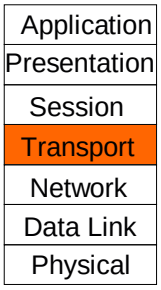
CSE/EE 461

Retransmission and Timers

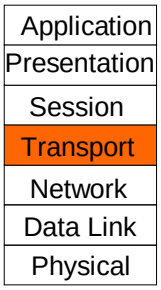
Last Time …

- More on the Transport Layer
- Focus
  - How do we manage connections?
- Topics
  - Three-Way Handshake
  - Close and TIME_WAIT



This Lecture

- Focus
  - How do we decide when to retransmit?
- Topics
  - RTT estimation
  - Karn/Partridge algorithm
  - Jacobson/Karels algorithm



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Deciding When to Retransmit

- How do you know when a packet has been lost?

```
again:
    Send(p);
    Wait(t);
    if (!p.acked)
        goto again;
```

- How long should the timer **t** be?
  - Too big: inefficient (large delays, poor use of bandwidth)
  - Too small: may retransmit unnecessarily (causing extra traffic)
  - A good retransmission timer is important for good performance
- Right timer is based on the round trip time (RTT)
  - Which varies greatly in the wide area (path length and queuing)

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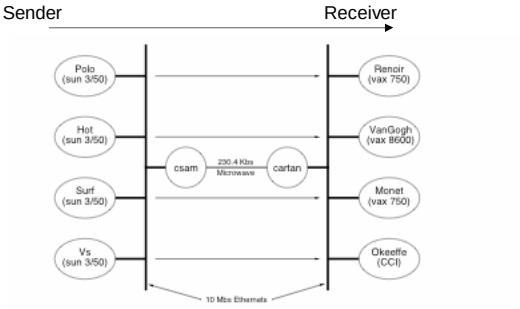
Congestion Collapse

- In the limit, early retransmissions lead to congestion collapse
  - Sending more packets into the network when it is overloaded exacerbates the problem of congestion
  - Network stays busy but very little useful work is being done
- This happened in real life ~1987
  - Led to Van Jacobson's TCP algorithms, which form the basis of congestion control in the Internet today [See "Congestion Avoidance and Control", SIGCOMM'88]
  - Observed 1000x bandwidth reduction between two hosts separated by 400 yards.
  - Led to researchers asking two questions:
    - Was TCP/IP misbehaving?
    - Could TCP/IP be "trained" to work better under 'absymal network conditions'

A Scenario

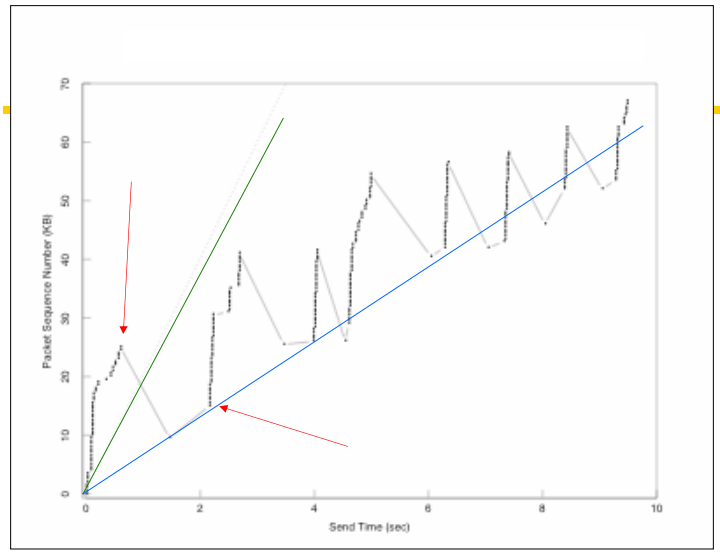


Test setup to examine the interaction of multiple, simultaneous TCP conversations sharing a bottleneck link. 1 MByte transfers (2048 512-data-byte packets) were initiated 3 seconds apart from four machines at LBL to four machines at UCB, one conversation per machine pair (the dotted lines above show the pairing). All traffic went via a 230.4 Kbps link connecting IP router **csam** at LBL to IP router **cartan** at UCB. The microwave link queue can hold up to 50 packets. Each connection was given a window of 16 KB (32 512-byte packets). Thus any two connections could overflow the available buffering and the four connections exceeded the queue capacity by 160%.

Receiver window size is 16KB.

Bottleneck router buffer size is 15 KB.

