Lecture 21: Congestion Control

CSE 123: Computer Networks
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UCSD CSE
Lecture 21 Overview

- How fast should a sending host transmit data?
  - Not to fast, not to slow, just right...

- Should not be faster than the sender’s share
  - Bandwidth allocation

- Should not be faster than the network can process
  - Congestion control

- Congestion control & bandwidth allocation are separate ideas, but frequently combined
● How much bandwidth should each flow from a source to a destination receive when they compete for resources?
  ♦ What is a “fair” allocation?
Buffer intended to absorb bursts when input rate > output
But if sending rate is persistently > drain rate, queue builds
Dropped packets represent wasted work; goodput < throughput
Drop-Tail Queuing

Loss due to Congestion

Congestion collapse

Network Load

Throughput

Latency
Congestion Collapse

- Rough definition: “When an increase in network load produces a decrease in useful work”

- Why does it happen?
  - Sender sends faster than bottleneck link speed
  - Packets queue until dropped
  - In response to packets being dropped, sender retransmits
  - All hosts repeat in steady state…
Mitigation Options

- Increase network resources
  - More buffers for queuing
  - Increase link speed
  - Pros/Cons of these approaches?

- Reduce network load
  - Send data more slowly
  - How much more slowly?
  - When to slow down?
Designing a Control

- **Open loop**
  - Explicitly reserve bandwidth in the network in advance

- **Closed loop**
  - Respond to feedback and adjust bandwidth allocation

- **Network-based**
  - Network implements and enforces bandwidth allocation

- **Host-based**
  - Hosts are responsible for controlling their sending rate
Proactive vs. Reactive

- **Congestion avoidance**: try to stay to the left of the knee
- **Congestion control**: try to stay to the left of the cliff
Challenges to Address

- How to detect congestion?
- How to limit sending data rate?
- How fast to send?
Detecting Congestion

- Explicit congestion signaling
  - Source Quench: ICMP message from router to sender
  - DECBit / Explicit Congestion Notification (ECN):
    » Router marks packet based on queue occupancy (i.e. indication that packet encountered congestion along the way)
    » Receiver tells sender if queues are getting too full

- Implicit congestion signaling
  - Packet loss
    » Assume congestion is primary source of packet loss
    » Lost packets indicate congestion
  - Packet delay
    » Round-trip time increases as packets queue
    » Packet inter-arrival time is a function of bottleneck link
Throttling Options

- **Window-based (TCP)**
  - Constrain number of outstanding packets allowed in network
  - Increase window to send faster; decrease to send slower
  - **Pro**: Cheap to implement, good failure properties
  - **Con**: Creates traffic bursts (requires bigger buffers)

- **Rate-based (many streaming media protocols)**
  - Two parameters (period, packets)
  - Allow sending of x packets in period y
  - **Pro**: smooth traffic
  - **Con**: fine-grained per-connection timers, what if receiver fails?
Choosing a Send Rate

- Ideally: Keep equilibrium at “knee” of power curve
  - Find “knee” somehow
  - Keep number of packets “in flight” the same
  - Don’t send a new packet into the network until you know one has left (i.e. by receiving an ACK)
  - What if you guess wrong, or if bandwidth availability changes?

- Compromise: adaptive approximation
  - If congestion signaled, reduce sending rate by $x$
  - If data delivered successfully, increase sending rate by $y$
  - How to relate $x$ and $y$? Most choices don’t converge…
Each source independently probes the network to determine how much bandwidth is available
- Changes over time, since everyone does this

Assume that packet loss implies congestion
- Since errors are rare; also, requires no support from routers
Window-based congestion control

- Unified congestion control and flow control mechanism
- $rwin$: advertised flow control window from receiver
- $cwnd$: congestion control window
  - Estimate of how much outstanding data network can deliver in a round-trip time
- Sender can only send $\text{MIN}(rwin, cwnd)$ at any time

Idea: decrease $cwnd$ when congestion is encountered; increase $cwnd$ otherwise

- Question: how much to adjust?
Goal: Adapt to changes in available bandwidth

Additive Increase, Multiplicative Decrease (AIMD)
- Increase sending rate by a constant (e.g. MSS)
- Decrease sending rate by a linear factor (e.g. divide by 2)

Rough intuition for why this works
- Let $L_i$ be queue length at time $i$
- In steady state: $L_i = N$, where $N$ is a constant
- During congestion, $L_i = N + yL_{i-1}$, where $y > 0$
- Consequence: queue size increases multiplicatively
  » Must reduce sending rate multiplicatively as well
AIMD

- Increase slowly while we believe there is bandwidth
  - Additive increase per RTT
  - $cwnd += 1$ full packet / RTT

- Decrease quickly when there is loss (went too far!)
  - Multiplicative decrease
  - $cwnd /= 2$
TCP Congestion Control

Only $W$ packets may be outstanding

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

TCP Congestion Control

W = 11.9

util = 100%

W

CSE 123 – Lecture 21: Congestion Control
For next time...

- Finish reading P&D 6.3-4, Sidebar on NAT on p.335
- See you Wednesday (yes, there is lecture!)
- Keep going on Project 2...