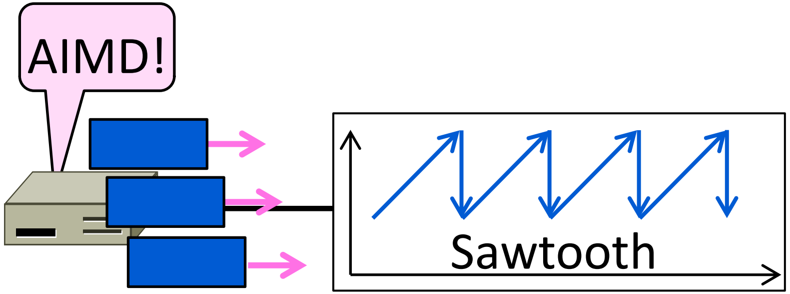
Topic

- Bandwidth allocation models
  - Additive Increase Multiplicative Decrease (AIMD) control law



Recall

- Want to allocate capacity to senders
  - Network layer provides feedback
  - Transport layer adjusts offered load
  - A good allocation is efficient and fair

- How should we perform the allocation?
  - Several different possibilities …

Bandwidth Allocation Models

- Open loop versus closed loop
  - Open: reserve bandwidth before use
  - Closed: use feedback to adjust rates
- Host versus Network support
  - Who is sets/enforces allocations?
- Window versus Rate based
  - How is allocation expressed?

TCP is a closed loop, host-driven, and window-based

Bandwidth Allocation Models (2)

- We'll look at closed-loop, host-driven, and window-based too

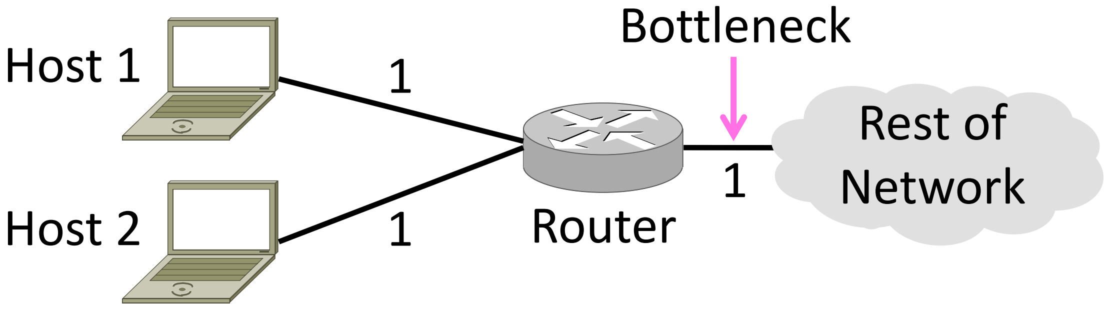
- Network layer returns feedback on current allocation to senders
  - At least tells if there is congestion
- Transport layer adjusts sender's behavior via window in response
  - How senders adapt is a control law

Additive Increase Multiplicative Decrease

- AIMD is a control law hosts can use to reach a good allocation
    - Hosts additively increase rate while network is not congested
    - Hosts multiplicatively decrease rate when congestion occurs
    - Used by TCP ☺
- Let's explore the AIMD game …

