Congestion Control

CS 168 Fall 2024

Sylvia Ratnasamy

Today: Congestion Control

- One of the core topics in networking
- Today: concepts, design space, TCP's approach
- Next lecture: TCP walk-through and advanced topics

Recall: Lecture 3



- If two packets arrive at a router at the same time, the router will transmit one and buffer the other
- If many packets arrive close in time
 - the router cannot keep up \rightarrow gets congested
 - causes packet delays and drops

Some History: TCP in the 1980s

- Sending rate only limited by flow control
 - Dropped packets → senders retransmit, repeatedly!

• Led to "congestion collapse" in Oct. 1986

In October of '86, the Internet had the first of what became a series of 'congestion collapses'. During this period, the data throughput from LBL to UC Berkeley (sites separated by 400 yards and two IMP hops) dropped from 32 Kbps to 40 bps. We were fascinated by this sudden factor-of-thousand drop in bandwidth and embarked on an investigation of why things had gotten so bad. In particular, we wondered if the 4.3BSD (Berkeley UNIX) TCP was mis-behaving or if it could be tuned to work better under abysmal network conditions. The answer to both of these questions was "yes".

-- Karels (UCB) and Jacobson(LBL)

Van Jacobson



- Researcher at the Lawrence Berkeley National Lab.
- Many contributions to the early TCP/IP stack
- Creator of many widely used network tools
 - traceroute, tcpdump, Berkeley Packet Filter, ...
- Later Chief Scientist at Cisco, now at Google
- Recently: BBR, new CC protocol used by youtube, etc.

Their Approach

- Incremental extension to TCP's existing protocol
 - Source adjusts its window size based on observed packet loss
- A pragmatic and effective solution
 - Required no upgrades to routers or applications!
 - Patch of a few lines of code to BSD's TCP implementation
 - Quickly adopted and has been the de-facto approach since
- Extensively researched and improved upon

CC more generally...

- Huge literature on the problem
 - In systems, control theory, game theory, stats, econ
- Recent resurgence of interest in industry
 - New emphasis on high-performance (cloud services)
 - New context (datacenters, LLMs)
 - New methods (ML)

Topics for today

- What makes CC a hard problem?
- Goals for a good solution
- Design space
- Components of a solution
- TCP's approach (high level)



At what rate should Host A send traffic?

*For this example, we'll ignore the BW of links attaching hosts to routers



Depends on the destination



Changes with routing dynamics



Depends on "competing" flows



Including "indirect" competition!

Congestion Control

- Fundamentally, a resource allocation problem
 - Flow is assigned a shared of the link BW along a path
- But more complex than traditional resource alloc.
 - Changing one link's allocation can have <u>global</u> impact
 - And we're changing allocations on every flow arrival/exit
 - No single entity has a complete view or complete control!
- Allocations in our context are highly interdependent

Topics for today

- What makes CC a hard problem?
- Goals for a good solution
- Design space
- Components of a solution
- TCP's approach (high level)

Goals

• From a resource allocation perspective

- Low packet delay and loss
- High link utilization
- "Fair" sharing across flows

Aim: a good <u>tradeoff</u> between the above goals

Goals

- From a resource allocation perspective
 - Low packet delay and loss
 - High link utilization
 - "Fair" sharing across flows
- From a systems perspective
 - **Practical:** scalable, decentralized, adaptive, *etc.*

Topics for today

- What makes CC a hard problem?
- Goals for a good solution
- Design space
- Components of a solution
- TCP's approach (high level)

(0) Send at will



What happens if A sends at 10Gbps?

(1) Reservations

- Pre-arrange bandwidth allocations
- Comes with all the problems we've discussed

- (1) Reservations
- (2) Pricing / priorities
 - Don't drop packets for the highest bidders/priority users
 - Charge users based on current congestion levels
 - Requires payment model

- (1) Reservations
- (2) Pricing / priorities
- (3) Dynamic Adjustment
 - Hosts dynamically learn current level of congestion
 - Adjust their sending rate accordingly

- (1) Reservations
- (2) Pricing / priorities
- (3) Dynamic Adjustment

In practice, the **generality** of dynamic adjustment has proven powerful

- Doesn't presume business model
- Doesn't assume we know app/user requirements
- But does assume good citizenship!



