Congestion Control

Outline

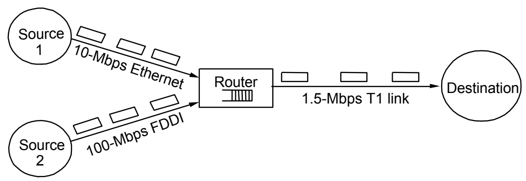
Queuing Discipline

Reacting to Congestion

Avoiding Congestion

Issues

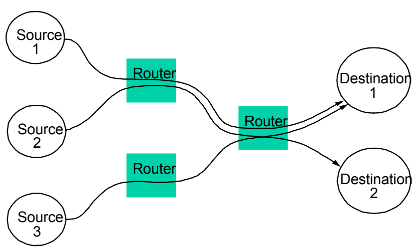
- Two sides of the same coin
  - pre-allocate resources so at to avoid congestion
  - control congestion if (and when) is occurs



- 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

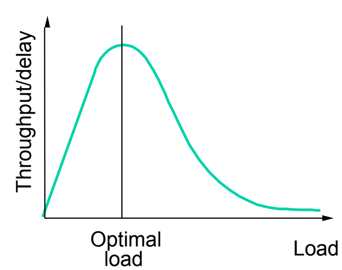
- Connectionless flows
  - sequence of packets sent between source/destination pair
  - maintain *soft state* at the routers



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

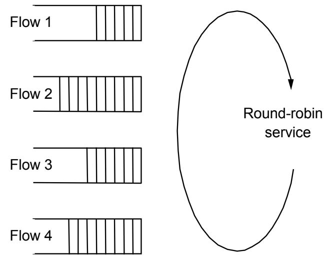
Evaluation

- 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?

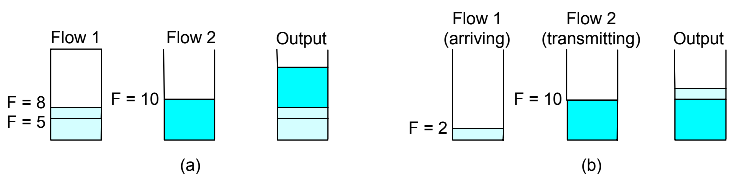


FQ Algorithm

- 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

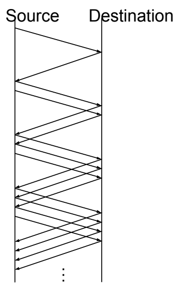
AIMD (cont)

- 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*)

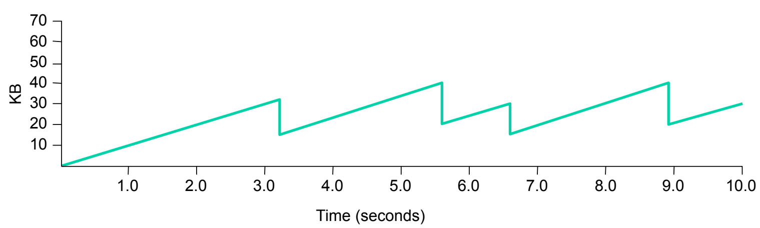


- In practice: increment a little for each ACK

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

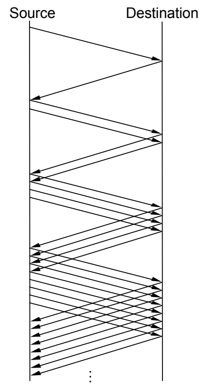

AIMD (cont)

- Trace: sawtooth behavior



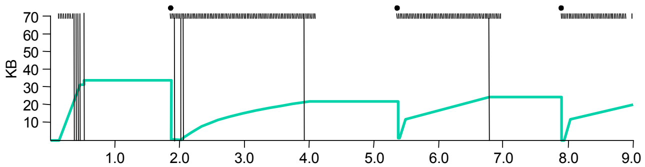
Slow Start

- 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)

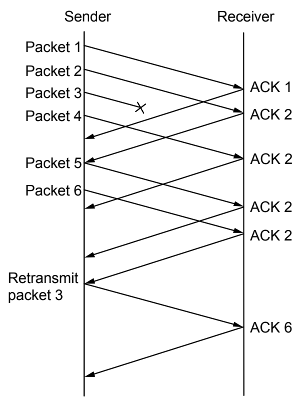
- Exponential growth, but slower than all at once
- Used…
  - when first starting connection
  - when connection goes dead waiting for timeout
- Trace



