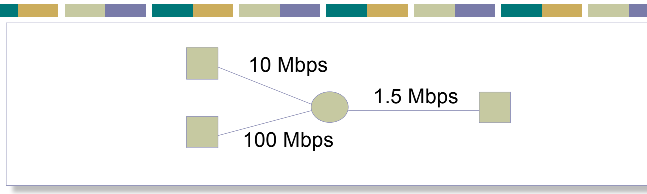
15-744: Computer Networking

L-5 Congestion Control

News

- Note problem set 1 update
  - Part E: Please draw three RTTs after the loss, not just one.
- Assigned reading
  - [JK88] Congestion Avoidance and Control
  - [CJ89] Analysis of the Increase and Decrease Algorithms for Congestion Avoidance in Computer Networks

Congestion

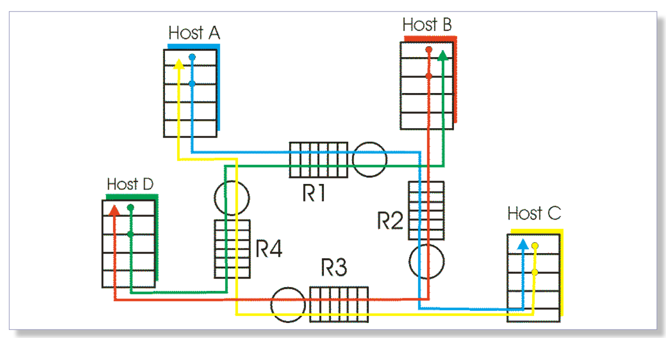


- Different sources compete for resources inside network: Link b/w and queue space
- Why is it a problem?
  - Sources are unaware of current state of resource
  - Sources are unaware of each other
  - In many situations will result in < 1.5 Mbps of throughput (congestion collapse)

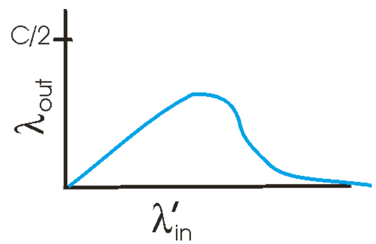
Causes & Costs of Congestion

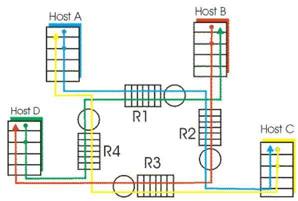
- Four senders – multihop paths
- Timeout/retransmit

Q: What happens as rate increases?



Causes & Costs of Congestion





- When packet dropped, any "upstream transmission capacity used for that packet was wasted!

Congestion Collapse

- Definition: *Increase in network load results in decrease of useful work done*
- Many possible causes
  - Spurious retransmissions of packets still in flight
    - Classical congestion collapse
    - How can this happen with packet conservation
    - Solution: better timers and TCP congestion control
  - Undelivered packets
    - Packets consume resources and are dropped elsewhere in network
    - Solution: congestion control for ALL traffic

Other Congestion Collapse Causes

- Fragments
  - Mismatch of transmission and retransmission units
  - Solutions
    - Make network drop all fragments of a packet (early packet discard in ATM)
    - Do path MTU discovery
- Control traffic
  - Large percentage of traffic is for control
    - Headers, routing messages, DNS, etc.
- Stale or unwanted packets
  - Packets that are delayed on long queues
  - "Push" data that is never used

Congestion Control and Avoidance

- Desirable properties:
  - Scalability:
    - # flows, range of capacities, range of delays
    - Do well in entire range!
  - Efficiency: High network utilization
  - Fairness
  - Works-ness: avoids collapse!
- Congestion collapse is not just a theory
  - Has been frequently observed in many networks

Fairness

- Jain's fairness index
  - $f = (\Sigma x_i)^2/n(\Sigma x_i^2)$
  - All x equal: 1
  - k/n get service: k/n
- Max-min fairness
  - No user receives more than their request, pi
  - No other allocation satisfying (1) has higher min allocation
  - condition 2 holds as we remove the minimal user & reduce total resource accordingly
  - aka: ui = MIN(u fair, pi)
- Goal: Something that works well enough.

Objectives

- Simple router behavior
- Distributedness
- Efficiency: $X_{knee} = \Sigma x_i(t)$
- Fairness: $(\Sigma x_i)^2/n(\Sigma x_i^2)$
- Power: $(\text{throughput}^{\alpha}/\text{delay})$
- Convergence: control system must be stable

Design questions

- Congestion
  - How is congestion signaled?
    - Either mark or drop packets
  - When is a router congested?
    - Drop tail queues – when queue is full
    - Average queue length – at some threshold
- Control questions:
  - How do senders react to congestion?
  - How do senders determine capacity for flow?