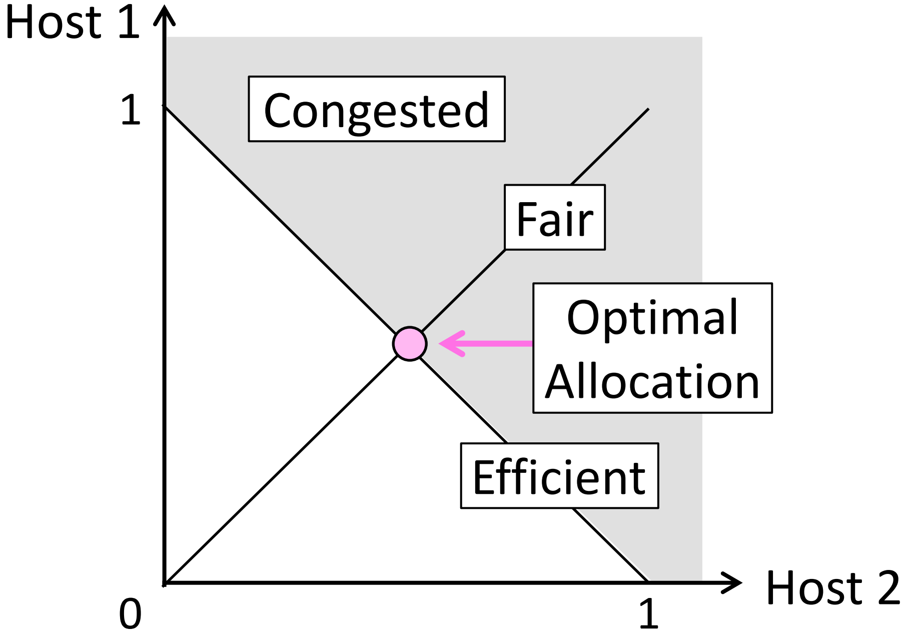
AIMD Game

- Hosts 1 and 2 share a bottleneck
  - But do not talk to each other directly
- Router provides binary feedback
  - Tells hosts if network is congested



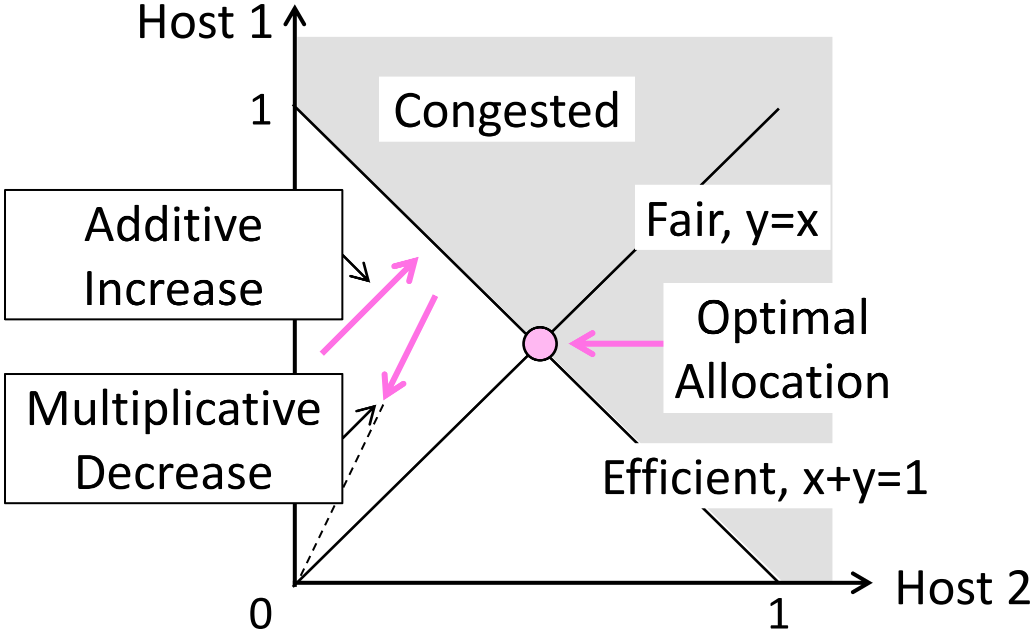
AIMD Game (2)

- Each point is a possible allocation



AIMD Game (3)

