Washington University in St. Louis

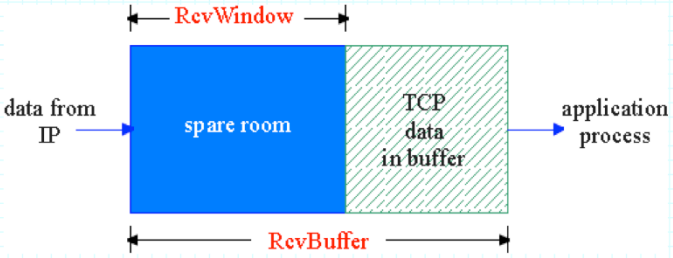
Engineering

13. TCP Flow Control and Congestion Control

- TCP Flow Control
- Congestion control – general principles
- TCP congestion control

Jon Turner – slides adapted from Kurose and Ross

TCP Flow Control



- Receive side of TCP connection has a receive buffer
- If receiver's application does not read data fast enough, the buffer may fill up
- The TCP flow control mechanism prevents the sender from sending more data than receiver can store
  - » essentially a speed-matching service that prevents the sender from going too fast for the receiver

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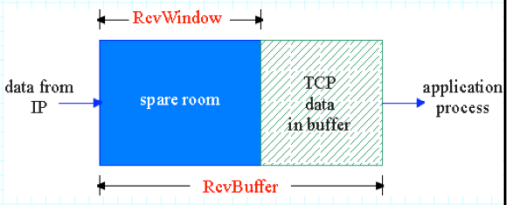
How TCP Flow Control Works

- To simplify discussion, pretend TCP receiver discards out-of-order segments
- Spare room in buffer




```
RcvWindow = RcvBuffer – (LastByteRcvd – LastByteRead)
```

- Receiver sends the value of RcvWindow in the value of `RcvWindow` field of each TCP segment
- Sender never allows the number of unacknowledged bytes to exceed `RcvWindow`
  - » guarantees receive buffer doesn't overflow
- To avoid deadlock, sender continues to send one byte segments when *RcvWindow*=0
  - » this ensures that sender is informed when *RcvWindow*>0

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Exercises

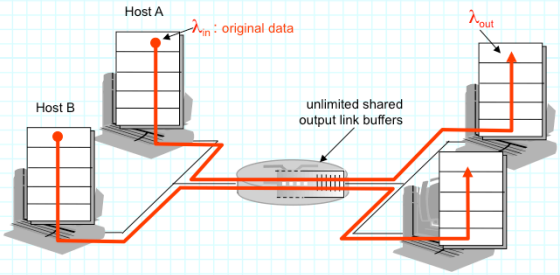
1. Suppose host *A* is sending data to host *B* using TCP. Assume that the DSL link leading to *B* has a rate of 2 Mb/s and that this is the "bottleneck" in the connection. Assume that *A* has 500 KB to send and that *B*'s receive buffer can only hold 20 KB. Also, assume that for the first three seconds, the application at *B* does not read any data from the connection, but that after that it reads data at the rate of 100 KB/s. Draw a chart showing the amount of data in the receive buffer at *B* as a function of time. Label the points on the curve where either *A*'s sending rate or *B*'s reading rate changes.
2. In the scenario from the previous question, how many single byte segments does *A* send, if the RTT is 100 ms.

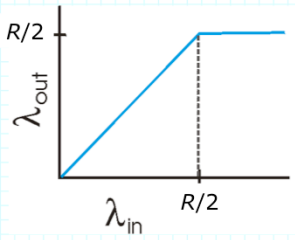
Principles of Congestion Control

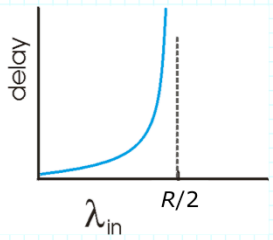
- What is meant by congestion?
  - » this is when the amount of traffic arriving at a network link exceeds the link rate for an "excessive time period"
  - » caused by sources sending too much data for the *link* to handle
    - note that it's the network that is limiting the traffic flow in this case, not the receiver
  - » can cause router queues to fill up and overflow
- Consequences
  - » packets get lost at routers
  - » network delays get large
  - » network throughput can actually drop as load increases
    - this happens because packets may be dropped after passing through several routers, wasting the capacity of "upstream" links
    - since some network effort is wasted, throughput drops below peak

