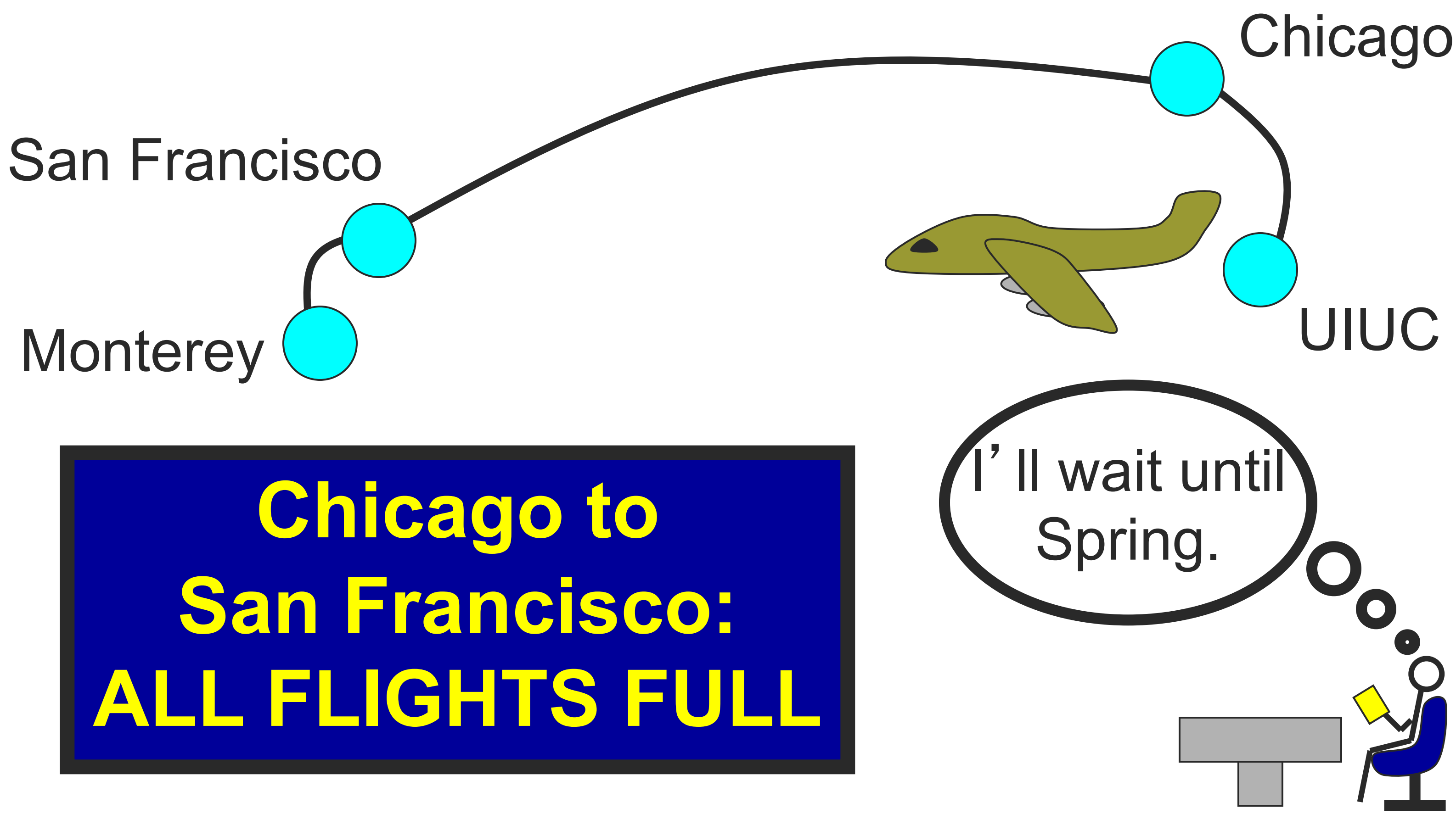
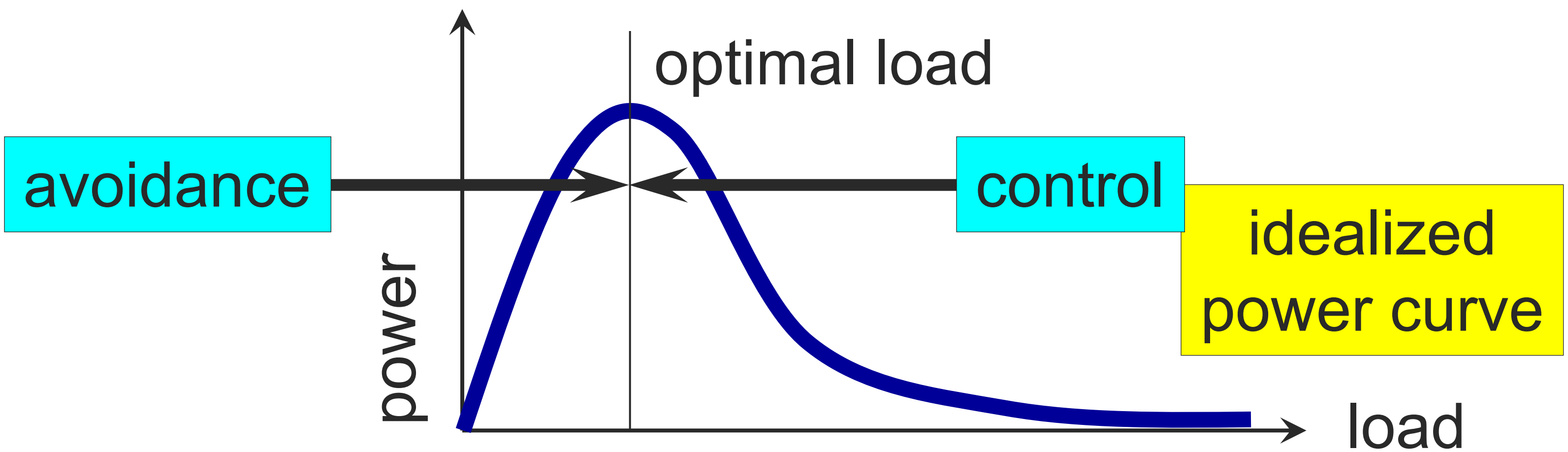
Next Topic: Vacation Planning



