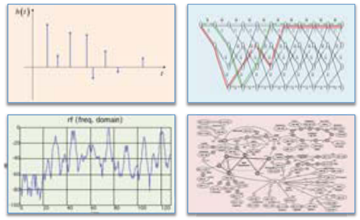


6.02 Fall 2012
Lecture #21: Reliable Data Transport

- Redundancy via careful retransmission
- Sequence numbers & acks
- Two protocols: stop-and-wait & sliding window
- Timeouts and round-trip time (RTT) estimation

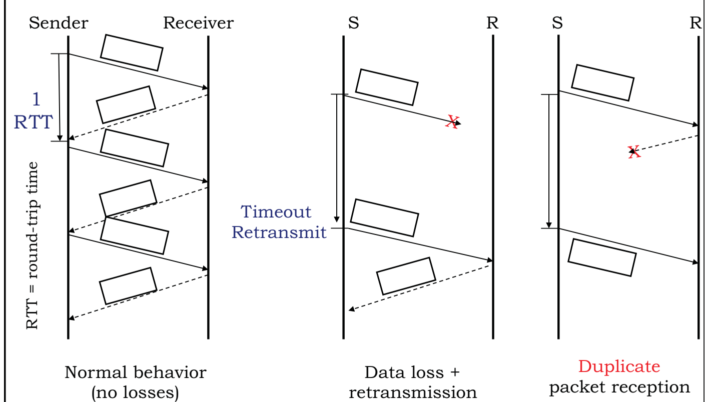
The Problem

- Given: Best-effort network in which
  - Packets may be lost arbitrarily
  - Packets may be reordered arbitrarily
  - Packet delays are variable (queueing)
  - Packets may even be duplicated
- Sender S and receiver R want to communicate reliably
  - Application at R wants *all* data bytes in exactly the same order that S sent them
  - Each byte must be delivered exactly once
- These functions are provided by a *reliable transport protocol*
  - Application "layered above" transport protocol

Proposed Plan

- Transmitter
  - Each packet includes a sequentially increasing sequence number
  - When transmitting, save (xmit time,packet) on un-ACKed list
  - When acknowledgement (ACK) is received from the destination for a particular sequence number, remove the corresponding entry from un-ACKed list
  - Periodically check un-ACKed list for packets sent awhile ago
    - Retransmit, update xmit time in case we have to do it again!
    - "awhile ago": xmit time < now — timeout
- Receiver
  - Send ACK for each received packet, reference sequence number
  - Deliver packet payload to application

Stop-and-Wait Protocol



Wanted "exactly once", got "at least once"

Revised Plan

- Transmitter
  - Each packet includes a sequentially increasing sequence number
  - When transmitting, save (xmit time,packet) on un-ACKed list
  - When acknowledgement (ACK) is received from the destination for a particular sequence number, remove the corresponding entry from un-ACKed list
  - Periodically check un-ACKed list for packets sent awhile ago
    - Retransmit, update xmit time in case we have to do it again!
    - "awhile ago": xmit time < now − timeout
- Receiver
  - Send ACK for each received packet, reference sequence number
  - Deliver packet payload to application in sequence number order
    - By keeping track of next sequence number to be delivered to app, it's easy to recognize duplicate packets and not deliver them a second time.

Issues

- Protocol must handle lost packets correctly
  - Lost data: retransmission will provide missing data
  - Lost ACK: retransmission will trigger another ACK from receiver
- Size of packet buffers
  - At transmitter
    - Buffer holds un-ACKed packets
    - Stop transmitting if buffer space an issue
  - At receiver
    - Buffer holds packets received out-of-order
    - Stop ACKing if buffer space an issue
- Choosing timeout value: related to RTT
  - Too small: unnecessary retransmissions
  - Too large: poor throughput
    - Delivery stalled while waiting for missing packets

Throughput of Stop-and-Wait

- We want to calculate the expected time, T (in seconds) between successful deliveries of packets. If N data packets are sent (N large), the time to send them will be N*T, so Throughput = N/NT = 1/T data packets per second
- We can't just assume T = RTT because packets get lost
  - E.g.: N links in the round trip between sender and receiver
  - If the per-link probability of losing a data/ACK packet is p, then the probability it's delivered over the link is (1-p), and thus the probability it's delivered over N links is $(1-p)^N$.
  - So the probability a data/ACK packet gets lost is $L = 1 - (1-p)^N$.
- Now we can write an equation for T in terms of RTT and the timeout, RTO: $T = (1-L)\cdot RTT + L\cdot\left(RTO + T\right)$

$$= RTT + \frac{L}{1-L} RTO$$

The Best Case

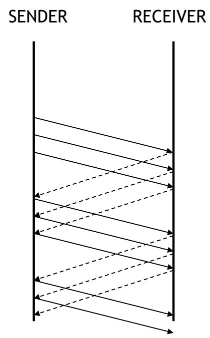
- Occurs when RTT is the same for every packet, so timeout is slightly larger than RTT

