Distributed Computer Systems Engineering

CIS 508: Lecture 9
Networking Protocols I

Lecturer: Sid C.K. Chau Email: ckchau@masdar.ac.ae



Distributed Resource Allocation

- Distributed systems often share resource
 - E.g. network capacity, computation, storage
- Not all processes are aware of the state of resource sharing
 - Communication is often needed between resource providers and resource customers
 - There may be delay between usage congestion (or idleness) and the awareness of resource customers
- Key example: sharing bandwidth over the Internet
- End-users need to adapt to the dynamic resource availability by changing data rate
- TCP is a congestion control algorithm that allows distributed resource allocation over network capacity

Toy Problem: El Farol Bar

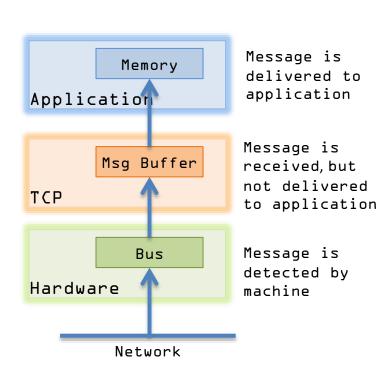


- At the weekend, people like to go to El Farol Bar to chill out
- El Farol Bar is finite, and can only accommodate limited number of people
 - Too many people: overcrowded
 - Too few people: not fun
 - Should you go to El Farol Bar or not?
- Every person makes his/her decision in a distributed manner
- A distributed congestion control problem

TCP (Transport Control Protocol)

- Point-to-point
 - One sender, one receiver
- Reliable end-to-end transmission
 - Overcome unreliable link-to-link
- Full duplex data
 - Bi-directional data flow in same connection
 - MSS: maximum segment size
- Connection-oriented
 - Handshaking (exchange of control messages) initiate the session states of sender & receiver before data exchange
- Flow controlled
 - Sender will not overwhelm receiver

TCP Flow Control



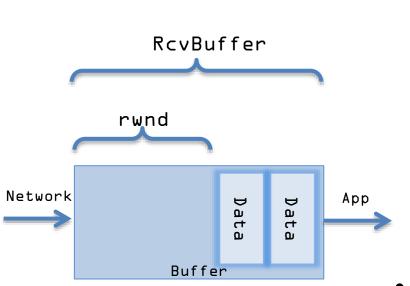
Flow Control

- Receiver controls sender
- Sender will not overflow receiver's buffer by transmitting too much and too fast

TCP Packet Structure

```
32 bits
  URG: urgent data
                                                            counting
                      source port#
                                         dest port#
(generally not used)
                                                            by bytes
                                                            of data
                            sequence number
                                                            (not segments!)
        ACK: ACK #
             valid
                         acknowledgement number
                      len used UAPRS
                      head not
                                            rwnd
 PSH: push data now
(generally not used)
                                                               Receiver
                          checksum
                                       Urg data pointr
                                                               window
                                                                # bytes
       RST, SYN, FIN:
                       options(variable length)
                                                               receiver
   connection estab
                                                               willing
    (setup, teardown
                                                               to accept
          commands)
                             application
                             data
          Internet
                             (variable length)
          checksum
       (as in UDP)
```

