min_{th}$ ,  $max_{th}$  and  $max_p$ ?
- Goal is to maintain mean queue size below the midpoint between  $min_{th}$  and  $max_{th}$  in times of normal congestion.
  - $max_{th}$  needs to be significantly below the maximum queue size, because short-term transients peak well above the average.
  - $max_p$  primarily determines the drop rate. Needs to be significantly higher than the drop rate required to keep the flows under control.
- In reality it's hard to set the parameters robustly, even if you know what you're doing.

## RED Drop Probabilities (Gentle Mode)



## Other AQM schemes.

- Adaptive RED (ARED)
- Proportional Integral (PI)
- Virtual Queue (VQ)
- Random Exponential Marking (REM)
- Dynamic-RED (DRED)
- Blue
- Many other variants... (a lot of PhDs in this area!)

## Explicit Congestion Notification

## Explicit Congestion Notification (ECN)

- Standard TCP:
  - Losses needed to detect congestion
  - Wasteful and unnecessary
- ECN:
  - Routers mark packets instead of dropping them.
  - Receiver returns marks to sender in ACK packets.
  - Sender adjusts it's window as it would have done if the packet had been dropped.
- Advantages:
  - Bandwidth up to bottleneck not wasted.
  - No delay imposed by retransmission.

## ECN: Backwards Compatibility

- When congestion experienced, a bit in the IP header indicates if both hosts implement ECN.
  - If they do, router marks packet.
  - If they don't, router drops packet.

## Explicit Congestion Notification Codepoints

```
+---+  
| ECN FIELD |  
+---+
```

ECT	CE	The ECT and CE bits defined in RFC 2481.
0	0	Not-ECT
0	1	ECT(1) (used as an ECN nonce)
1	0	ECT(0)
1	1	CE

The ECN Field in the IP Header.

- ECT** : ECN-Capable Transport
- CE** : Congestion Experienced.

## ECN and AQM

- ECN is only useful if the queue isn't full.
  - Otherwise the router has to drop the packet whether it wants to or not.
- An active queue management scheme like RED is needed to set the ECN *Congestion Experienced* bit before the queue fills up.

## Summary

Multimedia traffic has tight delay constraints.

- Droptail queuing gives unnecessarily large queuing delays if good utilization is needed.
- Packet loss as a signal of congestion hurts real-time traffic much more than it hurts file transfer.
  - No time to retransmit.

AQM combined with ECN can give low loss, low-ish delay, moderate jitter service.

- No admission control or charging needed.
- But no guarantees either - it's still best-effort.

## References

- S. Floyd and V. Jacobson, *Random Early Detection gateways for Congestion Avoidance*, IEEE/ACM Transactions on Networking, V.1 N.4, August 1993, p. 397-413.
- B. Braden et al, *Recommendations on Queue Management and Congestion Avoidance in the Internet*, April 1998, RFC 2309.
- S. Floyd, *TCP and Explicit Congestion Notification*, ACM Computer Communication Review, V. 24 N. 5, October 1994, p. 10-23.
- D. Wetherall, D. Ely, and N. Spring, *Robust ECN Signaling with Nonces*, RFC 3540, June 2003