$$T = RTT + \frac{L}{1-L} RTT = \frac{1}{1-L} RTT$$

$$\text{Throughput} = \frac{(1-L)}{RTT}$$

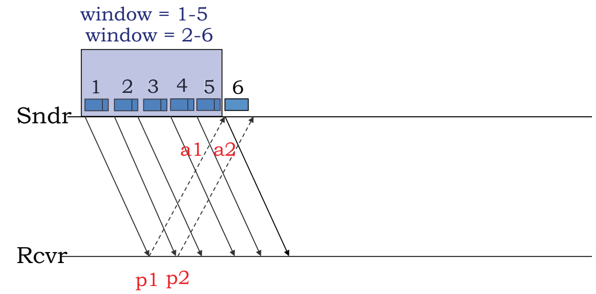
- If bottleneck link can support 100 packets/sec and the RTT is 100 ms, then, using stop-and-wait, the maximum throughput is *at most only* 10 packets/sec.
  - Urk! Only 10% utilization
  - We need a better reliable transport protocol…

Idea: *Sliding Window* Protocol



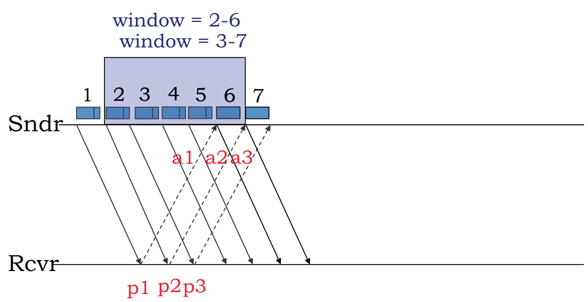
- Use a *window*
  - Allow W packets outstanding (i.e., unack'd) in the network at once (W is called the window size).
  - Overlap transmissions with ACKs
- Sender advances the window by 1 for each in-sequence ack it receives
  - I.e., window *slides*
  - So, idle period reduces
  - **Pipelining**
- Assume that the window size, W, is fixed and known
  - Later, we will discuss how one might set it
  - W = 3 in the example on the left

Sliding Window in Action



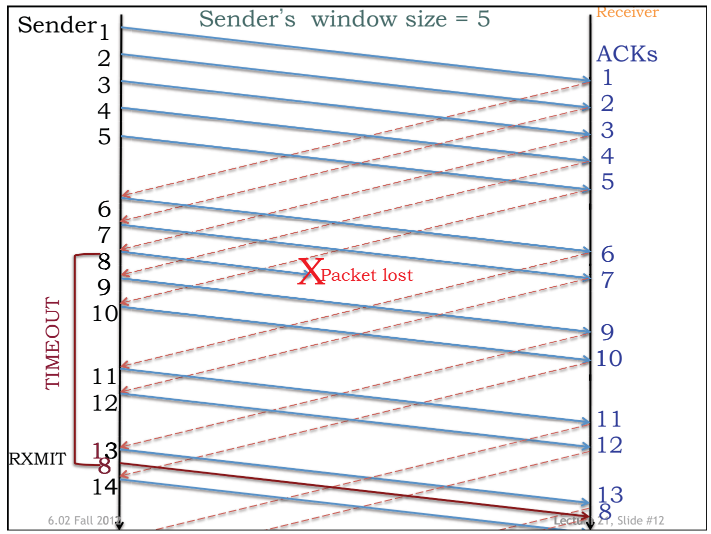
W = 5 in this example

Sliding Window in Action

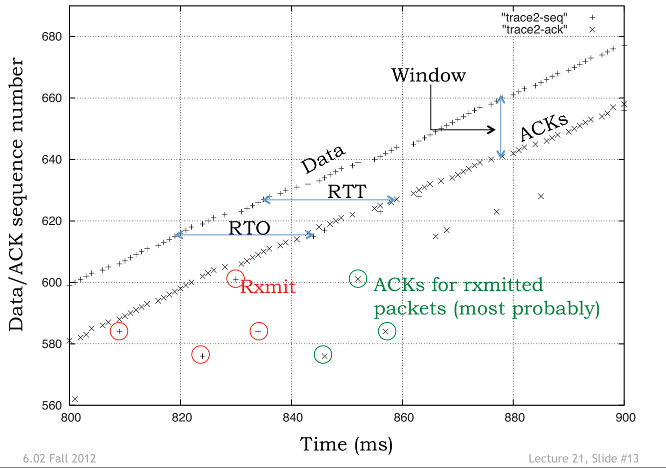


Window definition: If window is W, then max number of *unacknowledged packets* is W