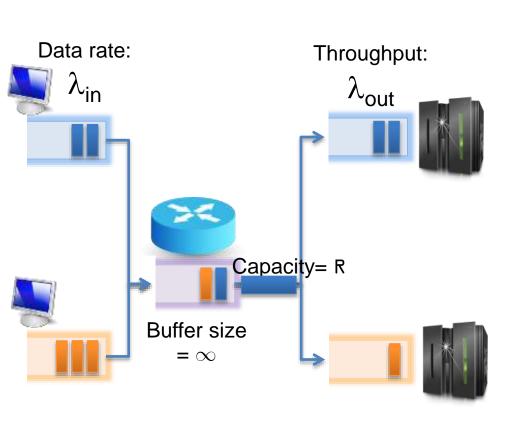
TCP Flow Control



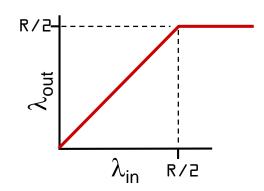
- Receiver advertises free buffer space by including rwnd value in TCP header of receiver-to-sender segments
 - RcvBuffer size set via socket options (default is 4096 bytes)
 - many operating systems autoadjust RcvBuffer
- Sender limits amount of unacknowledged ("in-flight") data to receiver's rwnd value
- Guarantee receiver buffer will not overflow

Principles of Congestion Control

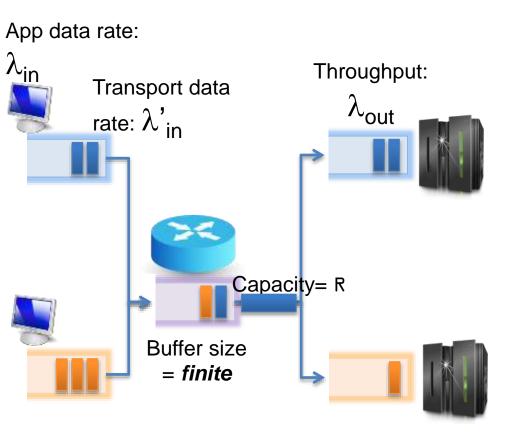
- Congestion
 - Basic idea
 - "Too many sources sending too much data too fast for the network to handle"
 - Different from flow control
 - "One sender sending too much data too fast for the receiver to handle"
 - Problems
 - Lost packets (buffer overflow at routers)
 - Long delays (queueing in router buffers)



- Two senders, two receivers
- One router, infinite buffers
- Output link capacity R
- No retransmission

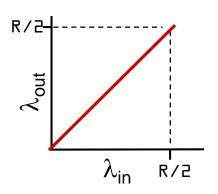


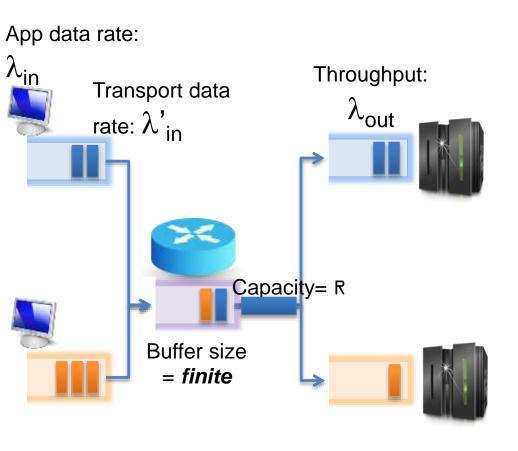
 maximum per-connection throughput R/2



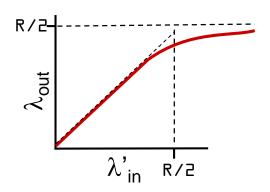
- One router, finite buffers
- Packets may be dropped
- Sender retransmits packets
- Transport data rate (including retransmission) is larger than application data rate