Congrol 2

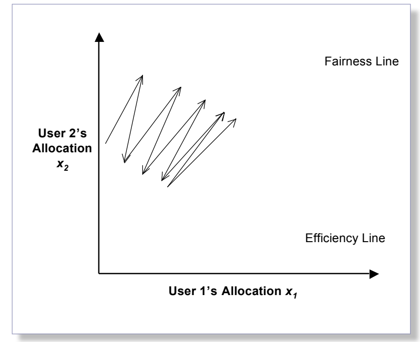
- Upon congestion, flows must reduce rate
- How? Decrease algorithm
- If no congestion, flows might try sending more. Increase algorithm.
- Let's assume window-based flow control
  - sender maintains "cwnd": # of unacknowledged packets in the network at any time
  - Transmission rate: cwnd / rtt
- (Alternate: Rate-based; equation-based)

Linear Control

- Many different possibilities for reaction to congestion and probing
  - Examine simple linear controls
  - Window(t + 1) = a + b Window(t)
  - Different a_i/b_i for increase and a_d/b_d for decrease
- Supports various reaction to signals
  - Increase/decrease additively
  - Increased/decrease multiplicatively
  - Which of the four combinations is optimal?

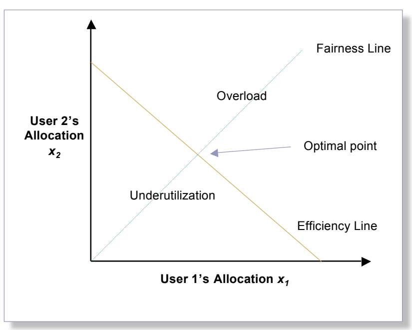
Phase plots

- Simple way to visualize behavior of competing connections over time



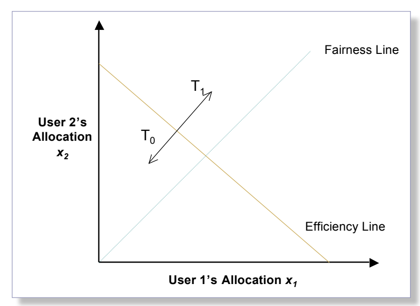
Phase plots

- What are desirable properties?
- What if flows are not equal?



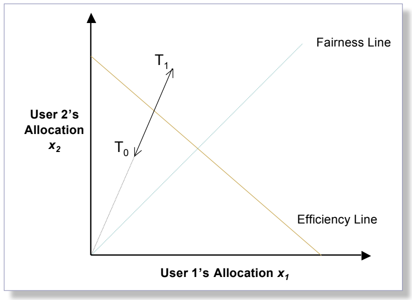
Additive Increase/Decrease

- Both X_1 and X_2 increase/decrease by the same amount over time
  - Additive increase improves fairness and additive decrease reduces fairness



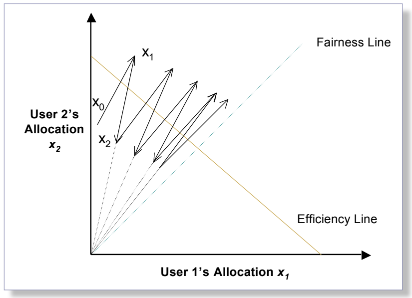
Multiplicative Increase/Decrease

- Both X_1 and X_2 increase by the same factor over time
  - Extension from origin – constant fairness



What is the Right Choice?

- Constraints limit us to AIMD
  - Can have multiplicative term in increase (MAIMD)
  - AIMD moves towards optimal point



TCP and linear controls

- Upon congestion:
  - w(t+1) = a*w(t)   0 < a < 1
- While probing
  - w(t+1) = w(t) + b   0 < b << wmax
- TCP sets a = 1/2, b = 1 (packet)

TCP Congestion Control

- Motivated by ARPANET congestion collapse
- Underlying design principle: packet conservation
  - At equilibrium, inject packet into network only when one is removed
  - Basis for stability of physical systems
- Why was this not working?
  - Connection doesn't reach equilibrium
  - Spurious retransmissions
  - Resource limitations prevent equilibrium

TCP Congestion Control - Solutions

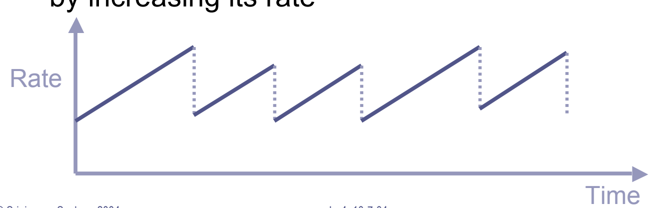
- Reaching equilibrium
  - Slow start
- Eliminates spurious retransmissions
  - Accurate RTO estimation
  - Fast retransmit
- Adapting to resource availability
  - Congestion avoidance