(1) First, host A discovers it can send at ~10Gbps
(2) A notices that ~10Gbps is congesting the network
(3) A figures out it should cut its rate to ~1Gbps



(4) A notices that 1Gbps is congesting the network(5) A figures out it should cut its rate to (say) ½ Gbps

Two broad classes of solutions

Host-based CC → Jacobson's original TCP approach

- No special support from routers
- Hosts adjust rate based on <u>implicit</u> feedback from routers

Router-assisted CC

- Routers signal congestion back to hosts
- Hosts pick rate based on <u>explicit</u> feedback from routers

Taking stock: where we are in the design space



Taking stock: where we are in the design space



Topics for today

- What makes CC a hard problem?
- Goals for a good solution
- Design space
- Components of a solution
- TCP's approach (high level)

Sketch of a (host-based) solution

How do we pick the initial rate? runs the following:

• Pick initial rate R

How do we detect congestion

- Try sending at a rate R for some period of time
 - Did I experience congestion in this time period?
 - If yes, reduce R
 - If no, increase R
 - Repeat

By how much should we increase/decrease

Components of a Solution

- Detecting congestion
- Discovering an initial rate
- Reacting to congestion (or lack thereof)
 - Increase/decrease rules

Detecting Congestion?

Packet loss

• Approach commonly used by TCP

Benefits

- Fail-safe signal
- Already something TCP detects to implement reliability

Cons

- Complication: non-congestive loss (e.g., checksum err.)
- Complication: reordering (e.g., with cumulative ACKs)
- Detection occurs after packets have experienced delay

Detecting Congestion?

• Increase in packet delay

- Long considered tricky to get right: packet delay varies with queue size and competing traffic
- Google's BBR protocol is challenging this assumption

Note: Not All Losses the Same

• Duplicate ACKs: isolated loss

- Packets and ACKs still getting through
- Suggests mild congestion levels
- **Timeout**: much more serious
 - Not enough packets/dupACKs getting through
 - Must have suffered several losses
- We'll see that TCP reacts differently in each case

Taking stock: where we are in the design space



Discovering an initial rate?

- Goal: estimate available bandwidth
 - Start slow (for safety)
 - But ramp up quickly (for efficiency)
- Toy example of an inefficient solution
 - Add ½ Mbps every 100ms until we detect loss
 - If available BW is 1Mbps, will discover rate in 200ms
 - If available BW is 1Gbps, will take 200 seconds
 - Either is possible!
Solution: "Slow Start"

- Start with a small rate (hence the name)
 - Might be much less than actual bandwidth
 - Linear increase takes too long to ramp up
- Increase exponentially until first loss
 - E.g., double rate until first loss
- A "safe" rate is half of that when first loss occurred
 - I.e., if first loss occurred at rate R, then R/2 is safe rate

Components of a Solution

- Discovering an initial rate
- Detecting congestion
- Reacting to congestion (or lack thereof)
 - Increase/decrease rules

Sketch of a solution

Each source independently runs the following:

- Pick initial rate R
- Try sending at a rate R for some time period
 - Did I experience congestion in this time period?
 - If yes, reduce R
 - If no, increase R
 - Repeat

By how much should we increase/decrease?

Rate adjustment

- This is a critical part of a CC design!
- Determines how quickly a host adapts to <u>changes</u> in available bandwidth
- Determines how effectively BW is consumed
- Determines how BW is shared (fairness)

Goals for rate adjustment

- Efficiency: High utilization of link bandwidth
- Fairness: Each flow gets equal share

How should we adjust rate?

