شبکه های کامپیوتری ۲

جلسه ۱۰ فصل ۳

Congestion Control

دانشگاه صنعتی اصفهان دانشکده مهندسی برق و کامپیوتر

Chapter 3 Transport Layer

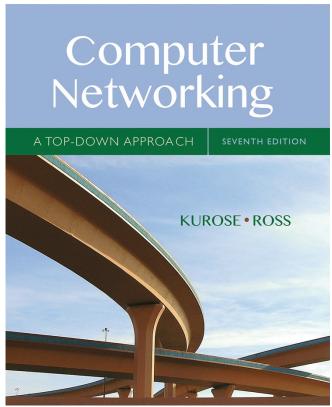
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Computer Networking: A Top Down Approach

7th edition Jim Kurose, Keith Ross Pearson/Addison Wesley April 2016

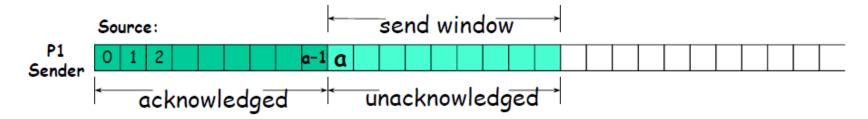
Chapter 3 outline

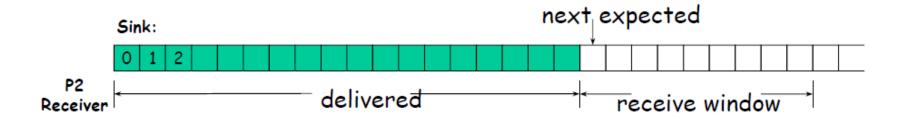
- 3.1 transport-layer services
- 3.2 multiplexing and demultiplexing
- 3.3 connectionless transport: UDP
- 3.4 principles of reliable data transfer

- 3.5 connection-oriented transport: TCP
 - segment structure
 - reliable data transfer
 - flow control
 - connection management
- 3.6 principles of congestion control
- 3.7 TCP congestion control

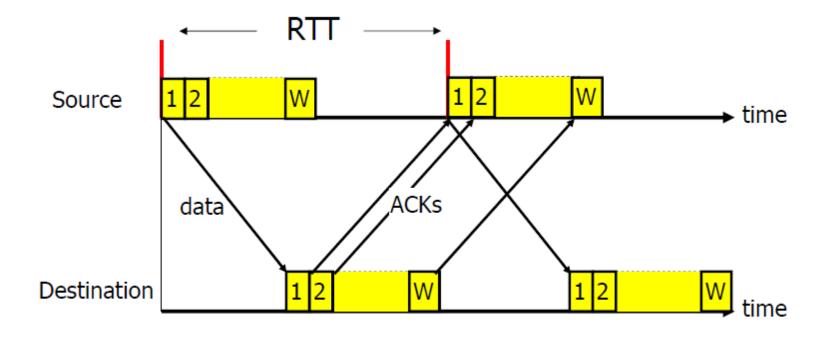
Sliding Window protocol

- Functions provided
 - reliable delivery (error and loss control)
 - in-order delivery
 - flow and congestion control
 - by varying send window size





Window Size Controls Sending Rate



~ W packets per RTT when no loss

Throughput

Max. throughput = W / RTT bytes/sec

- This is an upper bound
- Actual throughput is smaller
 - Average number in the send buffer is less than W
 - Retransmissions
- The throughput of a host's TCP send buffer is the host's send rate into the network (including original transmissions and retransmissions)

TCP Send Window Size

TCP flow control

- Avoid overloading receiver
- Receiver calculates flow control window size (rwnd) based on the available receiver buffer space
- Receiver sends flow control window size to sender in TCP segment heather
- Sender keeps Send Window size less than most recently received rwnd value

TCP Congestion Control

- Avoid overloading network
- Sender estimates network congestion from "loss indications"
- Sender calculates congestion window size (cwnd)
- Sender keeps Send Window size less than a maximum cwnd value
- Sender sets W = min (cwnd, rwnd)

TCP Congestion Control

- end-to-end control (no network assistance)
- Sender limits transmission

LastByteSent-LastByteAcked ≤ cwnd

Throughput ≤ **cwnd**/RTT bytes/sec

Note: For now consider rwnd to be very large such that the send window size is always set equal to cwnd