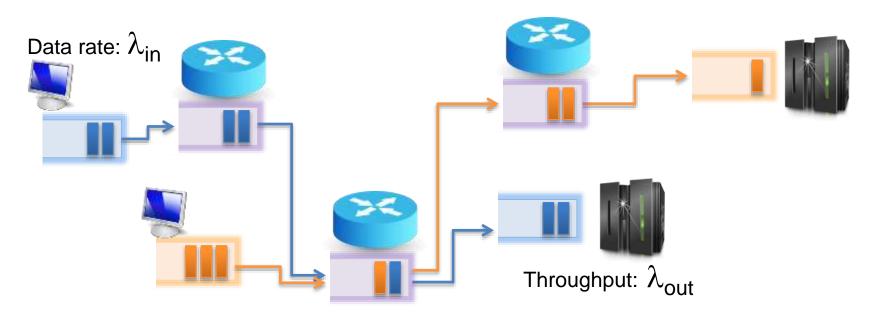
$$\lambda'_{in} \geq \lambda_{in}$$



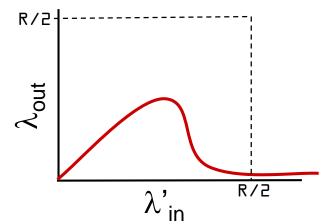


- $\lambda'_{in} \geq \lambda_{in}$
- Ideally, sender only resends if packet known to be lost
- When sending at R/2, some packets are retransmissions but asymptotic throughput is still R/2



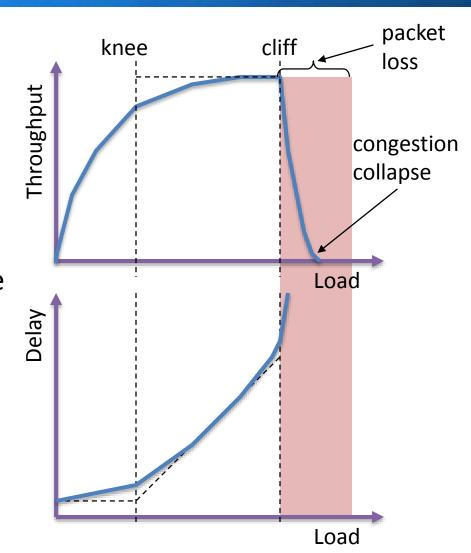


- Some intermediate links are shared
- When packet dropped, upstream transmission capacity used for that packet was wasted



Congestion Collapse

- Knee point after which
 - Throughput increases very slow
 - Delay increases fast
- Cliff point after which
 - Throughput starts to decrease very fast to zero (congestion collapse)
 - Delay approaches infinity
- In queueing theory
 - Delay = 1/(1 utilization)



Congestion Control Approaches

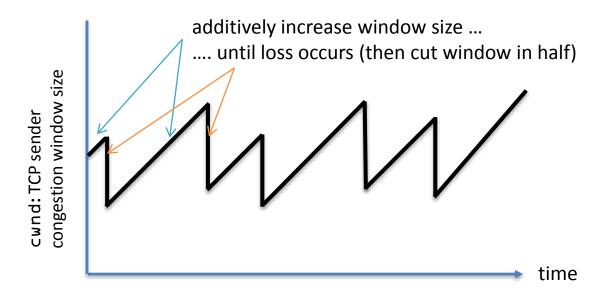
Two approaches:

- 1. End-to-end congestion control
 - No explicit feedback from network
 - Congestion inferred from end-system observed loss, delay
 - Approach taken by TCP
- 2. Network-assisted congestion control
 - Routers provide feedback to end systems
 - Single bit indicating congestion (SNA, DECbit, TCP/IP ECN, ATM)
 - Explicit rate for sender to send

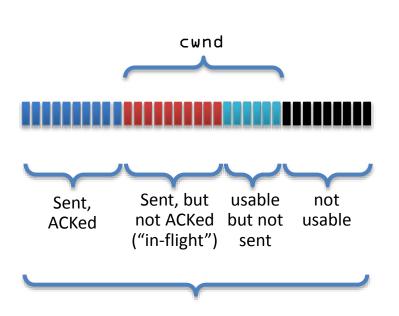
TCP Congestion Control

- Sender increases transmission rate (window size), probing for usable bandwidth, until loss occurs
 - Additive increase: increase cwnd by 1 every RTT until loss detected
 - Multiplicative decrease: cut cwnd in half after loss