TCP Congestion Control

- Changes to TCP motivated by ARPANET congestion collapse
- Basic principles
  - AIMD
  - Packet conservation
  - Reaching steady state quickly
  - ACK clocking

AIMD: Now you grok the sawtooth

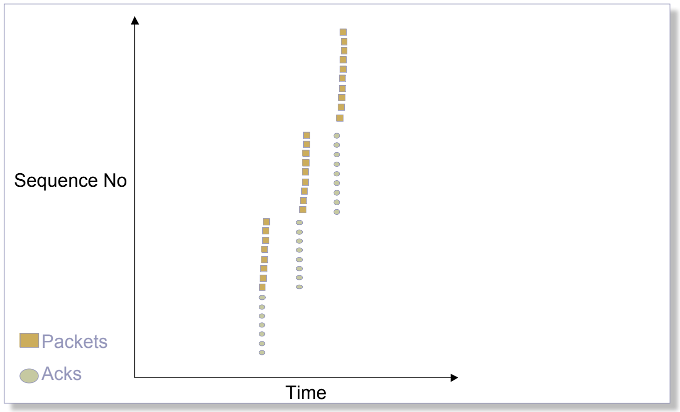
- Distributed, fair and efficient
- Packet loss is seen as sign of congestion and results in a multiplicative rate decrease
  - Factor of 2
- TCP periodically probes for available bandwidth by increasing its rate



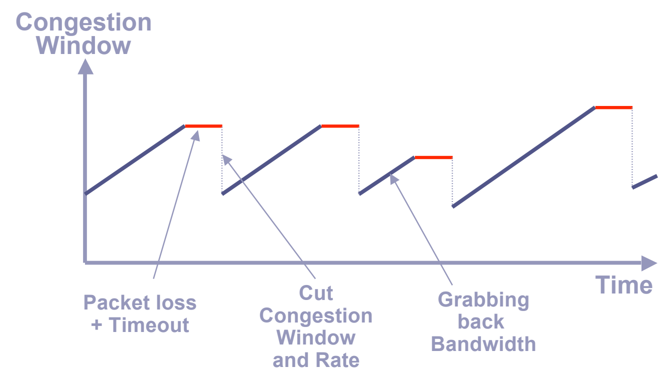
Congestion Avoidance

- If loss occurs when cwnd = W
  - Network can handle 0.5W ~ W segments
  - Set cwnd to 0.5W (multiplicative decrease)
- Upon receiving ACK
  - Increase cwnd by (1 packet)/cwnd
    - What is 1 packet? → 1 MSS worth of bytes
    - After cwnd packets have passed by → approximately increase of 1 MSS
- Implements AIMD

Congestion Avoidance Sequence Plot



Congestion Avoidance Behavior

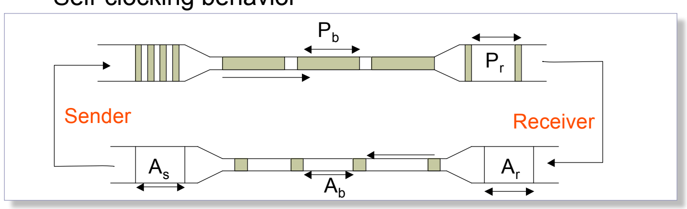


Packet Conservation

- At equilibrium, inject packet into network only when one is removed
  - Sliding window and not rate controlled
  - But still need to avoid sending burst of packets → would overflow links
    - Need to carefully pace out packets (ack clocking)!
    - Helps provide stability
- Need to eliminate spurious retransmissions
  - Accurate RTO estimation
  - Better loss recovery techniques (e.g. fast retransmit)

TCP Packet Pacing

- Congestion window helps to “pace” the transmission of data packets
- In steady state, a packet is sent when an ack is received
  - Data transmission remains smooth, once it is smooth
  - Self-clocking behavior