- At the highest level: fast or slow
- Fast: multiplicative increase/decrease
 - E.g., increase/decrease by $2x (R \rightarrow 2R \text{ or } R/2)$
- Slow: additive increase/decrease
 - E.g., increase/decrease by +1 ($R \rightarrow R+1$ or R-1)

Leads to four alternatives

- AIAD: gentle increase, gentle decrease
- AIMD: gentle increase, rapid decrease
- **MIAD**: rapid increase, gentle decrease
- **MIMD**: rapid increase, rapid decrease

Leads to four alternatives

- AIAD: gentle increase, gentle decrease
- AIMD: gentle increase, rapid decrease
- MIAD: rapid increase, gentle decrease
- MIMD: rapid increase, rapid decrease

Why AIMD? Intuition

- Consequences of sending too much are worse than sending too little
 - Too much: packets dropped and retransmitted
 - Too little: somewhat lower throughput
- General approach:
 - Gentle increase when uncongested (exploration)
 - Rapid decrease when congested

Why AIMD? In more detail...

- Consider a simple model
 - Two flows going over single link of capacity C
 - Sending at rates X1 and X2 respectively
- When X1+X2 > C, network is congested
- When X1+X2 < C, network is underloaded
- Would like *both*:
 - $X1 + X2 = C \rightarrow$ link is fully utilized with no congestion
 - X1 = X2 \rightarrow sharing is "fair"

Simple Model, C=1

- Two users with rates x₁ and x₂
- Congestion when x₁+x₂ > 1
- Unused capacity when x₁+x₂ < 1
- Fair when $x_1 = x_2$



Example Allocations, C=1



Example Adjustments



Example Adjustments



Our Four Options

- AIAD: gentle increase, gentle decrease
- AIMD: gentle increase, rapid decrease
- MIAD: rapid increase, gentle decrease
- MIMD: rapid increase, rapid decrease
- And now apply our simple model!

AIAD Dynamics

• Consider: Increase: +1 Decrease: -2

• First iteration: no congestion

•
$$X1 \rightarrow 2, X2 \rightarrow 4$$

Second iteration: congestion

•
$$X1 \rightarrow 0, X2 \rightarrow 2$$

Third iteration: no congestion Back where we started!
X1 → 1, X2 → 3 → Gap between X1 and X2

didn't change at all

AIAD

- Increase: x + a
- Decrease: x b

 Does not converge to fairness



User 1: x_1

MIMD Dynamics

- Consider: Increase: ×2 Decrease: ÷4
- Start at $X1 = \frac{1}{2}$, X2 = 1, with C = 5
- First iteration: no congestion
 - $X1 \rightarrow 1, X2 \rightarrow 2$
- Second iteration: no congestion
 - $X1 \rightarrow 2, X2 \rightarrow 4$
- Third iteration: congestion
 - $X1 \rightarrow \frac{1}{2}$, $X2 \rightarrow 1$

Again, no improvement in fairness

MIMD

- Increase: $x \times b_I$
- Decrease: $x \times b_D$

Does not converge to fairness



User 1: x_1

MIAD Dynamics

- Consider: Increase: ×2 Decrease: -1
- Start at X1 = 1, X2 = 3, with C = 5
- First iteration: no congestion; $X1 \rightarrow 2$, $X2 \rightarrow 6$
- Second iteration: congestion; $X1 \rightarrow 1$, $X2 \rightarrow 5$
- Third iteration: congestion; $X1 \rightarrow 0$, $X2 \rightarrow 4$
- Fourth iteration: no congestion; $X1 \rightarrow 0$, $X2 \rightarrow 8$

X1 pegged at 0; MIAD is maximally unfair!

AIMD Dynamics

- Consider: Increase: +1 Decrease: ÷ 2
- Start at X1 = 1, X2 = 2, with C = 5 Diff = 1
- First iteration: no congestion: $X1 \rightarrow 2$, $X2 \rightarrow 3$ Diff = 1
- Second: no congestion: $X1 \rightarrow 3$, $X2 \rightarrow 4$ Diff = 1
- Third: congestion: $X1 \rightarrow 1.5$, $X2 \rightarrow 2$ Diff = 0.5
- Fourth: no congestion: $X1 \rightarrow 2.5$, $X2 \rightarrow 3$ Diff = 0.5
- Fifth: congestion: $X1 \rightarrow 1.25$, $X2 \rightarrow 1.5$ Diff = 0.25
- Sixth: no congestion: X1 \rightarrow 2.25, X2 \rightarrow 2.5 Diff = 0.25
- Seventh: no congestion: X1 \rightarrow 3.25, X2 \rightarrow 3.5 Diff = 0.25
- Eighth: congestion: X1 \rightarrow 1.625, X2 \rightarrow 1.75 Diff = 0.125
- Ninth: no congestion: X1 \rightarrow 2.625, X2 \rightarrow 2.75 Diff = 0.125