AIMD saw tooth behavior: probing for bandwidth



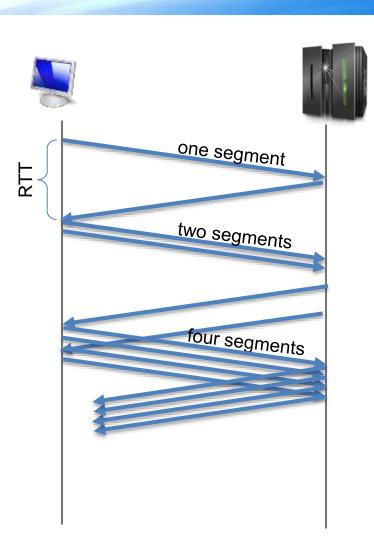
TCP Congestion Control



Sender sequence number space

- Sender limits transmission
- LastByteSent LastByteAcked < cwnd
- cwnd is dynamic, function of perceived network congestion
- TCP sending rate: send cwnd bytes, wait RTT for ACKS, then send more bytes (cwnd/RTT bytes/sec)

TCP Slow Start

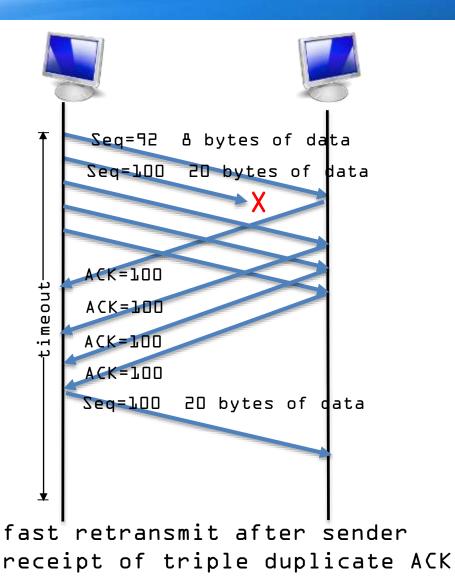


- When connection begins, increase rate exponentially until first loss event
 - Initially cwnd = 1
 - Double cwnd every RTT
 - Done by incrementing cwnd for every ACK received
- Basic idea: initial rate is slow but ramps up exponentially fast

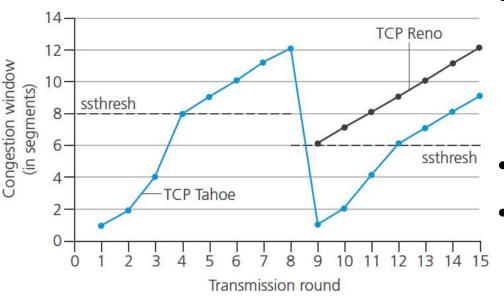
Reaction to Loss

- Loss indicated by timeout:
 - cwnd set to 1
 - window then grows exponentially (as in slow start) to threshold, then grows linearly
- Loss indicated by 3 duplicate ACKs (TCP RENO)
 - Duplicate ACKs indicate network capable of delivering some segments
 - cwnd is cut in half window then grows linearly
- TCP Tahoe always sets cwnd to 1 (timeout or 3 duplicate acks)