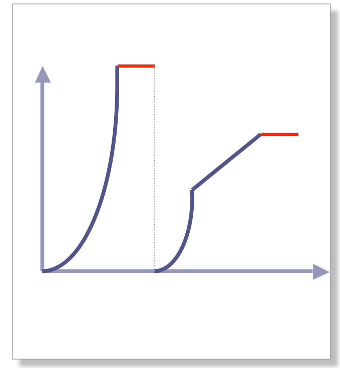


Reaching Steady State

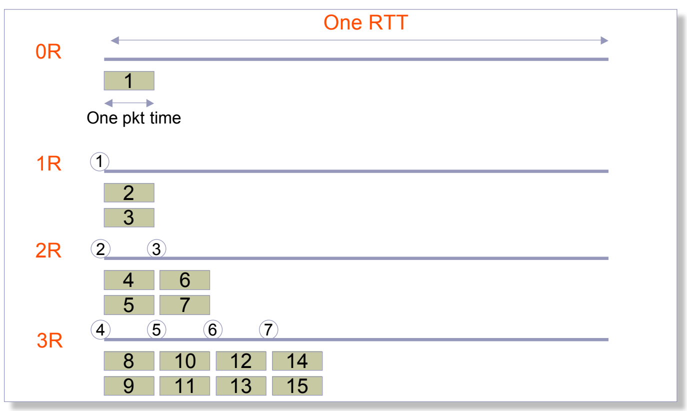
- Doing AIMD is fine in steady state but slow…
- How does TCP know what is a good initial rate to start with?
  - Should work both for a CDPD (10s of Kbps or less) and for supercomputer links (10 Gbps and growing)
- Quick initial phase to help get up to speed (slow start)

Slow Start Packet Pacing

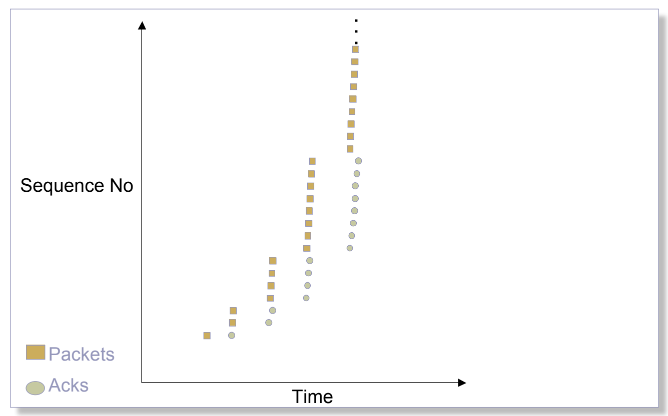
- How do we get this clocking behavior to start?
  - Initialize cwnd = 1
  - Upon receipt of every ack, cwnd = cwnd + 1
- Implications
  - Window actually increases to W in RTT * $\log_2(W)$
  - Can overshoot window and cause packet loss



Slow Start Example



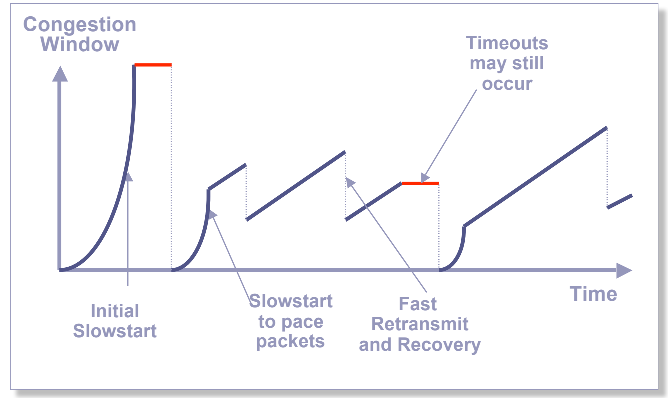
Slow Start Sequence Plot



Return to Slow Start

- If packet is lost we lose our self clocking as well
  - Need to implement slow-start and congestion avoidance together
- When timeout occurs set ssthresh to 0.5w
  - If cwnd < ssthresh, use slow start
  - Else use congestion avoidance

TCP Saw Tooth Behavior



TCP Modeling

- Given the congestion behavior of TCP can we predict what type of performance we should get?
- What are the important factors
  - Loss rate
    - Affects how often window is reduced
  - RTT
    - Affects increase rate and relates BW to window
  - RTO
    - Affects performance during loss recovery
  - MSS
    - Affects increase rate

Simple TCP Model

- Some additional assumptions
  - Fixed RTT
  - No delayed ACKs
- In steady state, TCP losses packet each time window reaches W packets
  - Window drops to W/2 packets
  - Each RTT window increases by 1 packet→W/2 * RTT before next loss
  - BW = MSS * avg window/RTT = MSS * (W + W/2)/(2 * RTT) = .75 * MSS * W / RTT

Simple Loss Model

- What was the loss rate?
  - Packets transferred = (.75 W/RTT) * (W/2 * RTT) = $3W^2/8$
  - 1 packet lost → loss rate = p = $8/3W^2$
  - W = sqrt(8 / (3 * loss rate))
- BW = .75 * MSS * W / RTT
  - BW = MSS / (RTT * sqrt (2/3p))