Congestion Scenario 1

- two senders, two receivers
- one router, infinite buffers
- no retransmission







- large delays when congested
- achieve max possible throughput

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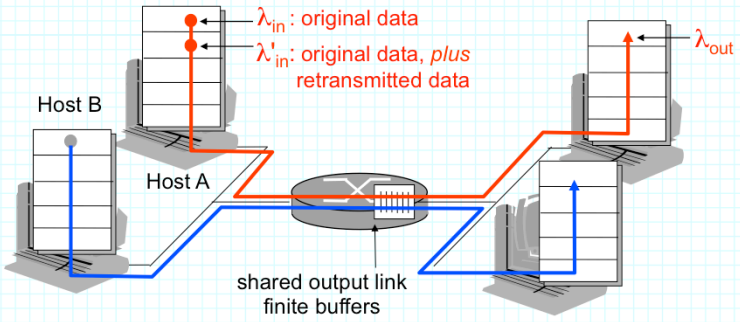
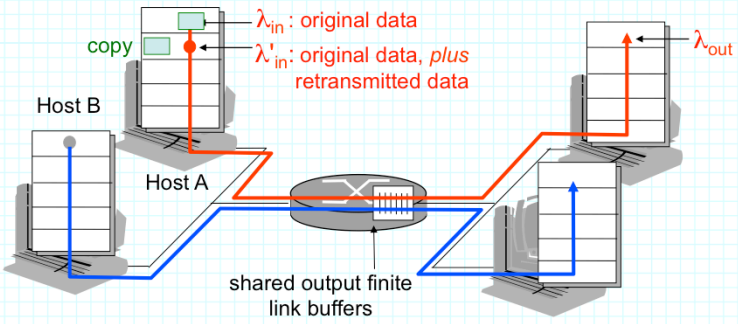
Congestion Scenario 2

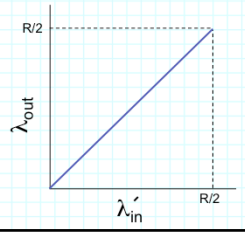


- One router, *finite* buffers
- Sender retransmission of timed-out packet
  - » application-layer input = application-layer output: $\lambda_{in} = \lambda_{out}$
  - » transport-layer input includes *retransmissions*: $\lambda_{in} \geq \lambda_{in}$

Congestion Scenario 2a



- Sender sends only when router buffers available



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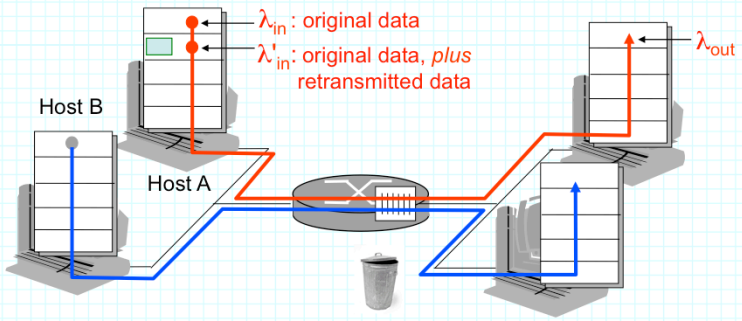
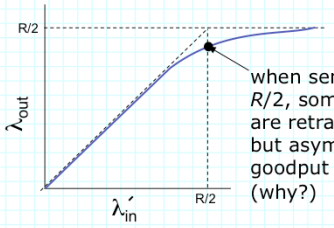
Congestion Scenario 2b