- AI and MD move the allocation



Host 1

1

Congested

Fair, y=x

Additive Increase

Optimal Allocation

Multiplicative Decrease

Efficient, x+y=1

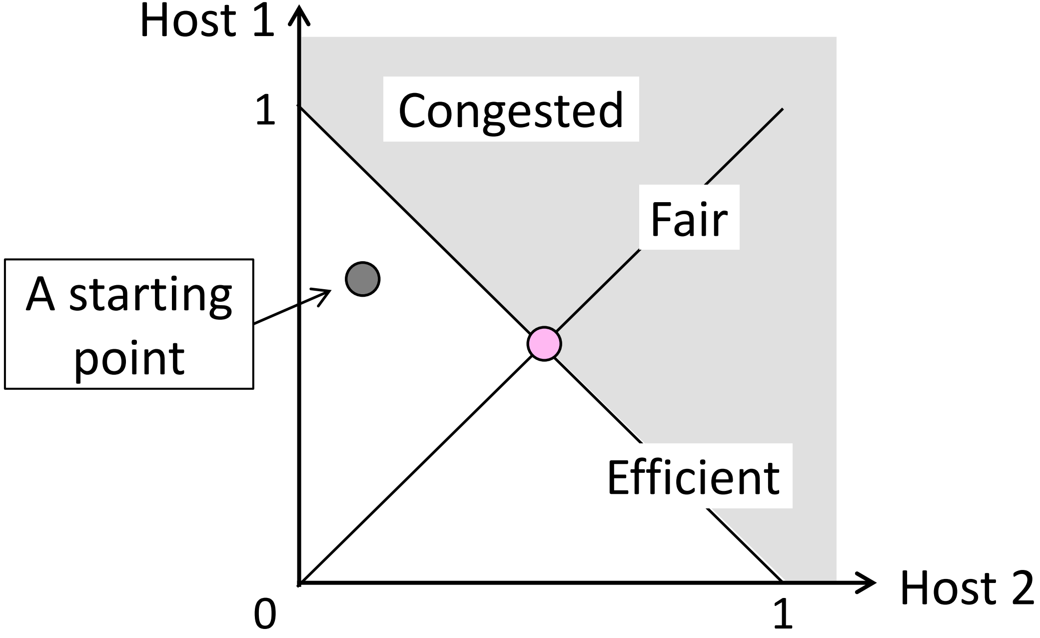
0

1

Host 2

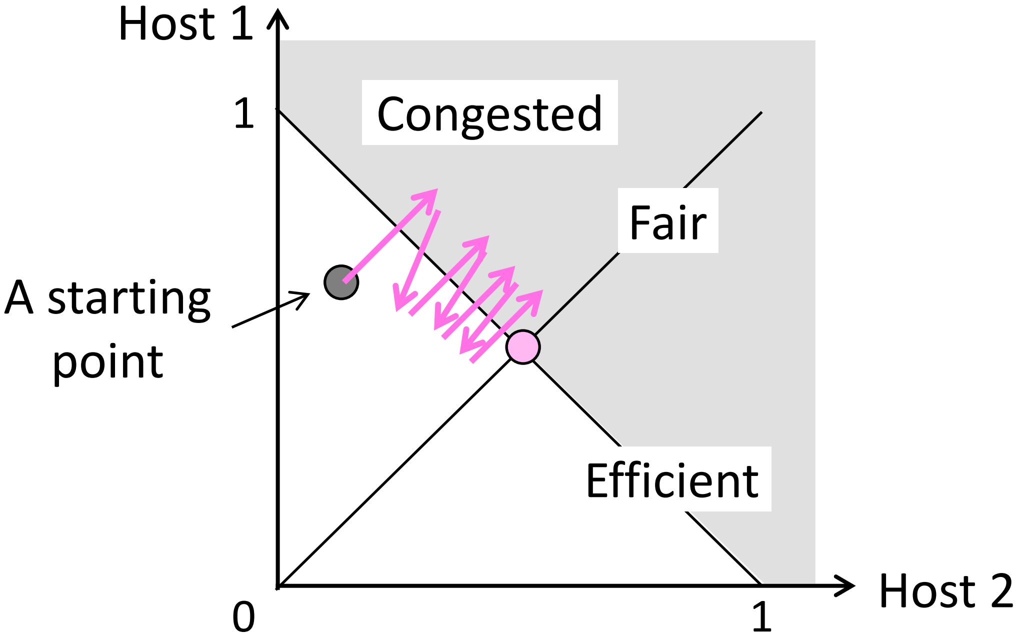
AIMD Game (4)

- Play the game!