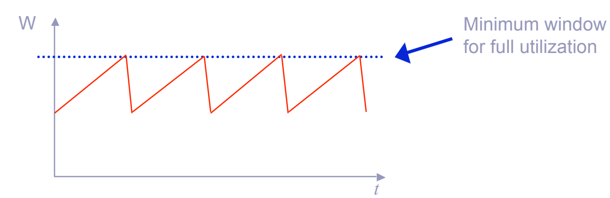
TCP Friendliness

- What does it mean to be TCP friendly?
  - TCP is not going away
  - Any new congestion control must compete with TCP flows
    - Should not clobber TCP flows and grab bulk of link
    - Should also be able to hold its own, i.e. grab its fair share, or it will never become popular
- How is this quantified/shown?
  - Has evolved into evaluating loss/throughput behavior
  - If it shows 1/sqrt(p) behavior it is ok
  - But is this really true?

TCP Performance

- Can TCP saturate a link?
- Congestion control
  - Increase utilization until… link becomes congested
  - React by decreasing window by 50%
  - Window is proportional to rate * RTT
- Doesn't this mean that the network oscillates between 50 and 100% utilization?
  - Average utilization = 75%??
  - No…this is *not* right!

Summary Unbuffered Link



- The router can't fully utilize the link
  - If the window is too small, link is not full
  - If the link is full, next window increase causes drop
  - With no buffer it still achieves 75% utilization

TCP Performance

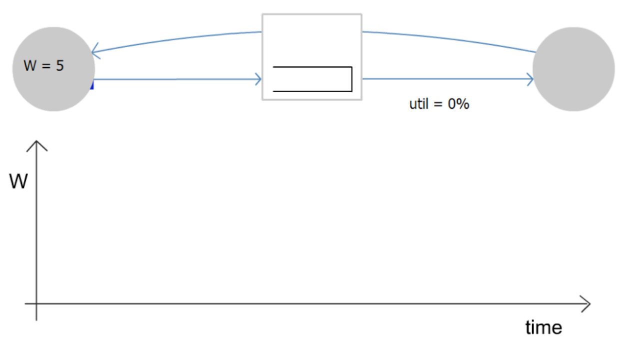
- In the real world, router queues play important role
  - Window is proportional to rate * RTT
    - But, RTT changes as well the window
  - Window to fill links = propagation RTT * bottleneck bandwidth
    - If window is larger, packets sit in queue on bottleneck link

TCP Performance

- If we have a large router queue → can get 100% utilization
  - But, router queues can cause large delays
- How big does the queue need to be?
  - Windows vary from W → W/2
    - Must make sure that link is always full
    - W/2 > RTT * BW
    - W = RTT * BW + Qsize
    - Therefore, Qsize > RTT * BW
  - Ensures 100% utilization
  - Delay?
    - Varies between RTT and 2 * RTT

Single TCP Flow

Router with large enough buffers for full link utilization



Important Lessons

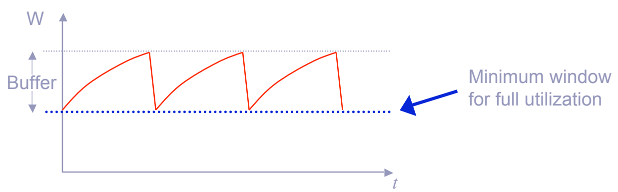
- How does TCP implement AIMD?
  - Sliding window, slow start & ack clocking
  - How to maintain ack clocking during loss recovery → fast recovery
- Modern TCP loss recovery
  - Why are timeouts bad?
  - How to avoid them? → fast retransmit, SACK
- How does TCP fully utilize a link?
  - Role of router buffers

Additional Slides for the Curious

Example

- 10Gb/s linecard
  - Requires 300Mbytes of buffering.
  - Read and write 40 byte packet every 32ns.
- Memory technologies
  - DRAM: require 4 devices, but too slow.
  - SRAM: require 80 devices, 1kW, $2000.
- Problem gets harder at 40Gb/s
  - Hence RLDRAM, FCRAM, etc.