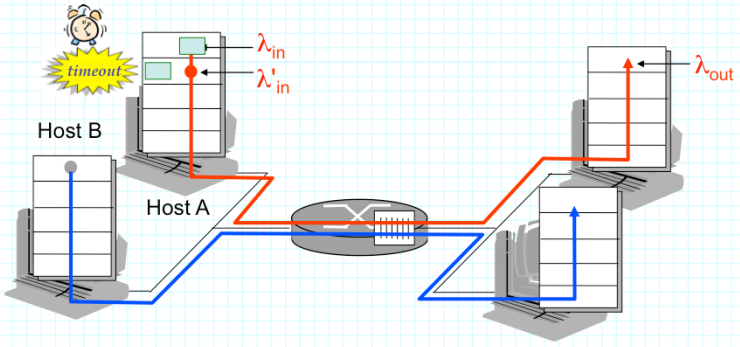


- Packets may get dropped at router due to full buffers
- Sender only resends if packet *known* to be lost
  - » no lost acks
  - » no premature timeouts

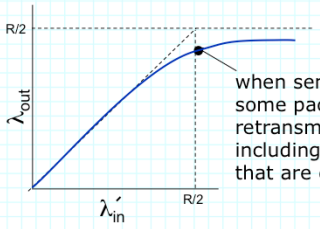


when sending at *R*/2, some packets are retransmissions but asymptotic goodput is still *R*/2 (why?)

Congestion Scenario 2c



- Packets may get dropped at router due to full buffers
- Sender times out prematurely, sending *two* copies, both of which are delivered



when sending at *R*/2, some packets are retransmissions including duplicated that are delivered!

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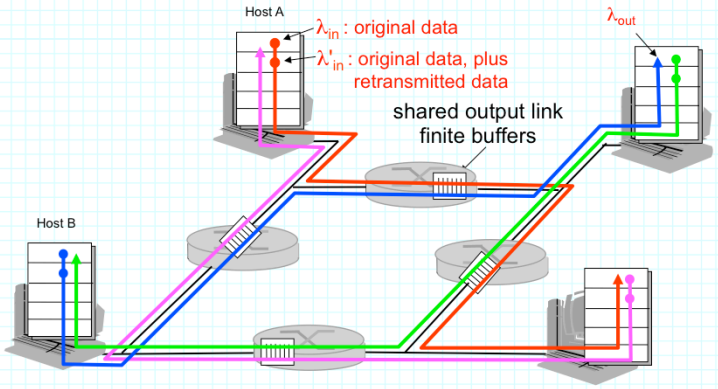
Congestion Scenario 3



- Four senders
- Multihop paths
- Timeout/retransmit

Q: What happens as λ'_{in} grows and exceeds $R/2$?

Congestion Control Options

- Two major approaches to congestion control
- End-to-end congestion control – used by TCP
  - » no explicit feedback from network
  - » congestion inferred from loss, delay observed by hosts
  - » relies on cooperation of end hosts
- Network-assisted congestion control
  - » routers provide feedback to end systems
    - more rapid response to traffic changes than end-to-end approach
  - » simplest approach – single bit congestion indication
    - TCP and IP support this, but capability is usually disabled
  - » explicit rate control
    - senders request a sending rate, routers decide allowable rates
    - can prevent "greedy hosts" from hogging network capacity
    - used in Asynchronous Transfer Mode (ATM) networks

Exercises

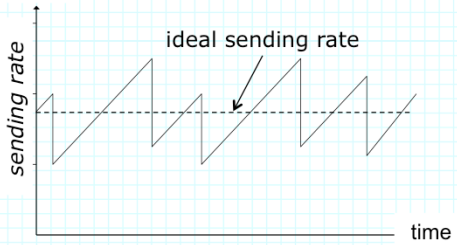
1. When TCP detects that a packet has been lost, it assumes that the loss was caused by congestion. Under what circumstances is this a reasonable assumption? In what common situation is it not reasonable? How does TCP's assumption affect performance when packets may be lost for other reasons?