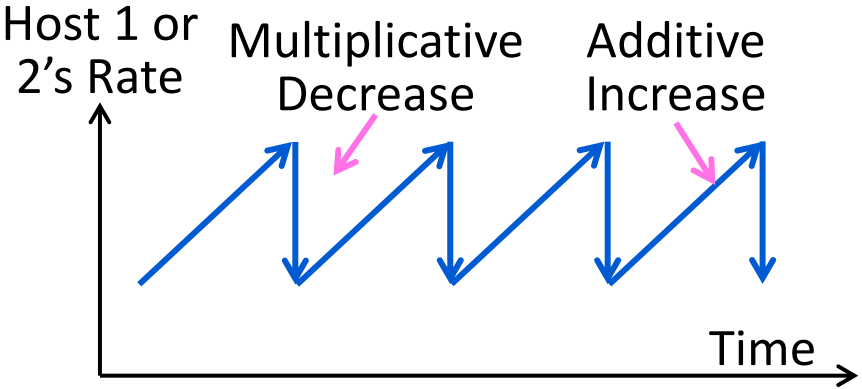
AIMD Game (5)

- Always converge to good allocation!



AIMD Sawtooth

- Produces a "sawtooth" pattern over time for rate of each host
  - This is the TCP sawtooth (later)



AIMD Properties

- Converges to an allocation that is efficient and fair when hosts run it
  - Holds for more general topologies
- Other increase/decrease control laws do not! (Try MIAD, MIMD, MIAD)
- Requires only binary feedback from the network

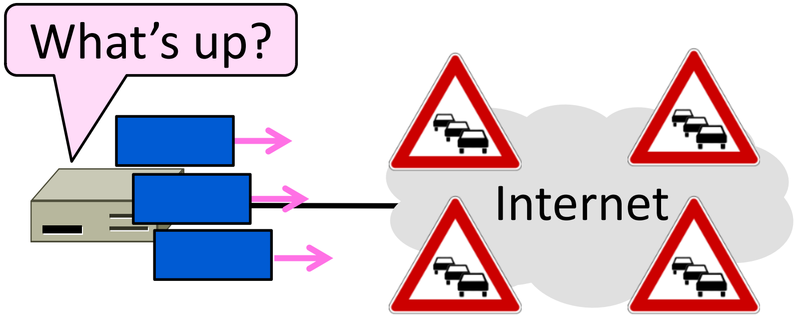
Feedback Signals

- Several possible signals, with different pros/cons
  - We'll look at classic TCP that uses packet loss as a signal

Signal	Example Protocol	Pros / Cons
Packet loss	TCP NewReno Cubic TCP (Linux)	Hard to get wrong Hear about congestion late
Packet delay	Compound TCP (Windows)	Hear about congestion early Need to infer congestion
Router indication	TCPs with Explicit Congestion Notification	Hear about congestion early Require router support

Topic

- The story of TCP congestion control
  - Collapse, control, and diversification

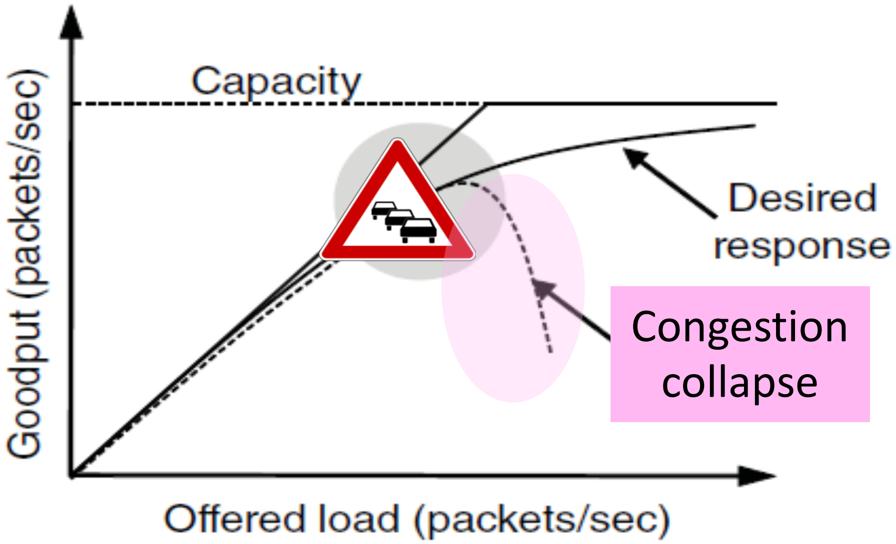


Congestion Collapse in the 1980s

- Early TCP used a fixed size sliding window (e.g., 8 packets)
  - Initially fine for reliability
- But something strange happened as the ARPANET grew
  - Links stayed busy but transfer rates fell by orders of magnitude!