Summary Buffered Link

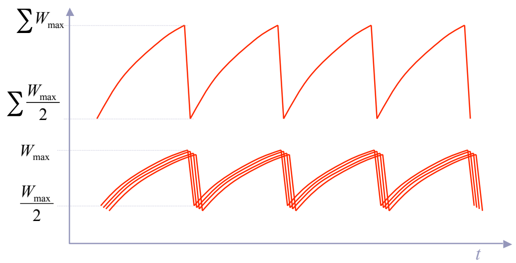


- With sufficient buffering we achieve full link utilization
  - The window is always above the critical threshold
  - Buffer absorbs changes in window size
    - Buffer Size = Height of TCP Sawtooth
    - Minimum buffer size needed is 2T*C
  - This is the origin of the rule-of-thumb

Rule-of-thumb

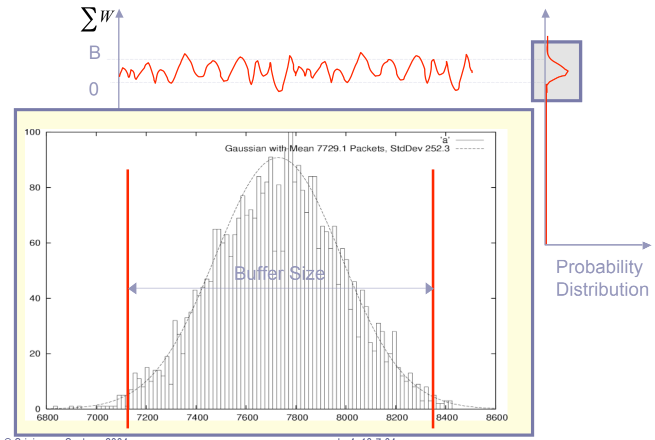
- Rule-of-thumb makes sense for one flow
- Typical backbone link has > 20,000 flows
- Does the rule-of-thumb still hold?

If flows are synchronized



- Aggregate window has same dynamics
- Therefore buffer occupancy has same dynamics
- Rule-of-thumb still holds.

If flows are not synchronized

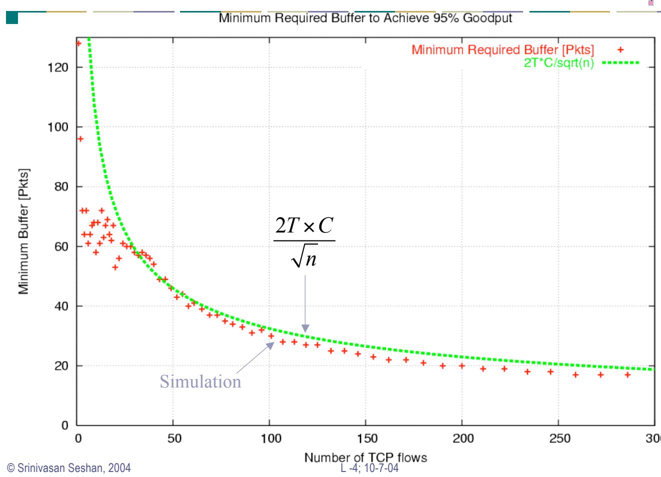


Central Limit Theorem

- CLT tells us that the more variables (Congestion Windows of Flows) we have, the narrower the Gaussian (Fluctuation of sum of windows)
  - Width of Gaussian decreases with $\frac{1}{\sqrt{n}}$
  - Buffer size should also decreases with $\frac{1}{\sqrt{n}}$

$$B \rightarrow \frac{B_{n=1}}{\sqrt{n}} = \frac{2T \times C}{\sqrt{n}}$$

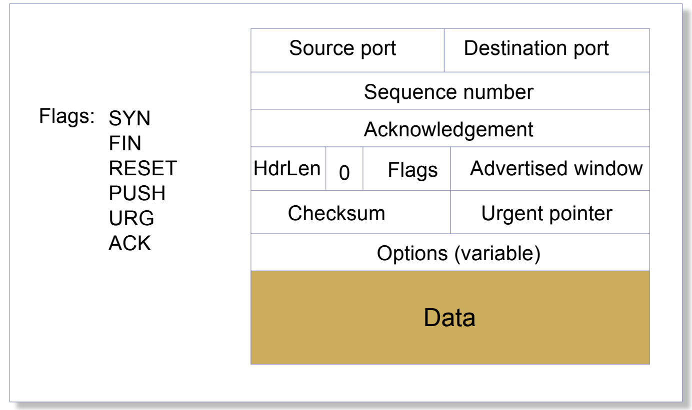
Required buffer size



Integrity & Demultiplexing

- Port numbers
  - Demultiplex from/to process
  - Servers wait on well known ports (/etc/services)
- Checksum
  - Is it sufficient to just checksum the packet contents?
  - No, need to ensure correct source/destination
    - Pseudoheader – portion of IP hdr that are critical
    - Checksum covers Pseudoheader, transport hdr, and packet body
- UDP provides just integrity and demux