TCP Congestion Control Big Picture

- Objective: send as fast as possible, but not too fast
  - » when lost segments are detected, sender reduces its rate
  - » when there are no lost segments, sender increases its rate
- Key questions
  - » when it's time to cut rate, by how much should it be cut?
    - TCP cuts sending rate in half in order to reduce congestion quickly
  - » when it's time to increase rate, by how much should it increase?
    - TCP makes small incremental increases to avoid going right back into congestion
    - but TCP allows a "new sender" to increase its rate more quickly
- Additive increase/multiplicative decrease + "slow-start"
  - » during stable traffic periods, rates oscillate around ideal rates
  - » different end-to-end flows get roughly "fair shares" of capacity
  - » can be slow to respond to traffic changes

Basic AIMD Behavior

- Sawtooth pattern
  - » sending rate oscillates around "ideal rate"
  - » when ideal rate is large, the "cycle time" can also be large
  - » implies slow response time



- Trends towards "fairness"
  - » when several TCP connections share a common "bottleneck" link, it's desirable that they each receive a roughly equal share
  - » AIMD makes things approximately equal in the long run
    - connections tend to oscillate in sync with each other, so when total rate is too large for link, all halve their rates together
    - this has bigger impact on higher rate senders
    - caveat: connections with short RTTs get more capacity

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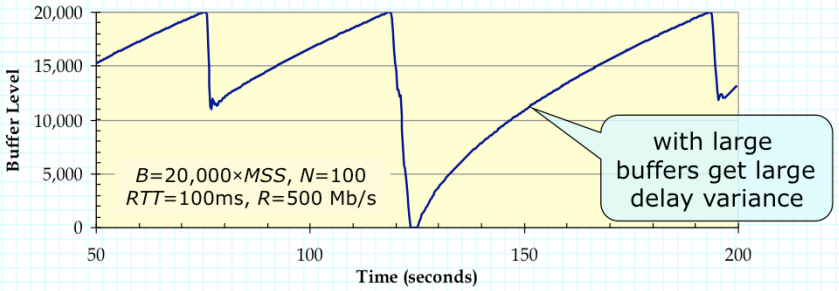
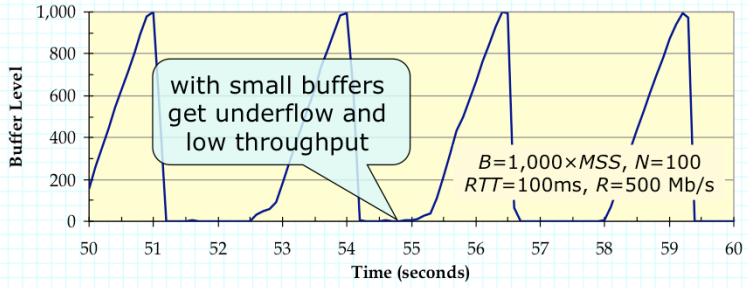
Simulation for Large and Small Buffers





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Exercises

1. Suppose that *N* TCP connections pass through a "bottleneck link" that has a rate of 1 Gb/s and a buffer capacity of 25 MB. Assume that for all connections, the RTT is 100 ms. Suppose that just before the buffer fills, the input rate is 1.2 Gb/s. Assuming that this causes all of the TCP senders to halve their sending rate, how much will the buffer level drop during the next 2 RTTs?

TCP Congestion Control Variants

- A series of congestion control algorithms have been developed and used for TCP
  - » the differences affect only the sender-side of a TCP connection, so hosts running different versions of TCP can still communicate
- TCP Tahoe
  - » the original approach developed in the late 1980s
  - » basic AIMD + slow-start strategy
- TCP Reno and New Reno
  - » New Reno is now most widely deployed approach
  - » added a transient "fast recovery" operating mode to TCP
- BIC and CUBIC
  - » provides faster congestion response in high speed networks
  - » CUBIC is now the default choice in Linux

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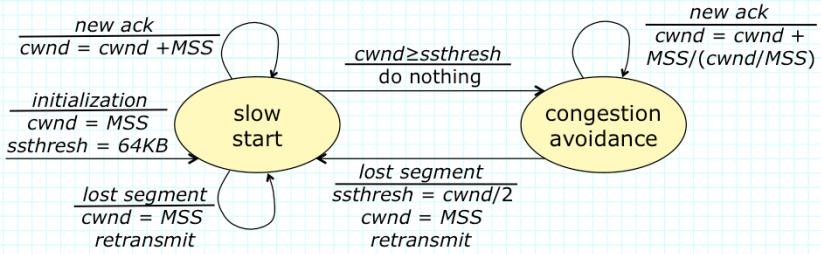
TCP Tahoe Overview