Congestion Collapse (2)

- Queues became full, retransmissions clogged the network, and goodput fell



Congestion collapse

Van Jacobson (1950—)

- Widely credited with saving the Internet from congestion collapse in the late 80s
  - Introduced congestion control principles
  - Practical solutions (TCP Tahoe/Reno)
- Much other pioneering work:
  - Tools like traceroute, tcpdump, pathchar
  - IP header compression, multicast tools

TCP Tahoe/Reno

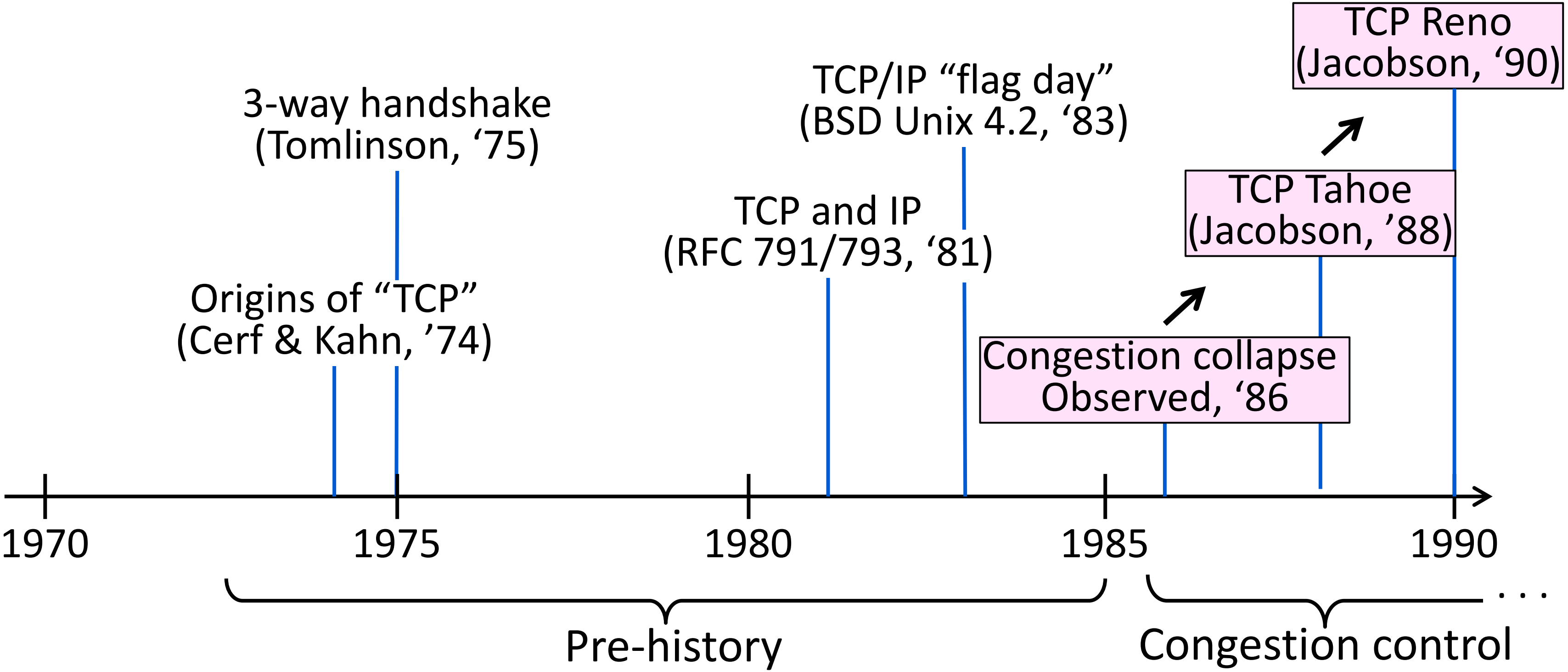
- Avoid congestion collapse without changing routers (or even receivers)
- Idea is to fix timeouts and introduce a congestion window (cwnd) over the sliding window to limit queues/loss
- TCP Tahoe/Reno implements AIMD by adapting cwnd using packet loss as the network feedback signal