AIMD

- Difference between X1 and X2 decreasing!
 - Difference stays constant when increasing
 - Halves every time there is a decrease

AIMD

- Increase: x+a_I
- Decrease: x*b_D

Converges to fairness



User 1: x_1

Answer to Why AIMD?

- AIMD embodies gentle increase, rapid decrease
- AIMD only choice that drives us towards "fairness"
- Out of the four options
 - AIAD, MIMD: retain unfairness
 - MIAD: maximally unfair
 - AIMD: fair and appropriate gentle/rapid actions

Any Questions?

Topics for today

- What makes CC a hard problem?
- Goals for a good solution
- Design space
- Components of a solution
- TCP's approach (high level)

Sketch of a solution

- Pick initial rate R
- Try sending at a rate R for some time period
 - Did I experience congestion in this time period?
 - If yes, reduce R
 - If no, increase R
 - Repeat

- Pick initial rate R
- Try sending at a rate R for some time period
 - Did I experience congestion in this time period?
 - If yes, reduce R
 - If no, increase R
 - Repeat

- Slow-start to find initial rate
- Try sending at a rate R for some time period
 - Did I experience congestion in this time period?
 - If yes, reduce R
 - If no, increase R
 - Repeat

- Slow-start to find initial rate
- Try sending at a rate R for some time period
 - Did I experience congestion loss in this time period?
 - If yes, reduce R
 - If no, increase R
 - Repeat

- Slow-start to find initial rate
- Try sending at a rate R for some time period
 - Did I experience congestion loss in this time period?
 - If yes, reduce R multiplicatively (2x)
 - If no, increase R additively (+1)
 - Repeat

Review: TCP's window-based operation

- Sender maintains a window of packets in flight. The window size is picked to balance three goals
 - Take advantage of network capacity ("fill the pipe")
 - Avoid overloading the receiver (flow control)
 - Avoid overloading links (congestion control)
- Flow control: sender maintains an advertised window; denoted RWND (for receiver window)
- CC: sender maintains a **congestion window (CWND)**
- Sender-side window = min{CWND, RWND}
 - Assume for this lecture that RWND > CWND



Review:

Sender maintains a **sliding** window of W **contiguous** bytes Sender maintains a single timer, for the LHS of window On timeout, sender retransmits the packet starting at *i*

Review:



Receiver sends cumulative ACKs; sender counts #dupACKs **Fast Retransmit**: Sender retransmits when #dupACKs = 3 Sender slides window on receiving an ACK for new data (j > i)

- Slow-start to find initial rate
- Try sending at a rate R for some time period
 - Did I experience congestion loss in this time period?
 - If yes, reduce R multiplicatively (2x)
 - If no, increase R additively (+1)
 - Repeat

Extending TCP with CC

- Add a congestion window parameter (CWND)
- The sender's rate is CWND/RTT
 - (assuming RWND > CWND)
- Hence, adapting CWND \rightarrow adapting sender's rate
How a TCP sender adapts its rate

- Detecting congestion
 - Loss-based
- Discovering an initial rate
 - Slow start
- Adapting rate to congestion (or lack thereof)
 - AIMD

Updating CWND (to implement slow-start and AIMD)

- CWND updates are event driven
- Three types of events relevant to CC
 - New ACK
 - k(=3) duplicate ACKs
 - Timeout

Adapting CWND based on events

- New ACK → increase CWND (based on slow-start or <u>AI</u>MD)
 - Indicates no congestion was encountered
- 3 dupACKs → decrease CWND (based on AI<u>MD</u>)
 - Indicates isolated loss
- **Timeout** → rediscover a good CWND (return to slow-start)
 - Indicates loss of several packets. Bad news!
- Let's take a closer look at how this is implemented...