- TCP sender has two primary operating "states"
  - » congestion avoidance
    - increase sending rate in small increments
  - » slow start
    - allows more rapid increase in rates for new senders
    - also entered after a packet loss is detected
- Sender maintains two variables to control congestion
  - » the *congestion window* variable (*cwnd*) limits number of unacknowledged bytes
  - » the *slow start threshold* (*ssthresh*) controls when sender leaves the slow-start state
  - » variables are updated in response to lost packets and reception of ACKs

TCP Tahoe Details



- Updating *cwnd*
  - in slow start, *cwnd* is effectively doubled each RTT (if no loss)
  - in congestion avoidance, *cwnd* grows by about 1 MSS per RTT
- After transition from congestion avoidance to slow start
  - it takes about 1 RTT for a new ACK to arrive and nothing much happens during this period
  - after ACK arrives, # of unACK-ed bytes becomes 0, sender can resume sending, and *cwnd* grows as ACKs arrive

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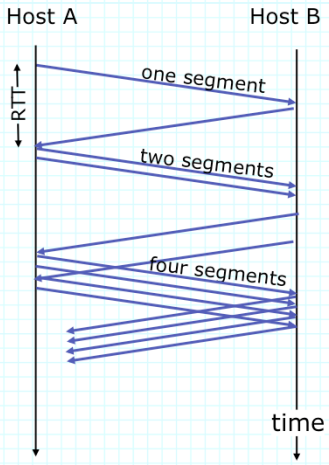
Understanding Slow Start

- A "new" source starts with a small window, but is allowed to increase it quickly
  - » initially *cwnd* = 1 MSS
  - » *cwnd* is effectively doubled for every RTT with no packet loss
  - » after first packet loss, sender halves *cwnd* and reverts to additive increase
- "Slow-start" something of a misnomer, since allows fast increase in rate



Exercises

1. Suppose that a TCP Tahoe connection in the congestion avoidance state has a *cwnd* value of 50 KB, an MSS of 1 KB and an RTT of 100 ms. Suppose that at this point, it detects a lost packet. How does this change the value of *cwnd* and *ssthresh*? Approximately how much time passes before the sender goes back into the congestion avoidance state? Assuming that no more packets are lost until *cwnd* exceeds 50 KB again, approximately how much time is spent in the congestion avoidance state? For this connection, does slow-start have a big impact on the throughput achieved?

TCP Reno

- Two ways to detect packet loss
  - » timeout or three duplicate ACKs
- TCP Reno treats these cases differently
  - » because timeout is an indication of more severe congestion
    - typically, many packets must be lost to trigger timeout
  - » on timeout, Reno behaves like Tahoe (goes to slow start)
  - » on triple-dup-ACK, it goes to a new *fast recovery* state
    - sets *ssthresh* = $cwnd_0/2$ and *cwnd* = *ssthresh*+3*MSS
- Fast recovery is a special transient state that is typically active during the first RTT after a triple-dup-ACK
  - » when the lost packet is ACK-ed (after about one RTT), sender goes back to congestion avoidance with *cwnd* = *ssthresh*
  - » before then, each dup-ACK increases cwnd by MSS
    - this allows sender to send ($cwnd_0/2$ bytes during this RTT)

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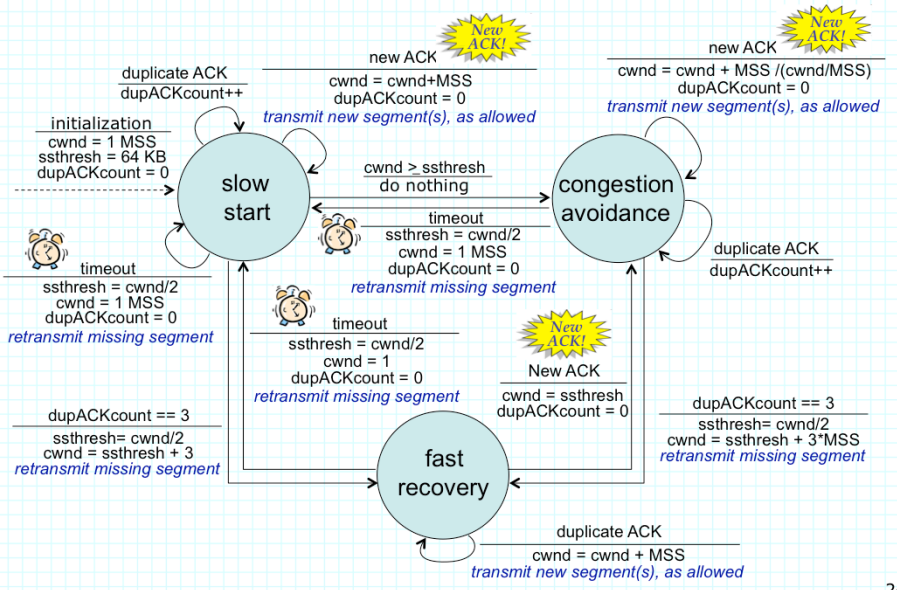
Putting it All Together