Congestion Avoidance

- Control vs. avoidance
  - Control: minimize impact of congestion when it occurs
  - Avoidance: avoid producing congestion
- In terms of operating point limits



Congestion Avoidance

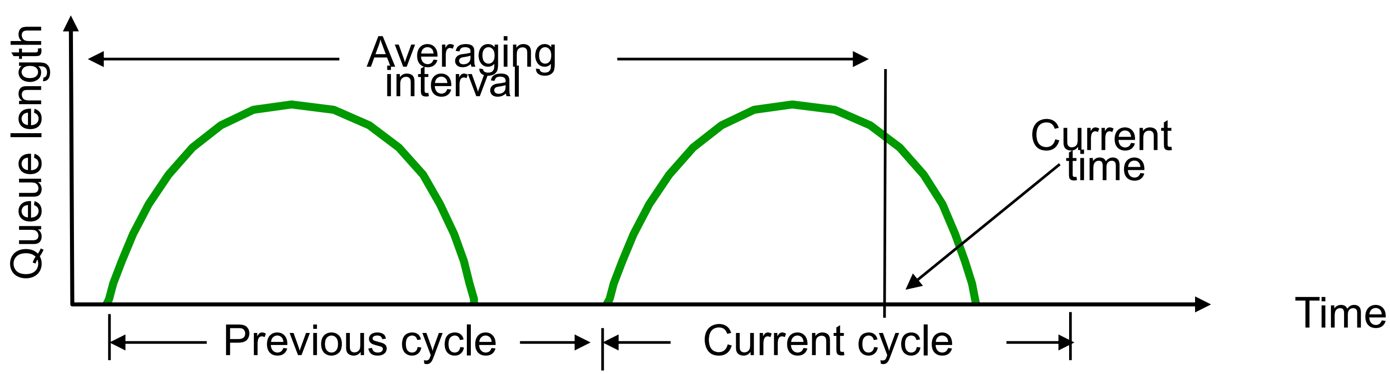
- TCP's strategy
  - Control congestion once it happens
  - Repeatedly increase load in an effort to find the point at which congestion occurs, then back off
- Alternative Strategy
  - Predict when congestion is about to happen and reduce the rate at which hosts send data just before packets start being discarded
  - Congestion avoidance, as compared to congestion control
- Two possibilities
  - Host-centric
    - TCP Vegas (may get some help from routers as in DECbit or via RED gateways)
  - Router-centric
    - Virtual circuits with reserved resources (ATM, RSVP)