TCP: Fast Retransmit

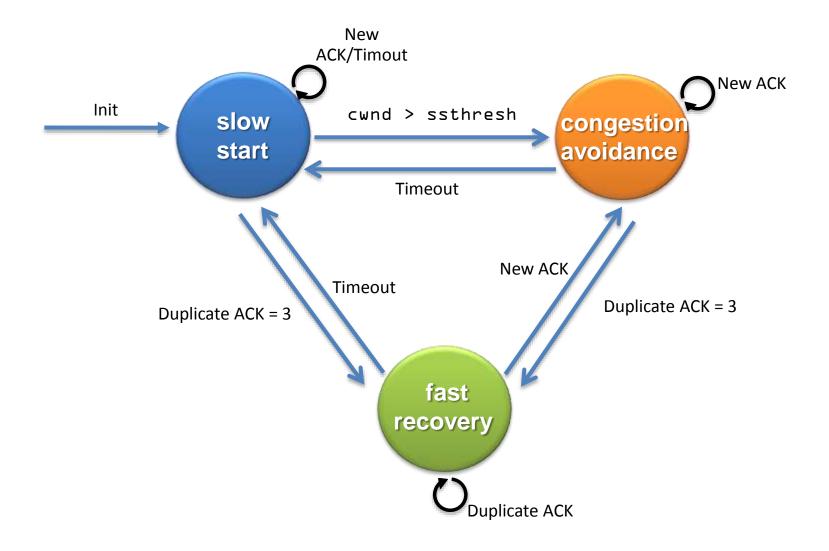


Slow Start to Congestion Avoidance



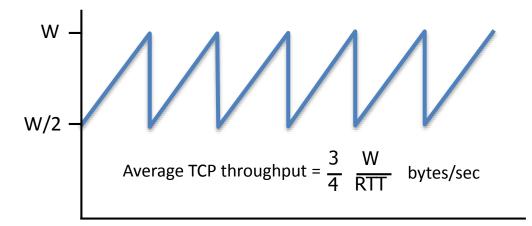
- Exponential increase switch to linear increase
 - when cwnd gets to 1/2 of its value before timeout
- Variable ssthresh
 - On loss event, ssthresh is set to 1/2 of cwnd just before loss event

TCP Congestion Control: Simplified

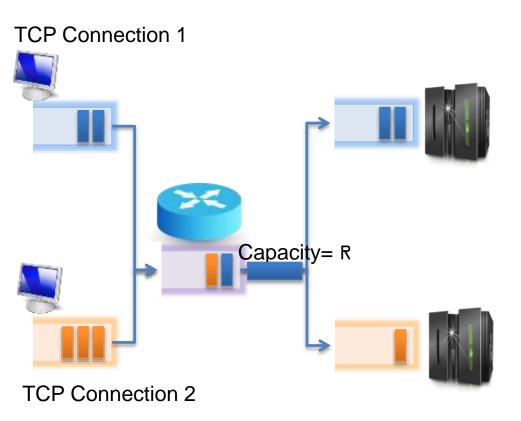


TCP Throughput

- Average TCP throughput as function of window size RTT
 - Ignore slow start, assume always data to send
- Window size W (measured in bytes) where loss occurs
 - Average window size (# in-flight bytes) is ¾ W
 - Average throughput is ¾ W per RTT

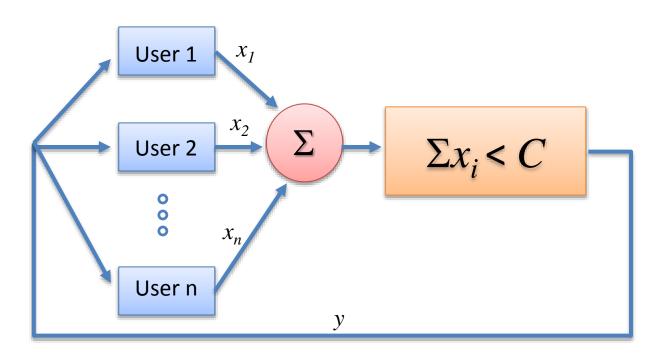


TCP Fairness



- Fairness goal:
 - If K TCP sessions share same bottleneck link of bandwidth R
 - Each should have average rate of R/K

System Control Model



- Simple, yet powerful model
- Explicit binary signal of congestion

Possible Strategies

$$x_i(t+1) = \begin{cases} a_I + b_I x_i(t) & \text{Increase} \\ a_D + b_D x_i(t) & \text{Decrease} \end{cases}$$

- Multiplicative increase, additive decrease
 - $lack a_I=0$, $b_I>1$, $a_D<0$, $b_D=1$
- Additive increase, additive decrease