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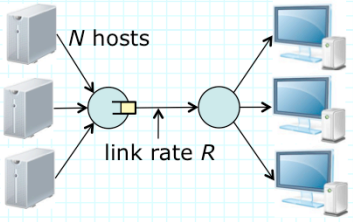
Understanding TCP Performance



- TCP seeks to keep the link busy while limiting congestion
  - » if link queue is large enough, per host throughput $T=R/N$
  - » for small queues, $T \approx .75\ R/N$
- The "cycle time" of TCP's control algorithm is approximately $(1+(R/1.5 \cdot N)(RTT/8 \cdot MSS))RTT$
  - » where R is the link rate and N is the number of flows
  - » note, the cycle time scales up with link rate
    - so, as links get faster, TCP reacts more slowly to changes in traffic
    - example: R=1Gb/s, N=10, RTT=.1s, $MSS=10^4$, cycle time≈8.5s
  - » also note that 1 packet is lost per cycle and number sent per cycle is (cycle time)$(R/N \cdot 8MSS)$=(cycle time)$\cdot T/(8 \cdot MSS)$
    - so losses occur less often as cycle time increases

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TCP Throughput Approximation

- The throughput of a TCP connection can be approximated by

$$T \approx \frac{1.22 \cdot MSS}{RTT\sqrt{L}} \quad \text{or equivalently} \quad L \approx \left(\frac{1.22 \cdot MSS}{T \cdot RTT}\right)^2$$

  - where L is the fraction of packets that are lost in transit
- If packet losses only due to TCP-induced buffer overflow
  - can derive expression using fact that loss rate is 1/(# of packets sent per cycle)
- If only losses are due to bit errors
  - can derive expression using fact that TCP goes through one cycle every 1/L packets, halves its rate at start of each cycle plus fact that average # of packets sent per RTT is $RTT \cdot (T/MSS)$

Fairness in the Internet

- TCP attempts to share available bandwidth "fairly"
  - » operates at the level of TCP connections or "flows", not at the level of application sessions or users
- But easy for "greedy" applications/users to get an "unfair share"
  - » use multiple TCP connections for a given application session
    - web servers commonly do this
  - » use UDP, which has no congestion control
    - many multimedia applications do this
- No clear solution
  - » host-based mechanisms must rely on well-behaved users
  - » internet lacks mechanisms for enforcement of fair usage
  - » potential solutions involve usage-based charging which is unpopular

Exercises

1. Suppose that a TCP Reno connection in the congestion avoidance state has a *cwnd* value of 50 KB, an MSS of 1 KB and an RTT of 100 ms. Suppose that at this point, it detects a lost packet (by duplicate ack). How does this change the value of *cwnd* and *ssthresh*? Approximately how much time passes before the sender goes back into the congestion avoidance state? Assuming that no more packets are lost until *cwnd* exceeds 50 KB again, approximately how much time is spent in the congestion avoidance state?
2. Consider a TCP Reno connection that is achieving a throughput of 40 Mb/s. Assume that the MSS is 1 KB and the RTT is 100 ms. Estimate the loss rate for this connection.
3. Consider a TCP Reno connection that is experiencing a packet loss rate of 4%. Assume that the MSS is 1 KB and the RTT is 100 ms. Estimate the throughput of this connection.