DECbit (Destination Experiencing Congestion Bit)

- Developed for the Digital Network Architecture
- Basic idea
  - One bit allocated in packet header
  - Any router experiencing congestion sets bit
  - Destination returns bit to source
  - Source adjusts rate based on bits
- Note that responsibility is shared
  - Routers identify congestion
  - Hosts act to avoid congestion

DECbit

- Router
  - Monitors length over last busy + idle cycle
  - Sets congestion bit if average queue length is greater then 1 when packet arrives
  - Attempts to balance throughput against delay
    - smaller values result in more idle time
    - larger values result in more queueing delay



DECbit

- End Hosts
  - Destination echoes congestion bit back to source
  - Source records how many packets resulted in set bit
  - If less than 50% of last window had bit set
    - Increase **`CongestionWindow`** by 1 packet
  - If 50% or more of last window had bit set
    - Decrease **`CongestionWindow`** by 0.875 percent
- Note:
  - Techniques used in DECbit known as explicit congestion notification (ECN)
  - Proposal to add ECN bit to TCP in progress

Router-Based Congestion Avoidance

- Random Early Detection (RED) gateways
  - Developed for use with TCP
  - Basic idea
    - Implicit rather than explicit notification
    - When a router is “almost” congested
    - Drop packets randomly
  - Responsibility is again shared
    - Router identifies, host acts
    - Relies on TCP’s response to dropped packets

RED Overview

- Observation
  - Transient congestion
    - Should be accommodated for by having large enough queues
  - Longer-lived congestion
    - Reflected as an increase in the average queue size
- Approach
  - Detect incipient congestion from average queue size
    - Upper bound for average queue length

RED Overview

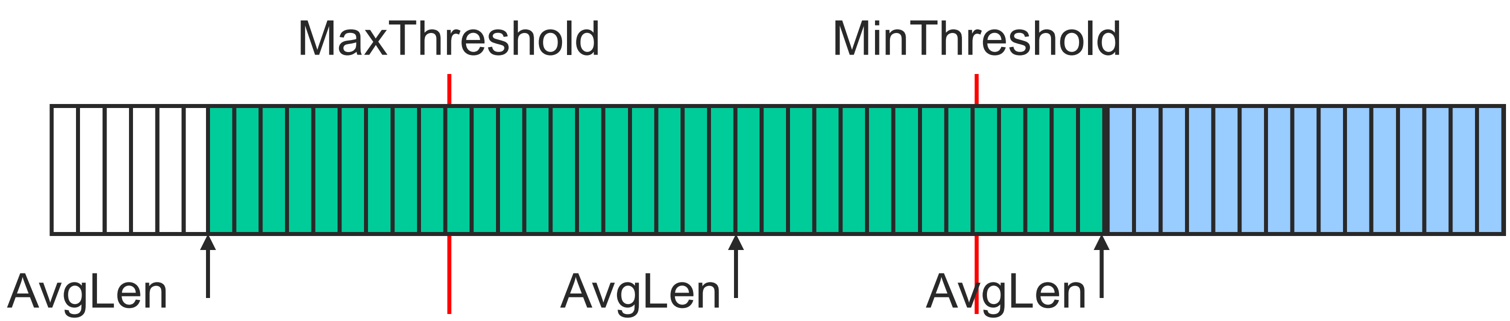
- Notify the end host of congestion
  - Dropping packet
  - Marking packet
- Select connections randomly
  - Avoid global synchronization
- Change dropping probability dynamically
- Avoid bias against bursty data
  - Use average queue length
  - Random marking