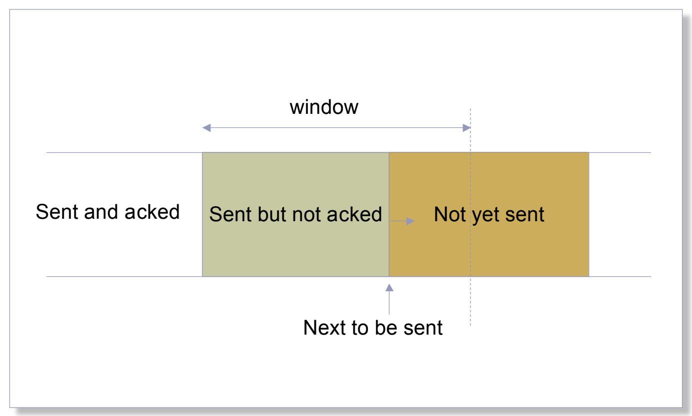
TCP Header



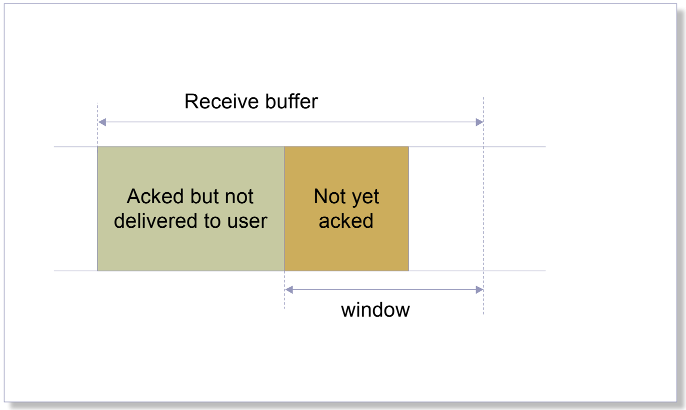
TCP Flow Control

- TCP is a sliding window protocol
  - For window size *n*, can send up to *n* bytes without receiving an acknowledgement
  - When the data is acknowledged then the window slides forward
- Each packet advertises a window size
  - Indicates number of bytes the receiver has space for
- Original TCP always sent entire window
  - Congestion control now limits this

Window Flow Control: Send Side



Window Flow Control: Receive Side

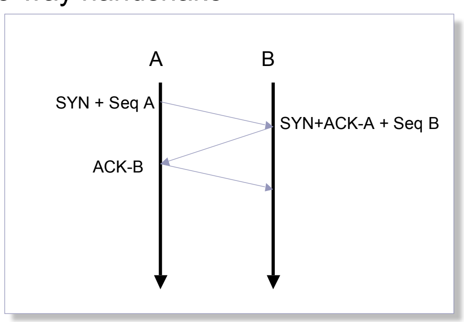


TCP Persist

- What happens if window is 0?
  - Receiver updates window when application reads data
  - What if this update is lost?
- TCP Persist state
  - Sender periodically sends 1 byte packets
  - Receiver responds with ACK even if it can't store the packet

Connection Establishment

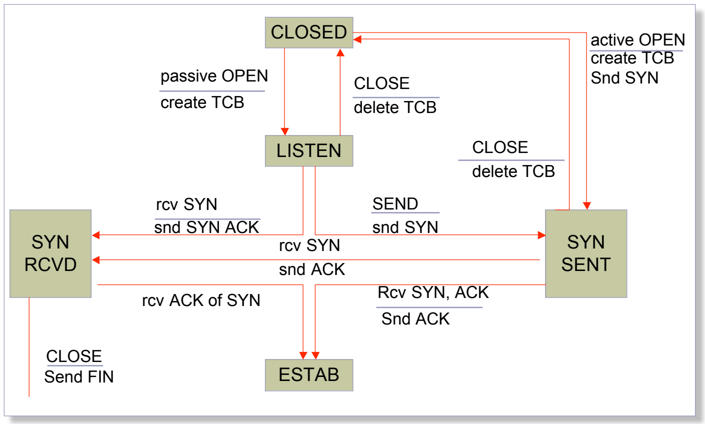
- A and B must agree on initial sequence number selection
  - Use 3-way handshake



