# Congestion Control

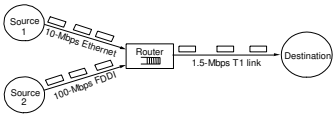
Outline

- Queuing Discipline
- Reacting to Congestion
- Avoiding Congestion



## Issues

- Two sides of the same coin
  - pre-allocate resources so at to avoid congestion
  - control congestion if (and when) is occurs

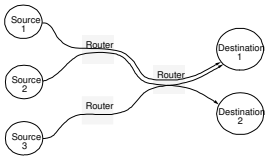



- Two points of implementation
  - hosts at the edges of the network (transport protocol)
  - routers inside the network (queuing discipline)
- Underlying service model
  - best-effort (assume for now)
  - multiple *qualities of service* (later)



## Framework

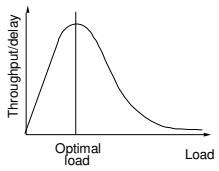
- Connectionless flows
  - sequence of packets sent between source/destination pair
  - maintain *soft state* at the routers




- Taxonomy
  - router-centric versus host-centric
  - reservation-based versus feedback-based
  - window-based versus rate-based

## Evaluation

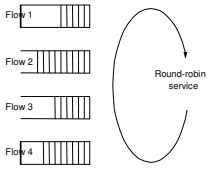
- Fairness
- Power (ratio of throughput to delay)






## Queuing Discipline

- First-In-First-Out (FIFO)
  - does not discriminate between traffic sources
- Fair Queuing (FQ)
  - explicitly segregates traffic based on flows
  - ensures no flow captures more than its share of capacity
  - variation: weighted fair queuing (WFQ)
- Problem?



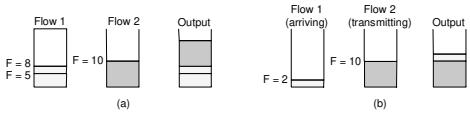



## FQ Algorithm

- Suppose clock ticks each time a bit is transmitted
- Let $P_i$ denote the length of packet $i$
- Let $S_i$ denote the time when start to transmit packet $i$
- Let $F_i$ denote the time when finish transmitting packet $i$
- $F_i = S_i + P_i$
- When does router start transmitting packet $i$?
  - if before router finished packet $i$ - 1 from this flow, then immediately after last bit of $i$ - 1 ($F_{i-1}$)
  - if no current packets for this flow, then start transmitting when arrives (call this $A_i$)
- Thus: $F_i = MAX\ (F_{i-1}, A_i) + P_i$

## FQ Algorithm (cont)

- For multiple flows
  - calculate $F_i$ for each packet that arrives on each flow
  - treat all $F_i$'s as timestamps
  - next packet to transmit is one with lowest timestamp
- Not perfect: can't preempt current packet
- Example






## TCP Congestion Control

- Idea
  - assumes best-effort network (FIFO or FQ routers) each source determines network capacity for itself
  - uses implicit feedback
  - ACKs pace transmission (*self-clocking*)
- Challenge
  - determining the available capacity in the first place
  - adjusting to changes in the available capacity



## Additive Increase/Multiplicative Decrease

- Objective: adjust to changes in the available capacity
- New state variable per connection: **`CongestionWindow`**
  - limits how much data source has in transit

```
MaxWin = MIN(CongestionWindow,
             AdvertisedWindow)
EffWin = MaxWin – (LastByteSent –
                   LastByteAcked)
```

- Idea:
  - increase **`CongestionWindow`** when congestion goes down
  - decrease **`CongestionWindow`** when congestion goes up

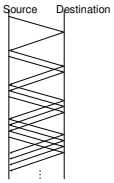
## AIMD (cont)

- Question: how does the source determine whether or not the network is congested?
- Answer: a timeout occurs
  - timeout signals that a packet was lost
  - packets are seldom lost due to transmission error
  - lost packet implies congestion



## AIMD (cont)

- Algorithm
  - increment **`CongestionWindow`** by one packet per RTT (*linear increase*)
  - divide **`CongestionWindow`** by two whenever a timeout occurs (*multiplicative decrease*)



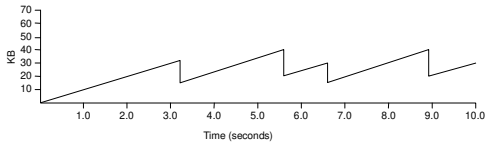

- In practice: increment a little for each ACK

```
Increment = (MSS * MSS)/CongestionWindow
CongestionWindow += Increment
```



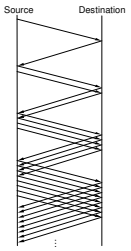
## AIMD (cont)

- Trace: sawtooth behavior



## Slow Start

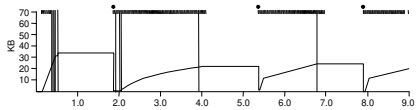
- Objective: determine the available capacity in the first
- Idea:
  - begin with `CongestionWindow` = 1 packet
  - double `CongestionWindow` each RTT (increment by 1 packet for each ACK)






## Slow Start (cont)

- Exponential growth, but slower than all at once
- Used…
  - when first starting connection
  - when connection goes dead waiting for timeout
- Trace