TCP Tahoe/Reno (2)

- TCP behaviors we will study:
  - ACK clocking
  - Adaptive timeout (mean and variance)
  - Slow-start
  - Fast Retransmission
  - Fast Recovery

- Together, they implement AIMD

TCP Timeline



3-way handshake
(Tomlinson, '75)

Origins of "TCP"
(Cerf & Kahn, '74)

TCP and IP
(RFC 791/793, '81)

TCP/IP "flag day"
(BSD Unix 4.2, '83)

Congestion collapse
Observed, '86

TCP Tahoe
(Jacobson, '88)

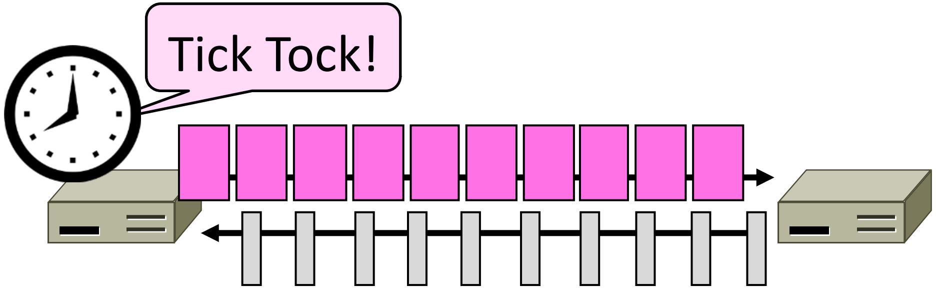
TCP Reno
(Jacobson, '90)

Pre-history

Congestion control

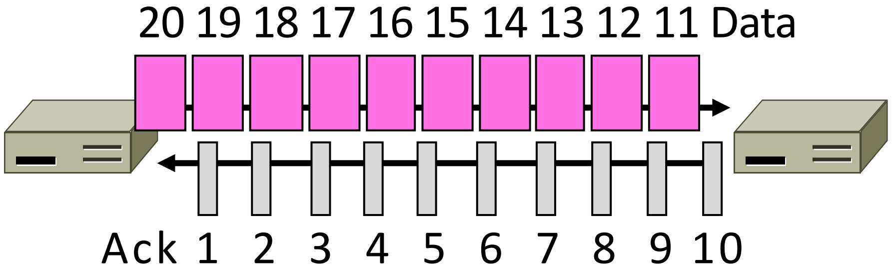
Topic

- The self-clocking behavior of sliding windows, and how it is used by TCP
  - The "ACK clock"



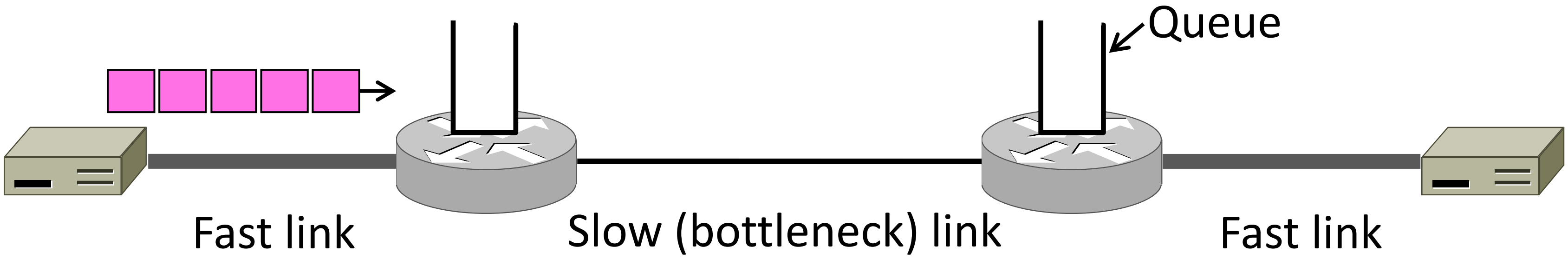
Sliding Window ACK Clock

- Each in-order ACK advances the sliding window and lets a new segment enter the network
  - ACKs “clock” data segments