Data bandwidth is about 20KB/s



Sequence #

TIME (SEC)

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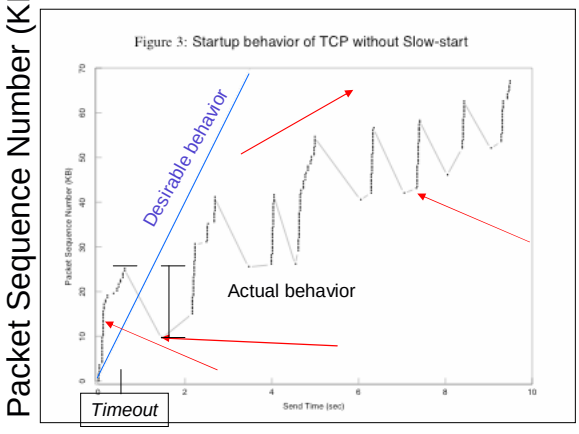
Effects of Early Retransmissions



Slope is bandwidth.

Steep smooth upward slope means good bandwidth.

Downward slope means retransmissions (*bad*).

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If only…

- We knew RTT and Current Router Queue Size,
  - Then we would send MIN(Router Queue Size, Effective Window Size)
  - And not resent a packet until it had been sent RTT ago.
- But we don't know these things, so we have to figure them out.
- And they may change dynamically due to other data sources

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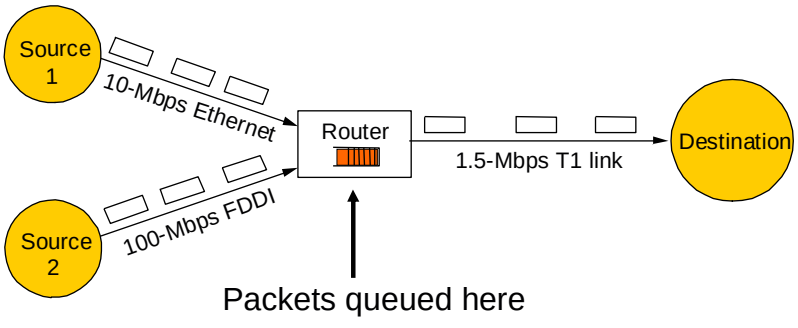
Congestion from Multiple Sources



- Buffers at routers used to absorb bursts when input rate > output
- Loss (drops) occur when sending rate is persistently > drain rate

Interpacket Spacing

Interpacket spacing mirrors that of slowest link

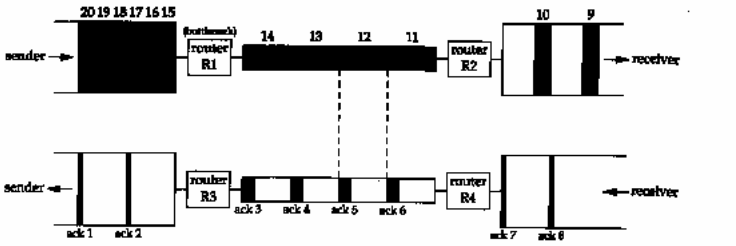


Figure 20.13 Congestion caused by a bigger pipe feeding a smaller pipe.

Inter-ACK spacing mirrors that of slowest downstream link

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1988 Observations on Congestion Collapse

- Implementation, not the protocol, leads to collapse
- "Obvious" ways of doing things lead to non-obvious and undesirable results
  - "send eff-wind-size # packets, wait rtt, try again"
- Remedial algorithms achieve network stability by forcing the transport connection to obey a 'packet conservation' principle.
  - For a connection in 'equilibrium, that is, running stably with a full window of data in transit, the packet flow is "conservative": a new packet is not put into the network until an old packet leaves.

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Resulting TCP/IP Improvements

- *Slow-start*
- *Round-trip time variance estimation*
- *Exponential retransmit timer backoff*
- *More aggressive receiver ack policy*
- *Dynamic window sizing on congestion*
- Clamped retransmit backoff (Karn)
- Fast Retransmit

Packet Conservation Principle

Congestion control means: "Finding places that violate the conservation of packets principle and then fixing them."

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In order to conserve packets

1. The connection must reach equilibrium.
   - Hurry up and stabilize!
   - When things get wobbly, put on the brakes and reconsider
2. A sender must not inject a new packet before an old packet has exited.
   - A packet "exits" when the receiver picks it up.
   - Or it gets lost
     - Damaged in transit
     - Dropped at a congestion point
     - Fewer than 1% of packets get damaged
   - Ack or packet timeout signals that a packet has "exited."
   - Acks are easy to detect.
   - Good timeouts are harder…. All about estimating RTT.
3. Equilibrium is lost because of resource contention along the way.
   - New competing stream appears

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1. The connection must reach equilibrium.