Random Early Detection (RED)

- Hosts
  - Implement TCP congestion control
  - Back off when a packet is dropped
- Routers
  - Compute average queue length (exponential moving average)
    - AvgLen = (1 – Weight) * AvgLen + Weight * SampleLen
    - 0 < Weight < 1 (usually 0.002)
    - SampleLen is queue length at packet arrival time

RED – Dropping Policy

- If AvgLen ≤ MinThreshold
  - Enqueue packet
- If MinThreshold < AvgLen < MaxThreshold
  - Calculate P and drop arriving packet with probability P
- If MaxThreshold ≤ AvgLen
  - Drop arriving packet

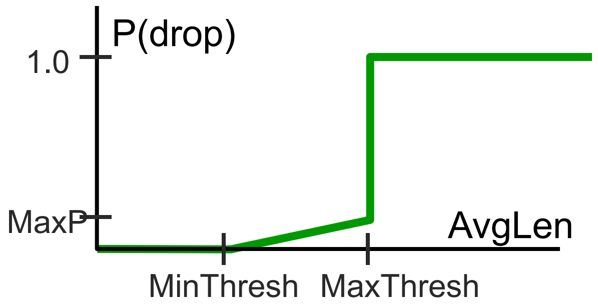


RED – Dropping Probability

- Computing P
  - P is a function of AvgLen and Count
  - Count is the number of packets that have arrived since last reset
  - Reset happens when either a packet is dropped or AvgLen is above MaxThreshold

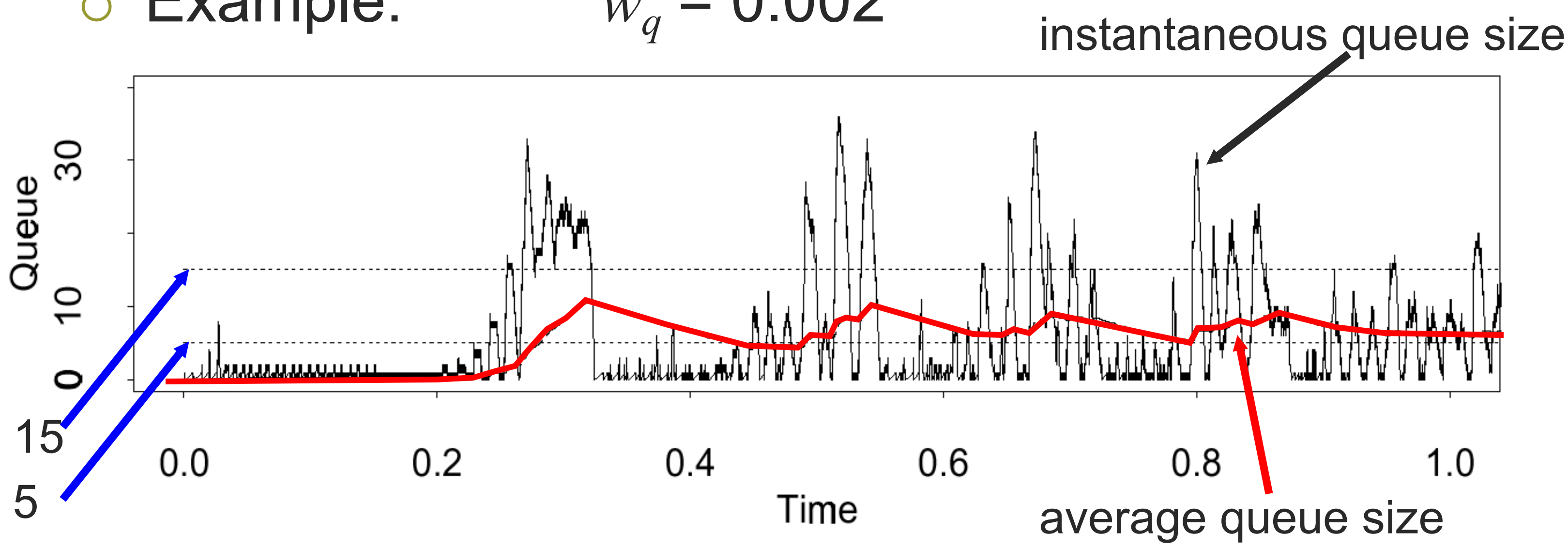
$$\text{TempP} = \frac{(\text{MaxP}) * (\text{AvgLen} - \text{MinThreshold})}{\text{MaxThreshold} - \text{MinThreshold}}$$