This is a fixed-size sliding window



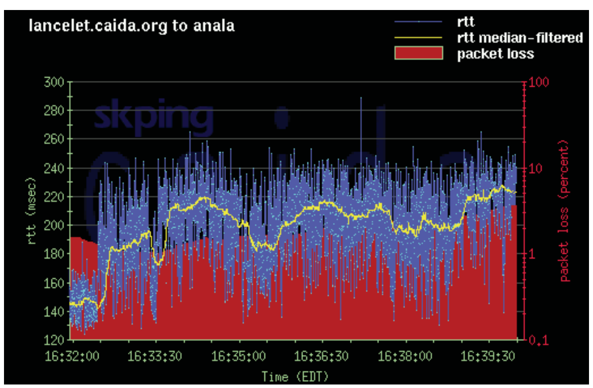
Data/ACK sequence trace



Sliding Window Implementation

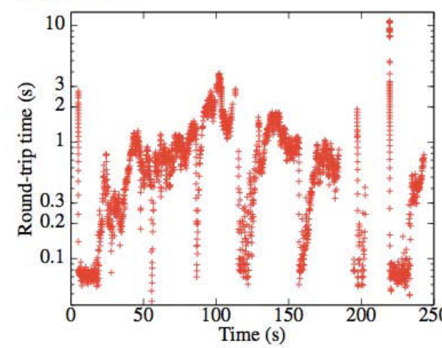
- Transmitter
  - Each packet includes a sequentially increasing sequence number
  - When transmitting, save (xmit time,packet) on un-ACKed list
  - Transmit packets if len(un-ACKed list) ≤ window size W
  - When acknowledgement (ACK) is received from the destination for a particular sequence number, remove the corresponding entry from un-ACKed list
  - Periodically check un-ACKed list for packets sent awhile ago
    - Retransmit, update xmit time in case we have to do it again!
    - "awhile ago": xmit time < now – timeout
- Receiver
  - Send ACK for each received packet, reference sequence number
  - Deliver packet payload to application in sequence number order
    - Save delivered packets in sequence number order in local buffer (remove duplicates). Discard incoming packets which have already been delivered (caused by retransmission due to lost ACK).
    - Keep track of next packet application expects. After each reception, deliver as many in-order packets as possible.

RTT Measurements

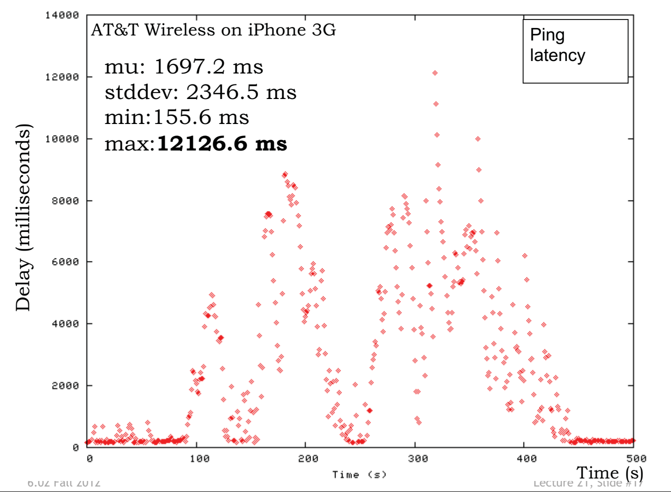


Courtesy of the Cooperative Association for Internet Data Analysis. Used with permission.

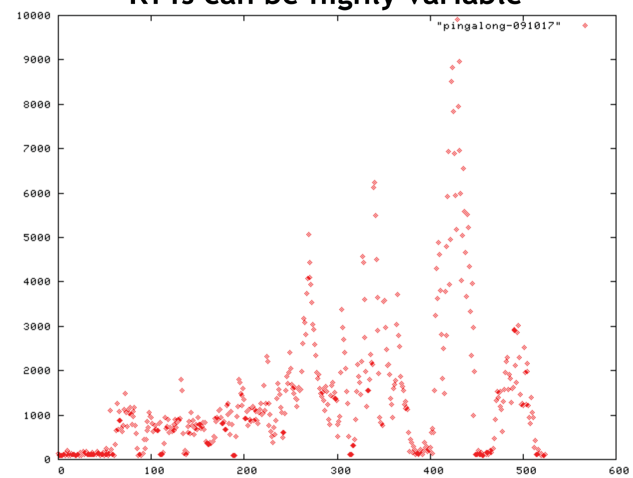
Figure 1: Round-trip time during a TCP download on the Verizon LTE network in Cambridge, Mass., Oct. 14, 2011 at 3 p.m.