TCP Congestion Control

- How does sender estimate network congestion?
 - Packet loss is considered as an indication of network congestion
 - Time Out
 - Duplicate Acks
 - TCP sender reduces cwnd after a loss event

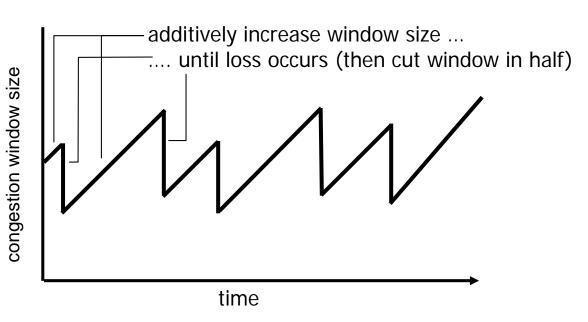
- How does sender determine cwnd size?
 - Sender adjusts existing cwnd according to the loss events
 - AIMD (Additive Increase Multiplicative Decrease)

Additive Increase Multiplicative Decrease

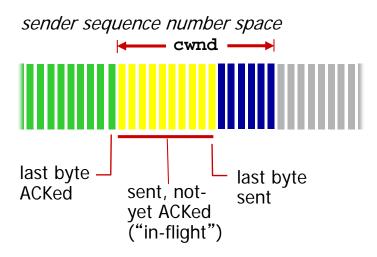
- approach: sender increases transmission rate (window size), probing for usable bandwidth, until loss occurs
 - additive increase: increase cwnd by I MSS every RTT until loss detected
 - multiplicative decrease: cut cwnd in half after loss

AIMD saw tooth behavior: probing for bandwidth

cwnd: TCP sender



TCP Congestion Control: details



sender limits transmission:

$$\begin{array}{ccc} {\tt LastByteSent-} & \leq & {\tt cwnd} \\ {\tt LastByteAcked} & \end{array}$$

 cwnd is dynamic, function of perceived network congestion

TCP sending rate:

 roughly: send cwnd bytes, wait RTT for ACKS, then send more bytes

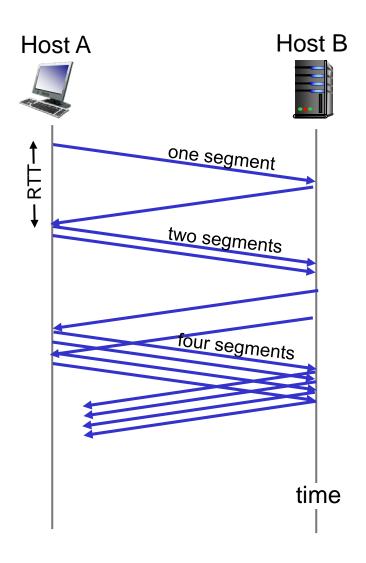
rate
$$\approx \frac{\text{cwnd}}{\text{RTT}}$$
 bytes/sec

At the beginning?

- TCP Slow Start:
 - Probing for usable bandwidth
 - When connection begins, cwnd = 1 MSS
 - Example: MSS = 500 bytes & RTT = 200 msec
 - Initial rate = 2500 bytes/sec = 20 kbps
 - Available bandwidth may be >> MSS/RTT
 - Desirable to quickly ramp up to a higher rate

TCP Slow Start

- when connection begins, increase rate exponentially until first loss event:
 - initially cwnd = I MSS
 - double cwnd every RTT
 - done by incrementing cwnd for every ACK received
- summary: initial rate is slow but ramps up exponentially fast



TCP: detecting, reacting to loss

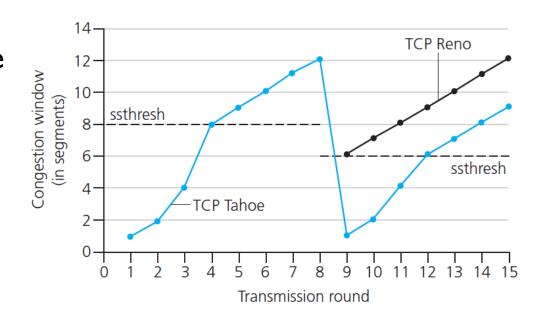
- loss indicated by timeout:
 - cwnd set to I MSS;
 - window then grows exponentially (as in slow start) to threshold, then grows linearly
- loss indicated by 3 duplicate ACKs: TCP RENO
 - dup ACKs indicate network capable of delivering some segments
 - cwnd is cut in half window then grows linearly
- TCP Tahoe always sets cwnd to I (timeout or 3 duplicate acks)

TCP: switching from slow start to CA

Q: when should the exponential increase switch to linear?