•
$$a_I > 0$$
 , $b_I = 1$, $a_D < 0$, $b_D = 1$

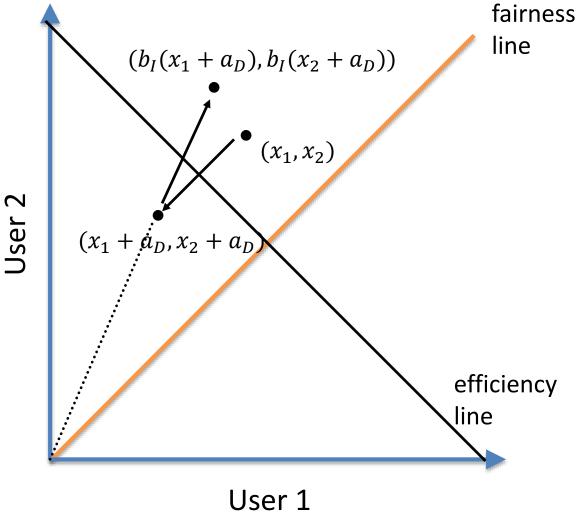
Multiplicative increase, multiplicative decrease

$$lack a_I=0$$
 , $b_I>1$, $a_D=0$, $b_D<1$

- Additive increase, multiplicative decrease
 - $a_I > 0$, $b_I = 1$, $a_D = 0$, $b_D < 1$
- Which one?

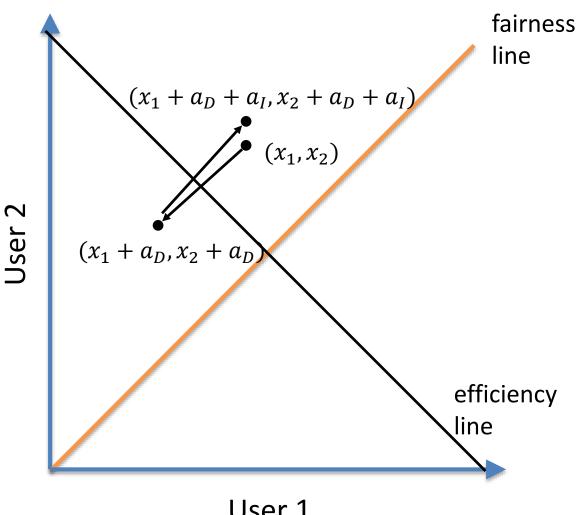
Multiplicative Increase, Additive Decrease

- Fixed point at $x_1 = x_2 = \frac{b_I a_D}{1 - b_I}$
- But fixed point is unstable



Additive Increase, Additive Decrease

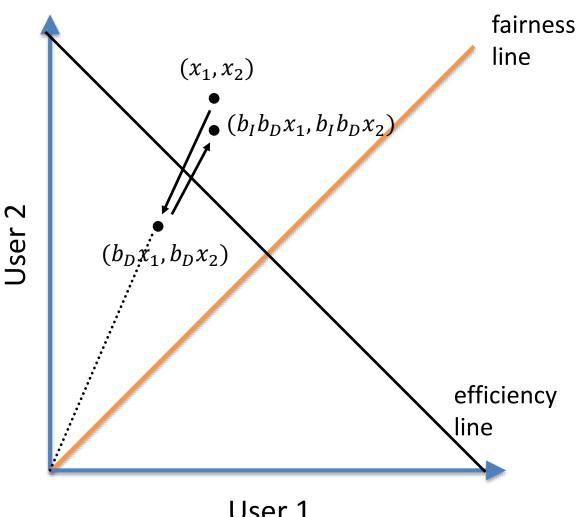
 Reaches stable cycle, but does not converge to fairness