Sequence Number Selection

- Why not simply chose 0?
- Must avoid overlap with earlier incarnation

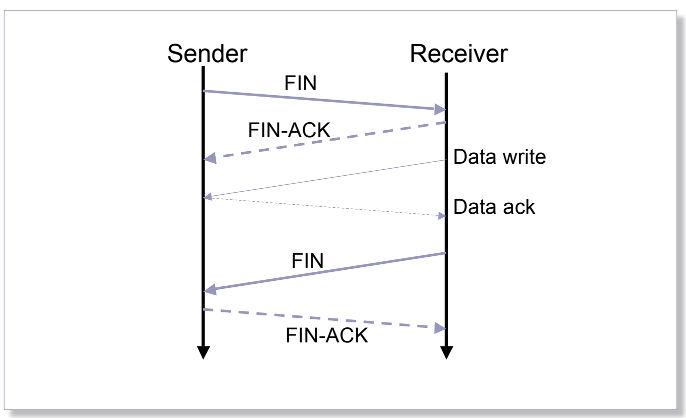
Connection Setup



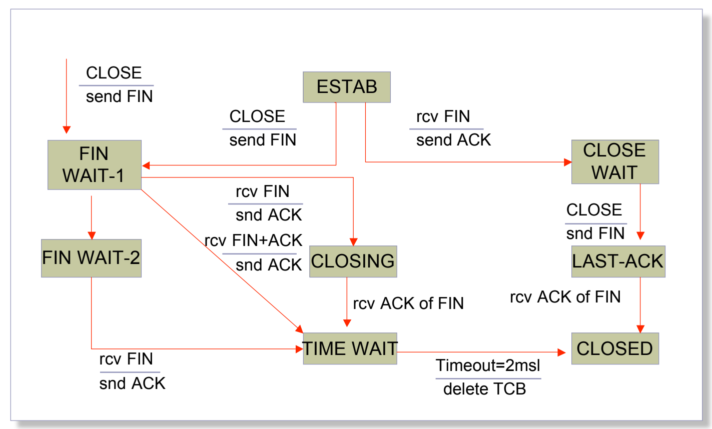
Connection Tear-down

- Normal termination
  - Allow unilateral close
- TCP must continue to receive data even after closing
- Cannot close connection immediately
  - What if a new connection restarts and uses same sequence number?

Tear-down Packet Exchange



Connection Tear-down



Detecting Half-open Connections

TCP A | TCP B

1. (CRASH) | (send 300, receive 100)
2. CLOSED | ESTABLISHED
3. SYN-SENT → <SEQ=400><CTL=SYN> → (??)
4. (!!) ← <SEQ=300><ACK=100><CTL=ACK> ← ESTABLISHED
5. SYN-SENT → <SEQ=100><CTL=RST> → (Abort!!)
6. SYN-SENT | CLOSED
7. SYN-SENT → <SEQ=400><CTL=SYN> →

Observed TCP Problems

- Too many small packets
  - Silly window syndrome
  - Nagel's algorithm
- Initial sequence number selection
- Amount of state maintained

Silly Window Syndrome

- Problem: (Clark, 1982)
  - If receiver advertises small increases in the receive window then the sender may waste time sending lots of small packets
- Solution
  - Receiver must not advertise small window increases
  - Increase window by min(MSS,RecvBuffer/2)

Nagel's Algorithm

- Small packet problem:
  - Don't want to send a 41 byte packet for each keystroke
  - How long to wait for more data?
- Solution:
  - Allow only one outstanding small (not full sized) segment that has not yet been acknowledged

Why is Selecting ISN Important?

- Suppose machine X selects ISN based on predictable sequence
- Fred has .rhosts to allow login to X from Y
- Evil Ed attacks
  - Disables host Y – denial of service attack
  - Make a bunch of connections to host X
  - Determine ISN pattern a guess next ISN
  - Fake pkt1: [<src Y><dst X>, guessed ISN]
  - Fake pkt2: desired command

Time Wait Issues

- Web servers not clients close connection first
  - Established → Fin-Waits → Time-Wait → Closed
  - Why would this be a problem?
- Time-Wait state lasts for 2 * MSL
  - MSL is should be 120 seconds (is often 60s)
  - Servers often have order of magnitude more connections in Time-Wait

TCP Extensions

- Implemented using TCP options
  - Timestamp
  - Protection from sequence number wraparound
  - Large windows

Protection From Wraparound

- Wraparound time vs. Link speed
  - 1.5Mbps: 6.4 hours
  - 10Mbps: 57 minutes
  - 45Mbps: 13 minutes
  - 100Mbps: 6 minutes
  - 622Mbps: 55 seconds → < MSL!
  - 1.2Gbps: 28 seconds
- Use timestamp to distinguish sequence number wraparound

Large Windows

- Delay-bandwidth product for 100ms delay
    - 1.5Mbps: 18KB
    - 10Mbps: 122KB > max 16bit window
    - 45Mbps: 549KB
    - 100Mbps: 1.2MB
    - 622Mbps: 7.4MB
    - 1.2Gbps: 14.8MB
- Scaling factor on advertised window
    - Specifies how many bits window must be shifted to the left
    - Scaling factor exchanged during connection setup

Maximum Segment Size (MSS)

- Exchanged at connection setup
    - Typically pick MTU of local link
- What all does this effect?
    - Efficiency
    - Congestion control
    - Retransmission
- Path MTU discovery
    - Why should MTU match MSS?