Note

- Recall: TCP operates on bytestreams
- Hence, real implementations maintain CWND in bytes

• This lecture will talk about CWND in units of MSS

- MSS: Maximum Segment Size, the max number of bytes of data that one TCP packet can carry in its payload
- This is only for pedagogical purposes

How TCP Implements Slow Start

- Sender starts at a slow rate; increases rate exponentially until first loss
- In TCP: start with a small CWND = 1 (MSS)
 - So, initial sending rate is MSS/RTT
- Then double CWND every RTT until first loss
- Implemented as: On each ACK: CWND += 1 (MSS)

Slow Start in Action

Goal: Double CWND every round-trip time

Simple implementation: On each ACK, CWND += 1 (MSS)



How TCP Implements Slow Start (contd.)

- Double CWND every RTT until first loss
- Introduce a "slow start threshold" parameter
 - **SSTHRESH**, used to remember last "safe" rate
- On first loss: SSTHRESH = CWND/2

Recall: how we adapt rate

- Detecting congestion
 - Loss-based
- Discovering an initial rate
 - Slow start
- Adapting rate to congestion (or lack thereof)
 - AIMD

AIMD in TCP

- Additive increase:
 - No loss → increase CWND by 1 MSS every RTT

Implementing Additive Increase

- Implementation works by adding a fraction of an MSS every time we receive an ACK
- On receiving an ACK (for new data)
 - $CWND \rightarrow CWND + \frac{1}{CWND}$
 - $CWND \rightarrow CWND + MSS \times \frac{MSS}{CWND}$ if counting CWND in bytes
- NOTE: after full window, CWND increases by 1 MSS
 - Thus, CWND increases by 1 MSS per RTT

AIMD in **TCP**

- Additive increase:
 - No loss → increase CWND by 1 MSS every RTT
- Multiplicative decrease
 - Loss detected by 3 dupACKs \rightarrow divide CWND in half

Implementing Multiplicative Decrease

• On receiving 3rd dupACK: • $CWND \rightarrow \frac{CWND}{2}$

On Timeout

- Rationale: lost multiple packets in a window
 - Current CWND may be way off
 - Hence, need to rediscover a good rate from scratch
 - Design decision that errs on the side of caution
- Hence, on timeout:
 - Set SSTHRESH $\leftarrow \frac{CWND}{2}$
 - Set CWND ← 1 MSS & enter Slow Start mode

Summary of Increase

- When in Slow-Start phase
 - Increase CWND by 1 MSS for each new ack
- When in AIMD phase
 - Increase by 1 (MSS) for each window's worth of acked data

Summary of Decrease

- Cut CWND in half on loss detected by dupACKs
- Cut CWND all the way to 1 (MSS) on timeout
- Never drop CWND below 1 (MSS)

Slow-Start vs. AIMD

- When does a sender stop Slow-Start and start Additive Increase?
- Determined by **SSTHRESH**
- When CWND > SSTHRESH, sender switches from slow-start to AIMD's additive increase

Recap: TCP congestion control

- Discovering an initial rate
 - Slow start
- Detecting congestion
 - Loss-based
- Adapting rate to congestion (or lack thereof)
 AIMD

TCP implements the above by updating CWND on ACK arrivals and timeouts

Leads to the TCP "Sawtooth"



Next Time

- TCP: reliability and CC together
- Analyzing TCP
- Router-assisted CC



Note: TCP is "ACK Clocked"

- A new ACK advances the sliding window and lets a new data segment enter the network
 - I.e., ACKs "clock" data segments
- What's the benefit of ACK clocking?

ACK Clocking





Consider: source sends a burst of packets Packets are queued and "spread out" at slow link ACKs maintain the spread on the return path





Sender clocks new packets with the spread

Now sending without queuing at the bottleneck link!