A: when cwnd gets to 1/2 of its value before timeout.

This is called slow start threshold (ssthreshold)



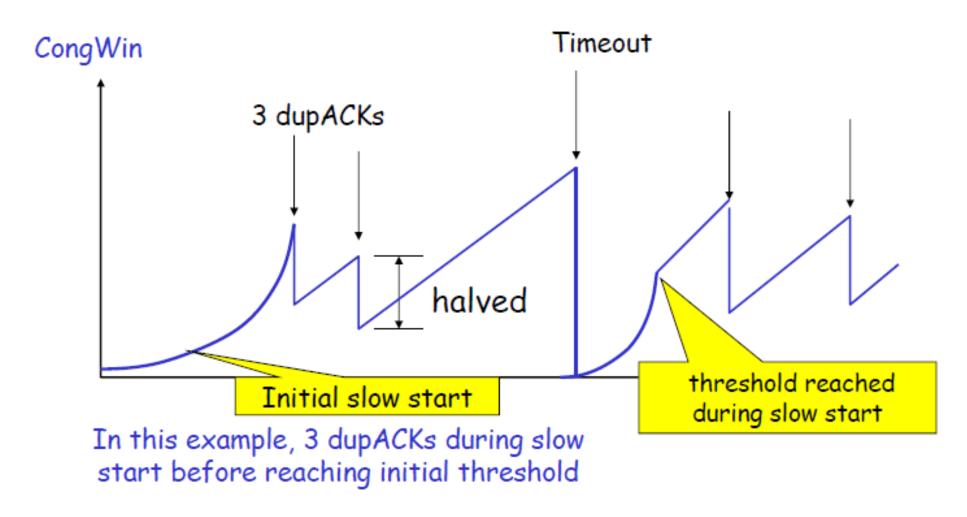
Note: For simplicity, CongWin is in number of segments in the above graph.

^{*} Check out the online interactive exercises for more examples: http://gaia.cs.umass.edu/kurose_ross/interactive/

Threshold

- for initial slow start, threshold is set to a large value
 - assume no threshold until the first loss event
- at a loss event, threshold is set to 1/2 of cwnd just before loss event
- subsequently, threshold is variable

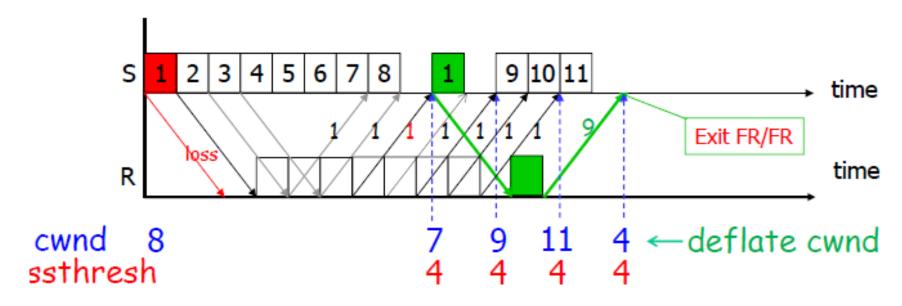
TCP Reno (example scenario)



Summary (TCP Reno)

- When cwnd is below Threshold, sender in slow-start phase, window grows exponentially (until loss event or exceeding threshold).
- When cwnd is above Threshold, sender is in congestionavoidance phase, window grows linearly.
- When timeout occurs, Threshold set to cwnd/2 and cwnd is set to I MSS.
- When a triple duplicate ACK occurs, Threshold set to cwnd/2 and cwnd set to Threshold (also fast retransmit happens).

Fast Recovery entry and exit



- Above scenario: Packet 1 is lost, packets 2, 3, and 4 are received; 3 dupACKs with seq. no. 1 returned
- Fast retransmit
 - Retransmit packet 1 upon 3 dupACKs
- Fast recovery (in steps)
 - Inflate cwnd with #dupACKs such that new packets 9, 10, and 11 can be sent while repairing loss

Summary: TCP Congestion Control