User 1

Multiplicative Increase, Multiplicative Decrease

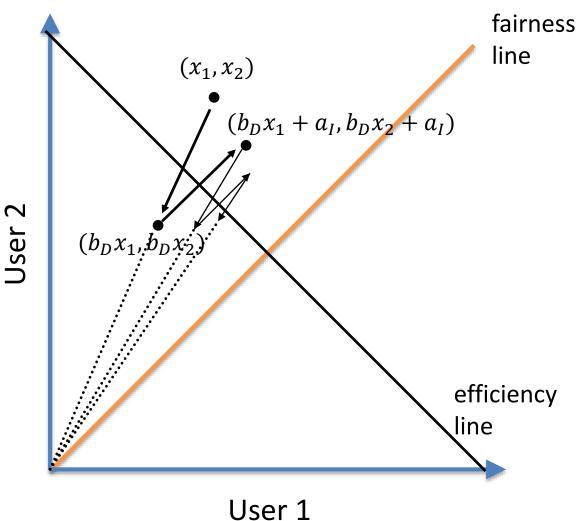
 Converges to stable cycle, but is not fair



User 1

Additive Increase, Multiplicative Decrease

 Converges to stable and fair cycle



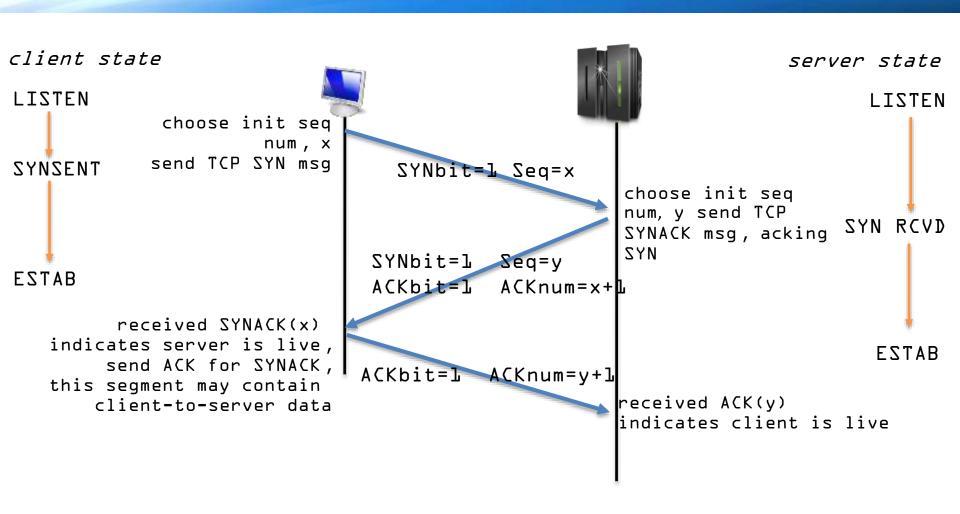
Extensions to TCP

- Selective acknowledgements: TCP SACK
- Explicit congestion notification: ECN
- Delay-based congestion avoidance: TCP Vegas
- Discriminating between congestion losses and other losses: cross-layer signaling and guesses
- Randomized drops (RED) and other router mechanisms
- Other Issues with TCP
 - Fairness: Throughput depends on RTT
 - High speeds: To reach 10gbps, packet losses occur every 90 minutes!
 - Selfish flows: Can get all the bandwidth they like

DDoS

- DDoS (distributed denial-of-service attack)
- When multiple systems flood the bandwidth or resources of a targeted system, usually one or more web servers
- These systems are compromised by attackers using a variety of methods
- Aim to reduce the availability of service
- Effects:
 - Loss of revenue
 - Server unable to preform timely operations (control system)
 - Prevent access to information (web server)

TCP: 3-way handshake



Example: SYN Flood

- When a client attempts to start a TCP connection to a server, the client and server exchange a series of messages which normally runs like this:
 - The client requests a connection by sending a SYN (synchronize) message to server
 - The server acknowledges sending SYN-ACK to client
 - The client responds with an ACK, and connection is established
- Attacker initiates a lot of SYN messages to a server
- Immediate abort (no closing connection message) after SYN
- Server left in time-out mode to detect aborted connections



- Textbook
 - Computer Networking (Chap. 3)

 James F. Kurose and Keith W. Ross; Pearson Addison-Wesley