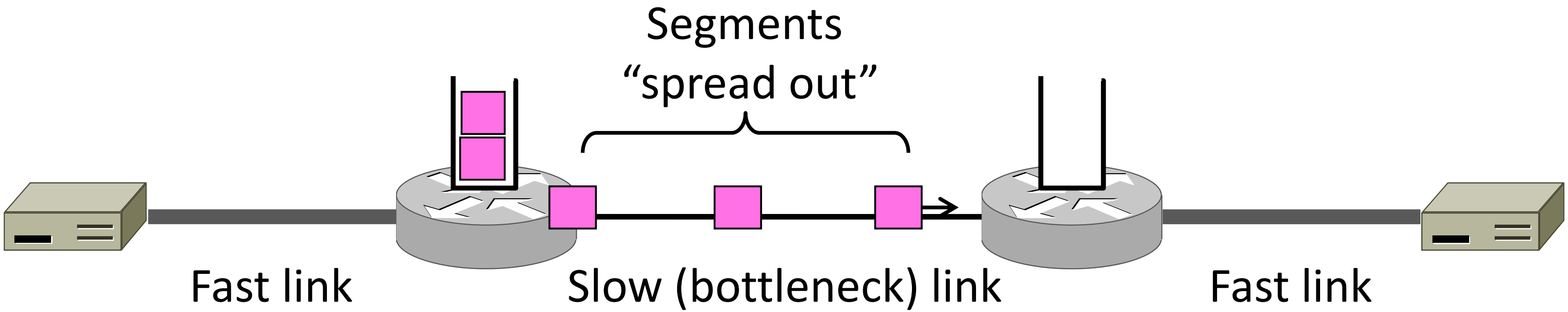
Benefit of ACK Clocking

- Consider what happens when sender injects a burst of segments into the network



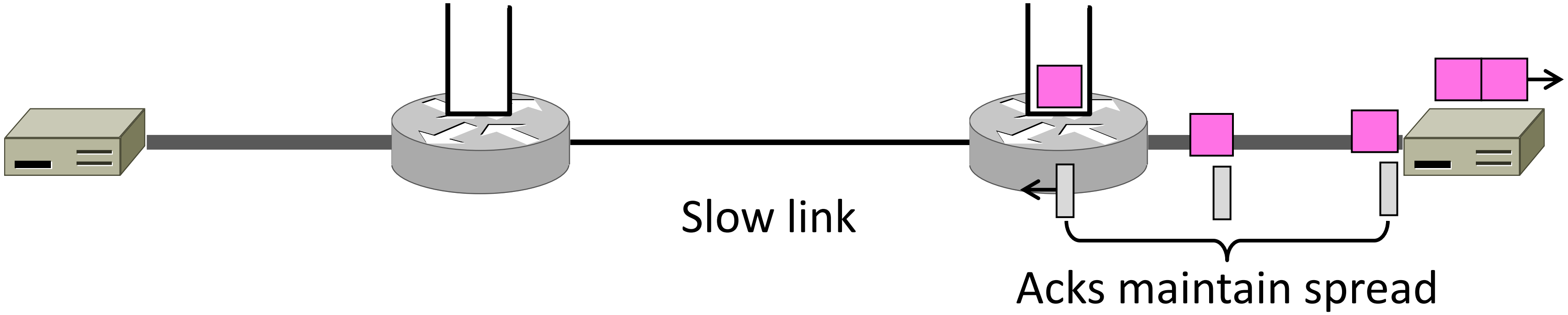
Benefit of ACK Clocking (2)