$$P = \frac{\text{TempP}}{(1 - \text{count} * \text{TempP})}$$



Calculate Average Queue Size

- Low pass filter $avg \leftarrow (1 - w_q)avg + w_q\, q$
  - If idle: $avg \leftarrow (1 - w_q)^m avg$
  - Example: w_q = 0.002



max_{th} = 15
min_{th} = 5

min_{th} and max_{th}

- Determined by the desired average queue size
  - Should be set sufficiently to maximize network utilization
- min_{th}
  - Controls the size of bursts
- max_{th}
  - Depends on the maximum average delay
- max_{th} - min_{th}
  - Should be larger than increase in average queue size in one round trip time
  - Avoid global synchronization

RED Parameters

- MaxP is typically set to 0.02
  - When the average queue size is halfway between the two thresholds, the gateway drops roughly 1 out of 50 packets.
- MinThreshold is typically max/2
- Choosing parameters
  - Carefully tuned to maximize power function
  - Confirmed through simulation
  - But answer depends on accuracy of traffic model

Tuning RED

- Probability of dropping a particular flow's packet(s)
  - Roughly proportional to the that flow's current share of the bandwidth
- If traffic is bursty
  - `MinThreshold` should be sufficiently large to allow link utilization to be maintained at an acceptably high level
  - If no buffer space is available, RED uses Tail Drop
- 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