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1. Getting to Equilibrium -- Slow Start

- Goal
  - Quickly determine the appropriate window size
- Strategy
  - Introduce *congestion_window (cwnd)*
  - When starting off, or *soft restarting*, set cwnd to 1
    - A soft restart occurs if TCP decides that a packet has been lost but there are no packets in transit
      - No acks coming
  - For each ack received, add 1 to cwnd
  - When sending, send the minimum of receiver's advertised window and cwd
- Guaranteed to not transmit at more than twice the max bw, and for no more than 2 RTT.
  - (bw delay product)

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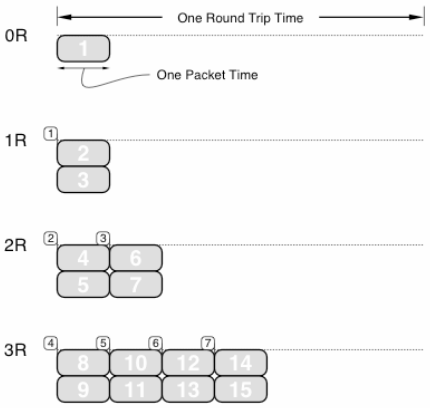
Cwnd doubles every RTT;
Opening the window of size W takes time (RTT)$\log_2 W$.

Figure 2: The Chronology of a Slow-start



The horizontal direction is time. The continuous time line has been chopped into one-round-trip-time pieces stacked vertically with increasing time going down the page. The grey, numbered boxes are packets. The white numbered boxes are the corresponding acks. As each ack arrives, two packets are generated: one for the ack (the ack says a packet has left the system so a new packet is added to take its place) and one because an ack opens the congestion window by one packet. It may be clear from the figure why an add-one-packet-to-window policy opens the window exponentially in time.

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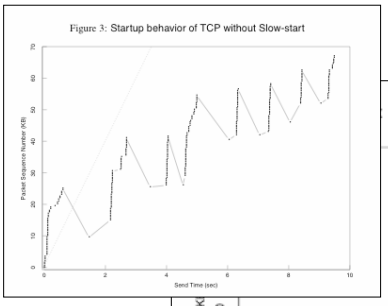
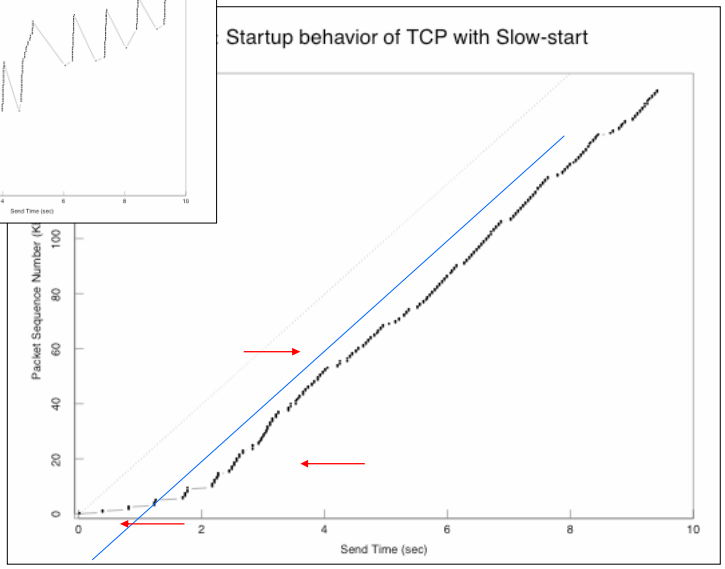
Figure 3: Startup behavior of TCP without Slow-start



Startup behavior of TCP with Slow-start



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2. A sender must not inject a new packet before an old packet has exited.

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2. Packet Injection. Estimating RTTs

- Do not inject a new packet until an old packet has left.
  - 1. Ack tells us that an old packet has left.
  - 2. Timeout expires tells us also.
    - *Gotta estimate RTT properly*.
- Strategy 1: Fixed RTT.
  - Simple, but probably wrong. (certainly not adaptive)
- Strategy 2: Estimate based on past behavior.
  - Tactic 1: Mean with exponential decay
  - Tactic 2:

