This Lecture: Congestion Control

- Congestion Control

- Assigned Reading
  - [Chiu & Jain] Analysis of Increase and Decrease Algorithms for Congestion Avoidance in Computer Networks
  - [Jacobson and Karels] Congestion Avoidance and Control

Introduction to TCP

- Communication abstraction:
  - Reliable
  - Ordered
  - Point-to-point
  - Byte-stream
  - Full duplex
  - Flow and congestion controlled

- Protocol implemented entirely at the ends
  - Fate sharing

- Sliding window with cumulative acks
  - Ack field contains last in-order packet received
  - Duplicate acks sent when out-of-order packet received

Key Things You Should Know Already

- Port numbers
- TCP/UDP checksum
- Sliding window flow control
  - Sequence numbers
- TCP connection setup
- TCP reliability
  - Timeout
  - Data-driven
Evolution of TCP

- 1973: Three-way handshake described by Vint Cerf and Bob Kahn in IEEE Trans Comm
- 1974: TCP described by Vint Cerf and Bob Kahn in IEEE Trans Comm
- 1983: BSD Unix 4.2 supports TCP/IP
- 1984: Nagel's algorithm to reduce overhead of small packets, predict congestion collapse
- 1987: Karn's algorithm to better estimate round-trip time
- 1990: 4.3BSD Reno fast retransmit delayed ACK's
- 1974: TCP described by Vint Cerf and Bob Kahn in IEEE Trans Comm
- 1982: TCP & IP RFC 793 & 791
- 1983: BSD Unix 4.2 supports TCP/IP
- 1984: Nagel's algorithm to reduce overhead of small packets, predict congestion collapse
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Overview

- Congestion sources and collapse
- Congestion control basics
- TCP congestion control
- TCP modeling

TCP Through the 1990s

- 1993: TCP Vegas (Brakmo et al) real congestion avoidance
- 1994: T/TCP (Braden) Transaction TCP
- 1996: SACK TCP (Floyd et al) Selective Acknowledgement
- 1993: TCP Vegas (Brakmo et al) real congestion avoidance
- 1994: ECN (Floyd) Explicit Congestion Notification
- 1996: Hoe improving TCP startup
- 1996: FACK TCP (Mathis et al) extension to SACK

Congestion

- Different sources compete for resources inside network
- 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

\[ \frac{C}{2} \]

- 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
Where to Prevent Collapse?

- Can end hosts prevent problem?
  - Yes, but must trust end hosts to do right thing
  - E.g., sending host must adjust amount of data it puts in the network based on detected congestion
- Can routers prevent collapse?
  - No, not all forms of collapse
  - Doesn’t mean they can’t help
    - Sending accurate congestion signals
    - Isolating well-behaved from ill-behaved sources

Congestion Control and Avoidance

- A mechanism which:
  - Uses network resources efficiently
  - Preserves fair network resource allocation
  - Prevents or avoids collapse
- Congestion collapse is not just a theory
  - Has been frequently observed in many networks

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Objectives

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

- Let’s assume window-based control
- Reduce window when congestion is perceived
  - 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
- Increase window otherwise
  - Probe for available bandwidth – how?

Linear Control

- Many different possibilities for reaction to congestion and probing
  - Examine simple linear controls
    - \( \text{Window}(t + 1) = a + b \times \text{Window}(t) \)
    - Different \( a/b \) 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

Convergence to Efficiency

Distributed Convergence to Efficiency
Convergence to Fairness

Convergence to Efficiency & Fairness

Increase

Constraints

- Distributed efficiency
  - i.e., $\sum \text{Window}(t+1) > \sum \text{Window}(t)$ during increase
  - $a_i > 0$ & $b_i \geq 1$
  - Similarly, $a_d < 0$ & $b_d \leq 1$
- Must never decrease fairness
  - $a$ & $b$'s must be $\geq 0$
  - $a_i/b_i > 0$ and $a_d/b_d \geq 0$
- Full constraints
  - $a_d = 0$, $0 \leq b_d < 1$, $a_i > 0$ and $b_i \geq 1$
What is the Right Choice?