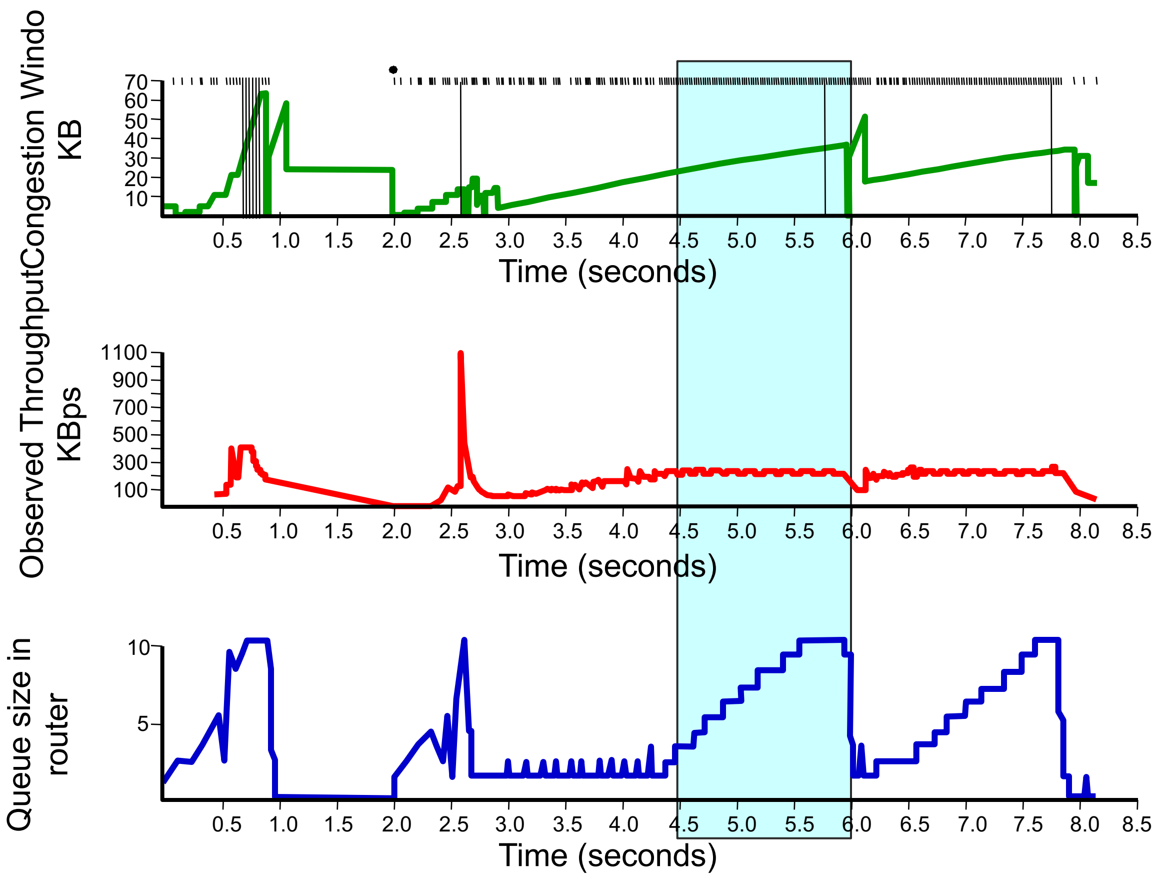
Source-Based Congestion Avoidance

- Idea
  - Source watches for some sign that some router’s queue is building up and congestion will happen soon
- Examples
  - RTT is growing
  - Sending rate flattens

Source-Based Congestion Avoidance

- Observe RTT
  - If current RTT is greater than average of minRTT and maxRTT, decrease congestion window by one-eigth
- Observe RTT and Window Size
  - Adjust window once every two RTT
    - If (CurrWindow – OldWindow) * (CurrRTT – OldRTT) > 0, decrease window by one-eigth, otherwise increase window my one MSS
- Observe sending rate
  - Increase window and compare throughput to previous value
- Observe throughput
  - Compare measured throughput with observed throughput
  - TCP Vegas

TCP Vegas



Congestion Window
KB
70
60
50
40
30
20
10
0.5 1.0 1.5 2.0 2.5 3.0 3.5 4.0 4.5 5.0 5.5 6.0 6.5 7.0 7.5 8.0 8.5
Time (seconds)
Observed Throughput
KBps
1100
900
700
500
300
100
0.5 1.0 1.5 2.0 2.5 3.0 3.5 4.0 4.5 5.0 5.5 6.0 6.5 7.0 7.5 8.0 8.5
Time (seconds)
Queue size in router
10
5
0.5 1.0 1.5 2.0 2.5 3.0 3.5 4.0 4.5 5.0 5.5 6.0 6.5 7.0 7.5 8.0 8.5
Time (seconds)

TCP Vegas

- Basic idea
  - Watch for signs of queue growth
  - In particular, difference between
    - increasing congestion window
    - stable throughput (presumably at capacity)
  - Keep just enough “extra data” in the network
    - Time to react if bandwidth decreases
    - Data available if bandwidth increases

TCP Vegas

- Implementation
  - Estimate uncongested RTT
    - baseRTT = minimum measured RTT
    - Expected throughput = congestion window / baseRTT
  - Measure throughput each RTT
    - Mark time of sending distinguished packet
    - Calculate data sent between send time and receipt of ACK

TCP Vegas

- Act to keep the difference between estimated and actual throughput in a specified range
  - Below minimum threshold
    - Increase congestion window
  - Above maximum threshold
    - Decrease congestion window
- Additive decrease used only to avoid congestion
- Want between 1 and 3 packets of extra data (used to pick min/max thresholds)

TCP Vegas Algorithm

- Let BaseRTT be 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 < a
    - Increase CongestionWindow linearly
  - else if Diff > b
    - Decrease CongestionWindow linearly
  - else
    - Leave CongestionWindow unchanged

TCP Vegas Algorithm

- Parameters
  - α = 1 packet
  - β = 3 packets
- Even faster retransmit
  - Keep fine-grained timestamps for each packet
  - Check for timeout on first duplicate ACK