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Simple Estimator

- Simple algorithm:
  - For each packet, note time sent and time ack received
  - Compute RTT samples and average recent samples for timeout

$$EstimatedRTT = (1-g)(EstimatedRTT) + g(SampleRTT)$$

  - This is an exponentially-weighted moving average (low pass filter) that smoothes the samples with a gain of *g*
    - Big *g* can be jerky (think static on a walkie talkie)
    - Small g can be soothing, but slow to respond (more stable)
    - Typically, g = .1 or .2, --> stable is better than precise
      - In other words, a lousy estimate of the RTT *right now* causes much more damage than an ok estimate right now followed by a better one a little later on.
        - » The Airplane Rule.
  - Conservatively set timeout to small multiple (2) of the estimate

$$Timeout = 2(EstimatedRTT)$$

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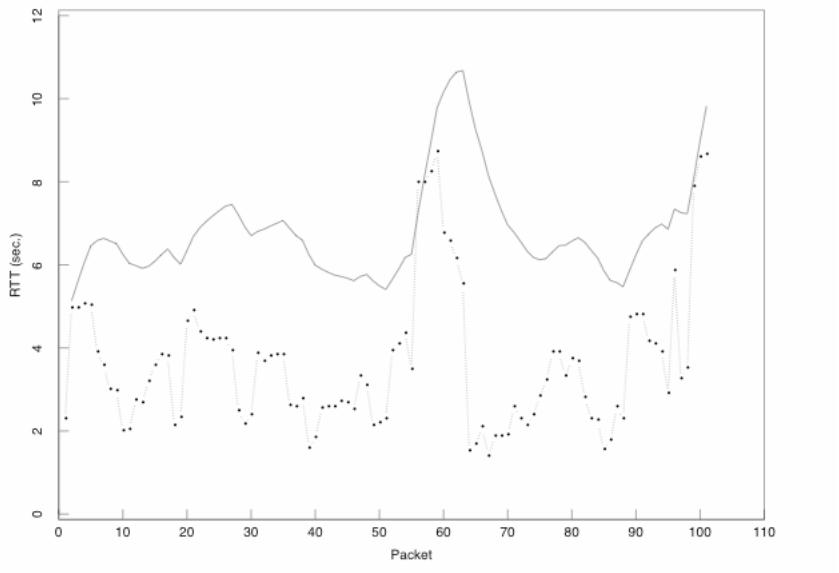
Figure 5: Performance of an RFC793 retransmit timer



Bad Estimators and the Bad Things They Do

- Problem:
  - Variance in RTTs gets large as network gets loaded
  - So an average RTT isn't a good predictor when we need it most
    - Time out too soon, unnecessarily drop another packet onto the network.
    - Timing out too soon occurs during load increase
      - if we time out when load increases but packet not yet lost, then we'll inject another packet onto the network which will increase load, which will cause more timeouts, which will increase load, until we actually starting dropping packets!

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Jacobson/Karels Algorithm

- Solution: Compute timeout on basis of:
  - Mean round trip time, AND
  - Mean deviation (*mean prediction error)*
- *TimeOut = f1 * (EstimatedRTT) + f2 * (EstimatedDeviation)*
    - *EstimatedRTT --> running guess for RTT*
    - *EstimatedDeviation --> running guess for prediction error*
  - Consider what happens when:
    - EstimatedRTT is really good
    - EstimatedRTT is really bad
- Trick is to compute these estimates cheaply

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Cheap Algorithm For Keeping Running Tabs

- Solution: Track variance too.
  - Difference = SampleRTT – EstimatedRTT
  - EstimatedRTT = EstimatedRTT + (δ x Difference)
  - Deviation = Deviation + δ(| Difference | - Deviation)
  - ***Timeout = μ x EstimatedRTT + ϕ x Deviation***
  - In practice, δ = 1/8, μ = 1 and ϕ = 4
- See paper for details

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A Fast Algorithm for RTT Mean and Variation

- Let *a* = estimated round trip time, v = estimated error, *g* = gain *(0 <g < 1)*, *m* = new sampled round trip time
- *a = (1-g)a + gm* // compute new estimate using gain
- *a = a + g(m-a)* // rearrange terms:
  - a is a prediction of next measurement, and (m-a) is the "error" in that prediction.
  - so, the new prediction is the old prediction plus some fraction of the prediction error.
  - The prediction error is the sum of two components:
    - E_r = noise (random unpredictable effects like fluctations in competing traffic)
    - E_e = bad choice of *a*
    - $a = a + g E_r + g E_e$
      - The term g E_e kicks *a* in the right direction towards the real estimate
      - The term g E_r kicks it off in the random direction
      - Over many samples, the random errors cancel each other so we get closer and closer to the real estimate
      - **But, g represents a compromise.**
        - » **Big 'g' means that we get a lot of value out of a prediction error, but it also means that the random errors introduce a lot of noise.**
      - **Since g E_e moves *a* in the right direction regardless of *g*, we're better off using a small *g* and waiting a bit longer to get a better estimate than to very quickly get a lousy estimate**
- *Or,*
  - *Err = (m - a)* // *Sampled Error*
  - *a = a + g (Err)* // *Estimate of round trip time*
  - *v = v + g(|Err| - v)* // *Estimate of error*
- Not necessary to use same gain; in general want to force timer to go up