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

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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

- 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

Implementation Issue

- Operating system timers are very coarse – how to pace packets out smoothly?
- Implemented using a congestion window that limits how much data can be in the network.
  - TCP also keeps track of how much data is in transit
- Data can only be sent when the amount of outstanding data is less than the congestion window.
  - The amount of outstanding data is increased on a “send” and decreased on “ack”
  - (last sent – last acked) < congestion window
- Window limited by both congestion and buffering
  - Sender’s maximum window = Min (advertised window, cwnd)

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? \( \rightarrow \) 1 MSS worth of bytes
    - After cwnd packets have passed by \( \rightarrow \) approximately increase of 1 MSS
- Implements AIMD
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
    - 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

How to Change Window

- When a loss occurs have W packets outstanding
- New cwnd = 0.5 * cwnd
  - How to get to new state?

Fast Recovery

- Each duplicate ack notifies sender that single packet has cleared network
- When < cwnd packets are outstanding
  - Allow new packets out with each new duplicate acknowledgement
- Behavior
  - Sender is idle for some time – waiting for $\frac{1}{2}$ cwnd worth of dupacks
  - Transmits at original rate after wait
    - Ack clocking rate is same as before loss
Fast Recovery

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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

Overall TCP Behavior

- Let's concentrate on steady state behavior with no timeouts and perfect loss recovery
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 before next loss
  - \[ BW = MSS * \frac{W + W/2}{2 * RTT} \]
  - \[ BW = \frac{.75 * MSS * W}{RTT} \]

Simple Loss Model

- What was the loss rate?
  - Packets transferred between losses = \[ \text{Avg BW} * \text{time} = (.75 W/RTT) * (W/2 * RTT) = 3W^2/8 \]
  - 1 packet lost \( \Rightarrow \) loss rate = \( p = 8/3W^2 \)
  - \( W = \sqrt{\frac{8}{3 * \text{loss rate}}} \)
  - \[ BW = .75 * MSS * W / RTT \]
  - \[ BW = MSS / (RTT * \sqrt{2/3p}) \]

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!
TCP Congestion Control

Rule for adjusting $W$
- If an ACK is received: $W \leftarrow W + 1/W$
- If a packet is lost: $W \leftarrow W/2$

Only $W$ packets may be outstanding

Source $\rightarrow$ Dest

Window size $W_{\text{max}}$

$W_{\text{max}} / 2$

Summary Unbuffered Link

$W_{\text{max}}$

Minimum window for full utilization

- 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

Single TCP Flow

Router without buffers

$W = 1$

util = 0%

$W$

time
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 $W \rightarrow W/2$
  - Must make sure that link is always full
  - $W/2 > \text{RTT} \times \text{BW}$
  - $W = \text{RTT} \times \text{BW} + \text{Qsize}$
  - Therefore, Qsize > RTT \times BW
- **Ensures 100% utilization**
- Delay?
  - Varies between RTT and $2 \times \text{RTT}$

**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 $2T^2 C$
  - This is the origin of the rule-of-thumb

**Single TCP Flow**

Router with large enough buffers for full link utilization

- $W = 5$
- $\text{util} = 0\%$

**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.
- **Problem gets harder at 40Gb/s**
  - Hence RLDRAM, FCRAM, etc.
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

\[ \sum w_i, \sum \frac{w_i}{2}, \frac{w_{max}}{2} \]

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

If flows are not synchronized

\[ \sum w_i, B, B_0, \sum \frac{w_i}{n}, \frac{w_{max}}{n} \]

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

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 \( \frac{1}{\sqrt{n}} \)

\[ B \rightarrow \frac{B_{max}}{\sqrt{n}} = \frac{2T \times C}{\sqrt{n}} \]
Required buffer size

\[ 2T \times \frac{C}{\sqrt{n}} \]

Simulation

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

Next Lecture

- Fair-queueing
- Assigned reading
  - [Demers, Keshav, Shenker] Analysis and Simulation of a Fair Queueing Algorithm
  - [Stoica, Shenker, Zhang] Core-Stateless Fair Queueing: Achieving Approximately Fair Bandwidth Allocations in High Speed Networks*