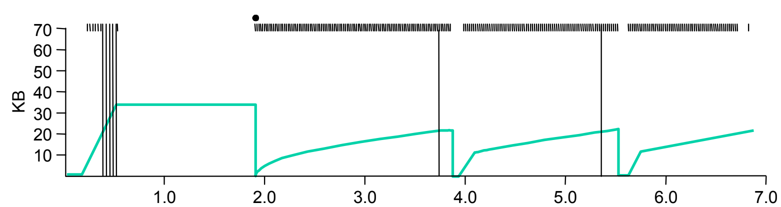
- Problem: lose up to half a **`CongestionWindow`**'s worth of data

Fast Retransmit and Fast Recovery

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



Results



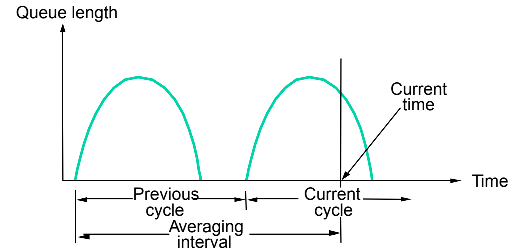
- Fast recovery
  - skip the slow start phase
  - go directly to half the last successful **`CongestionWindow (ssthresh)`**

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*

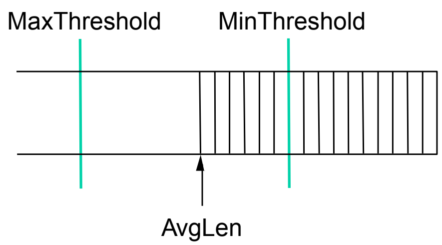
RED Details

- 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

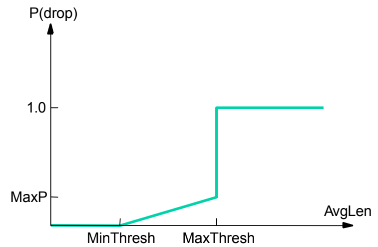
```
if AvgLen <= MinThreshold then
   enqueue the packet
if MinThreshold < AvgLen < MaxThreshold then
   calculate probability P
   drop arriving packet with probability P
if MaxThreshold <= AvgLen then
   drop arriving packet
```


RED Details (cont)

- Computing probability P

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

- Drop Probability Curve

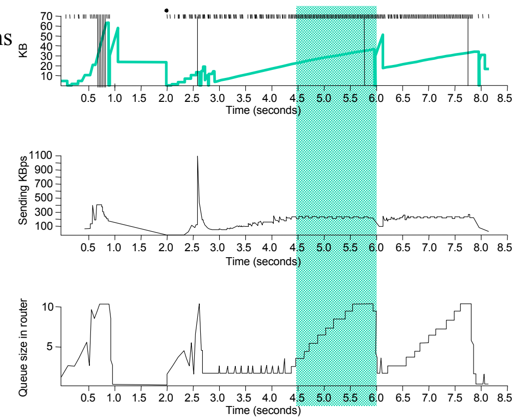


Tuning RED

- 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

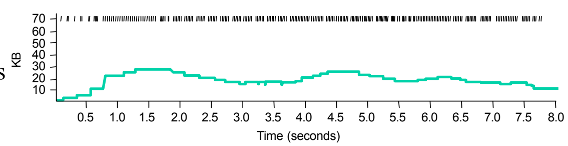
```
    ExpectRate = CongestionWindow/BaseRTT
```

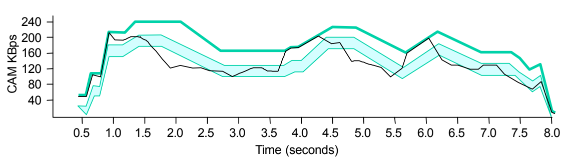
- Source calculates sending rate (**ActualRate**) once per RTT
- Source compares **ActualRate** with **ExpectRate**

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


Algorithm (cont)

- Parameters
  - α = 1 packet
  - β = 3 packets





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