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Estimate with Mean + Variance

Figure 5: Performance of an RFC793 retransmit timer

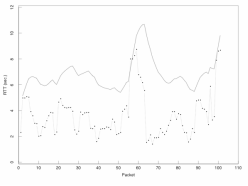
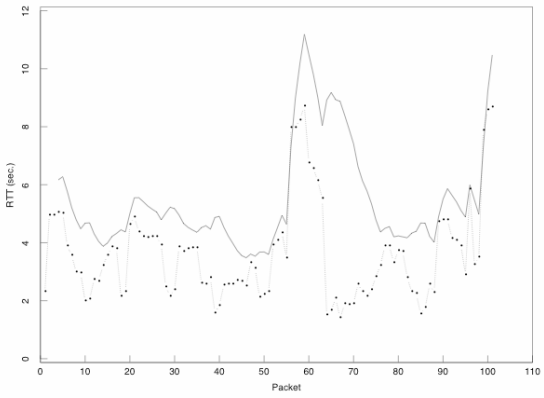


Figure 6: Performance of a Mean+Variance retransmit timer



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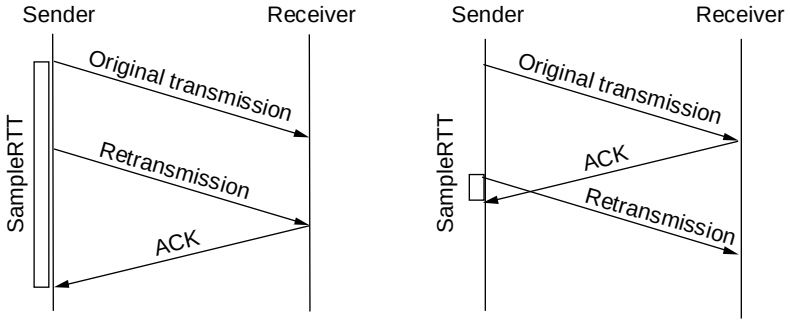
Karn/Partridge Algorithm

- Problem: RTT for retransmitted packets ambiguous



- Solution: Don't measure RTT for retransmitted packets and do not relax backed off timeout until valid RTT measurements

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3. Equilibrium is lost because of resource contention along the way.

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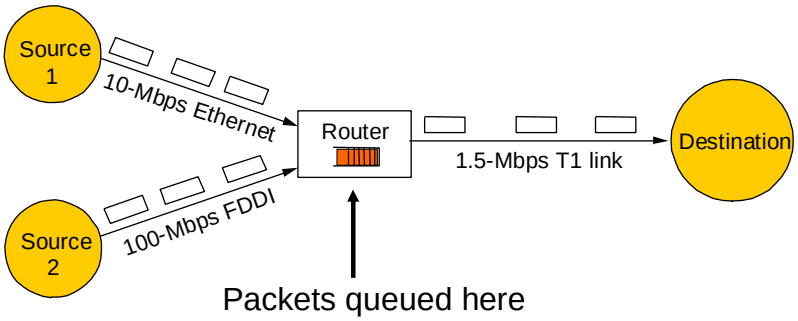
Congestion from Multiple Sources



- Buffers at routers used to absorb bursts when input rate > output
- Loss (drops) occur when sending rate is persistently > drain rate

3. Congestion Avoidance

- An avoidance strategy must have two components
  - Network must signal that congestion is, or is about to, occur
  - Endpoints must decrease load when signal occurs and increase it otherwise
- Assumption:
  - Packet loss always due to congestion and timeout always due to lost packet

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Key Concepts

- Packet conservation is a fundamental concept in TCP's congestion management
  - Get to equilibrium
    - slow start
  - Do nothing to get out of equilibrium
    - Good RTT to determine when to inject a packet
  - Adapt when equilibrium has been lost due to other's attempts to get to/stay in equilibrium
    - Congestion Avoidance

  A good retransmit timer is important for good performance
  - Too long leads to poor performance
  - Too short leads to wasted bandwidth
- An estimated timeout must adapt to Internet queuing
  - High variance at high load

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