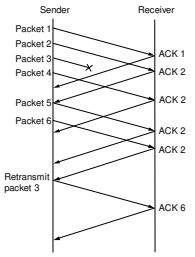



- Problem: lose up to half a `CongestionWindow`'s worth of data

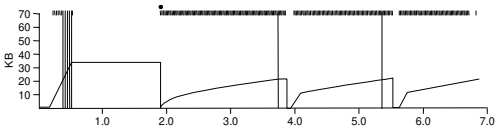


## Fast Retransmit and Fast Recovery

- Problem: coarse-grain TCP timeouts lead to idle periods
- Fast retransmit: use duplicate ACKs to trigger retransmission



## Results




- Fast recovery
  - skip the slow start phase
  - go directly to half the last successful `CongestionWindow (ssthresh)`



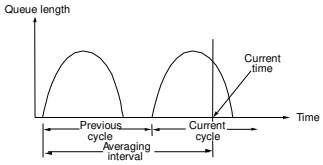
## Congestion Avoidance

- TCP's strategy
  - control congestion once it happens
  - repeatedly increase load in an effort to find the point at which congestion occurs, and then back off
- Alternative strategy
  - predict when congestion is about to happen
  - reduce rate before packets start being discarded
  - call this congestion *avoidance*, instead of congestion *control*
- Two possibilities
  - router-centric: DECbit and RED Gateways
  - host-centric: TCP Vegas



## DECbit

- Add binary congestion bit to each packet header
- Router
  - monitors average queue length over last busy+idle cycle




  - set congestion bit if average queue length > 1
  - attempts to balance throughout against delay

## End Hosts

- Destination echoes bit back to source
- Source records how many packets resulted in set bit
- If less than 50% of last window's worth had bit set
  - increase **`CongestionWindow`** by 1 packet
- If 50% or more of last window's worth had bit set
  - decrease **`CongestionWindow`** by 0.875 times



## Random Early Detection (RED)

- Notification is implicit
  - just drop the packet (TCP will timeout)
  - could make explicit by marking the packet
- Early random drop
  - rather than wait for queue to become full, drop each arriving packet with some *drop probability* whenever the queue length exceeds some *drop level*



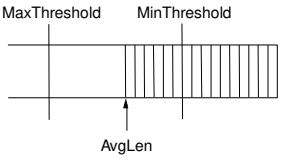
## RED Details

- Compute average queue length

```
AvgLen = (1 - Weight) * AvgLen +
         Weight * SampleLen
```

0 < **`Weight`** < 1 (usually 0.002)

**`SampleLen`** is queue length each time a packet arrives



## RED Details (cont)

- Two queue length thresholds

```
if AvgLen <= MinThreshold then
   enqueue the packet
if MinThreshold < AvgLen < MaxThreshold then
   calculate probability P
   drop arriving packet with probability P
if ManThreshold <= AvgLen then
   drop arriving packet
```

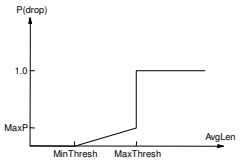


## RED Details (cont)

- Computing probability P

```
TempP = MaxP * (AvgLen – MinThreshold)/
        (MaxThreshold – MinThreshold)
P = TempP/(1 – count * TempP)
```

- Drop Probability Curve



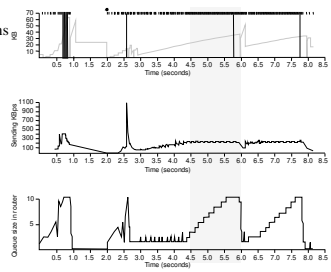



## Tuning RED

- Probability of dropping a particular flow's packet(s) is roughly proportional to the share of the bandwidth that flow is currently getting
- **`MaxP`** is typically set to 0.02, meaning that when the average queue size is halfway between the two thresholds, the gateway drops roughly one out of 50 packets.
- If traffic id bursty, then **`MinThreshold`** should be sufficiently large to allow link utilization to be maintained at an acceptably high level
- Difference between two thresholds should be larger than the typical increase in the calculated average queue length in one RTT; setting **`MaxThreshold`** to twice **`MinThreshold`** is reasonable for traffic on today's Internet
- Penalty Box for Offenders

# TCP Vegas

- Idea: source watches for some sign that router's queue is building up and congestion will happen too; e.g.,
  - RTT grows
  - sending rate flattens






# Algorithm

- Let **`BaseRTT`** be the minimum of all measured RTTs (commonly the RTT of the first packet)
- If not overflowing the connection, then

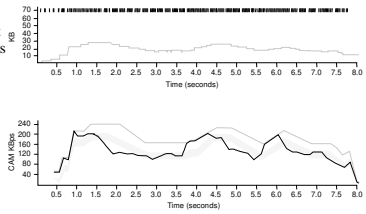
  **`ExpectRate = CongestionWindow/BaseRTT`**
- Source calculates sending rate (**`ActualRate`**) once per RTT
- Source compares **`ActualRate`** with **`ExpectRate`**

```
Diff = ExpectedRate – ActualRate
if Diff < α
    increase CongestionWindow linearly
else if Diff > β
    decrease CongestionWindow linearly
else
    leave CongestionWindow unchanged
```



# Algorithm (cont)

- Parameters
  - $\alpha$ = 1 packet
  - $\beta$ = 3 packets




- Even faster retransmit
  - keep fine-grained timestamps for each packet
  - check for timeout on first duplicate ACK