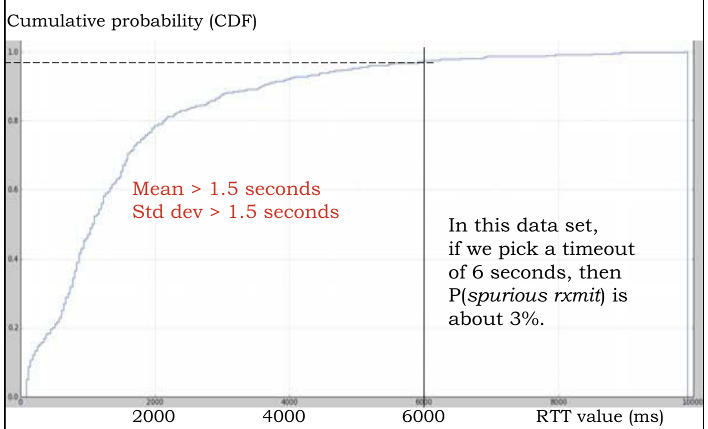
http://nms.csail.mit.edu/papers/index.php?detail=208



RTTs can be highly variable



CDF of RTT over Verizon Wireless 3G Network



Estimating RTT from Data

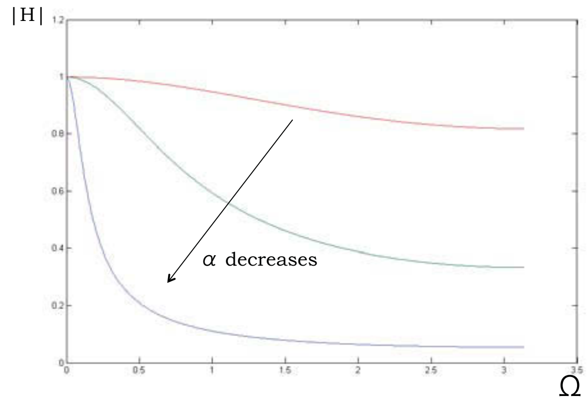
- Gather samples of RTT by comparing time when ACK arrives with time corresponding packet was transmitted
  - Sample of random variable with some unknown distribution (not necessarily Gaussian!)
- Chebyshev's inequality tells us that for a random variable X with mean μ and finite variance σ^2:

$$P(|X-\mu| \geq k\sigma) \leq \frac{1}{k^2}$$

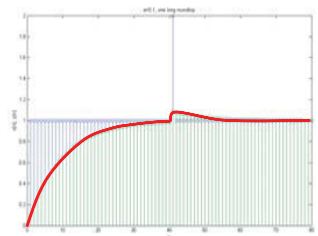
  - To reduce the chance of a *spurious* (i.e., unnecessary) retransmission – packet wasn't lost, just the round trip time for packet/ACK was long – we want our timeout to be greater than most observed RTTs
  - So choose a k that makes the chances small…
  - We need an estimate for μ and σ

Exponential Weighted Moving Average (EWMA) [A low-pass filter - see frequency response]

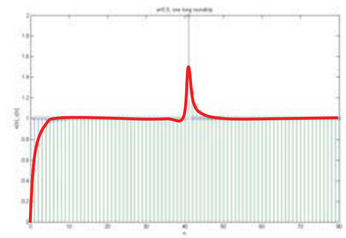
srtt ← α*rtt_sample + (1-α)*srtt



Response to One Long RTT Sample



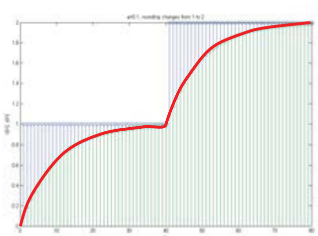
α = 0.1



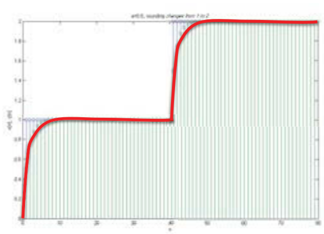
α = 0.5

Responds too quickly?

RTT changes from 1 to 2



α = 0.1



α = 0.5

Doesn't respond quickly enough?

Timeout Algorithm

- EWMA for smoothed RTT (srtt)
  - srtt ← α*rtt_sample + (1-α)*srtt
  - Typically 0.1 ≤ α ≤ 0.25 on networks prone to congestion. TCP uses α=0.125.
- Use another EWMA for smoothed RTT deviation (srttdev)
  - Mean linear deviation easy to compute (but could also do std deviation)
  - dev_sample = |rtt_sample – srtt|
  - srttdev ← β*dev_sample + (1-β)*srttdev
    TCP uses β = 0.25
- Retransmit Timeout, RTO
  - RTO = srtt + k·srttdev
  - k = 4 for TCP
  - Makes the "tail probability" of a spurious retransmission low
  - On successive *retransmission* failures, double RTO (exponential backoff)

6.02 Introduction to EECS II: Digital Communication Systems
Fall 2012