- Segments are buffered and spread out on slow link



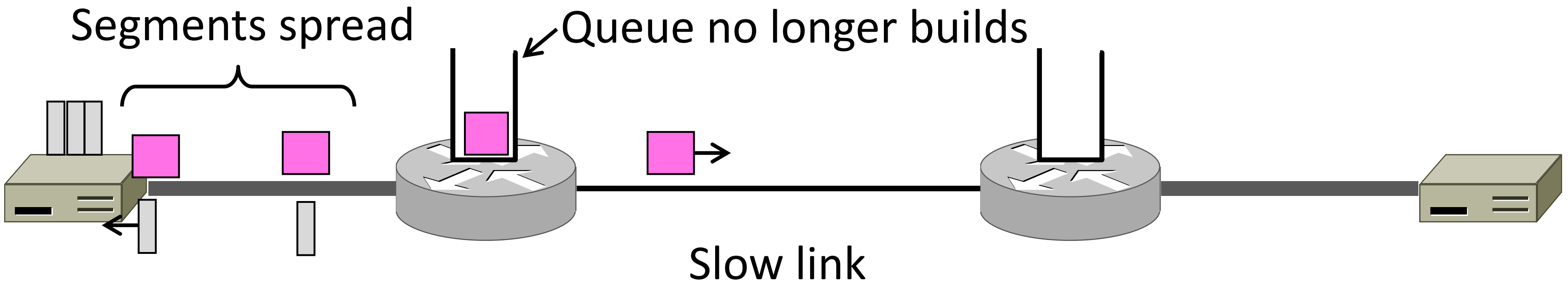
Benefit of ACK Clocking (3)

- ACKs maintain the spread back to the original sender



Benefit of ACK Clocking (4)

- Sender clocks new segments with the spread
  - Now sending at the bottleneck link without queuing!



Benefit of ACK Clocking (4)

- Helps the network run with low levels of loss and delay!

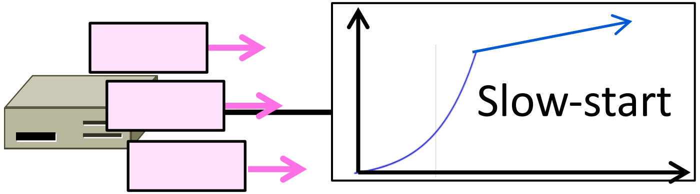
- The network has smoothed out the burst of data segments
- ACK clock transfers this smooth timing back to the sender
- Subsequent data segments are not sent in bursts so do not queue up in the network

TCP Uses ACK Clocking

- TCP uses a sliding window because of the value of ACK clocking
- Sliding window controls how many segments are inside the network
  - Called the congestion window, or cwnd
  - Rate is roughly cwnd/RTT
- TCP only sends small bursts of segments to let the network keep the traffic smooth

Topic

- How TCP implements AIMD, part 1
  - “Slow start” is a component of the AI portion of AIMD



Recall

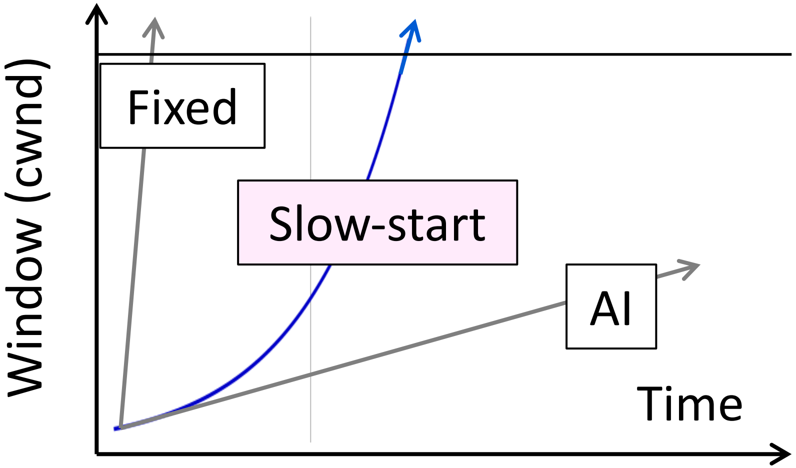
- We want TCP to follow an AIMD control law for a good allocation
- Sender uses a congestion window or cwnd to set its rate (≈cwnd/RTT)
- Sender uses packet loss as the network congestion signal
- Need TCP to work across a very large range of rates and RTTs

TCP Startup Problem

- We want to quickly near the right rate, $\text{cwnd}_{\text{IDEAL}}$, but it varies greatly
  - Fixed sliding window doesn’t adapt and is rough on the network (loss!)
  - AI with small bursts adapts cwnd gently to the network, but might take a long time to become efficient

Slow-Start Solution

- Start by doubling cwnd every RTT
  - Exponential growth (1, 2, 4, 8, 16, …)
  - Start slow, quickly reach large values

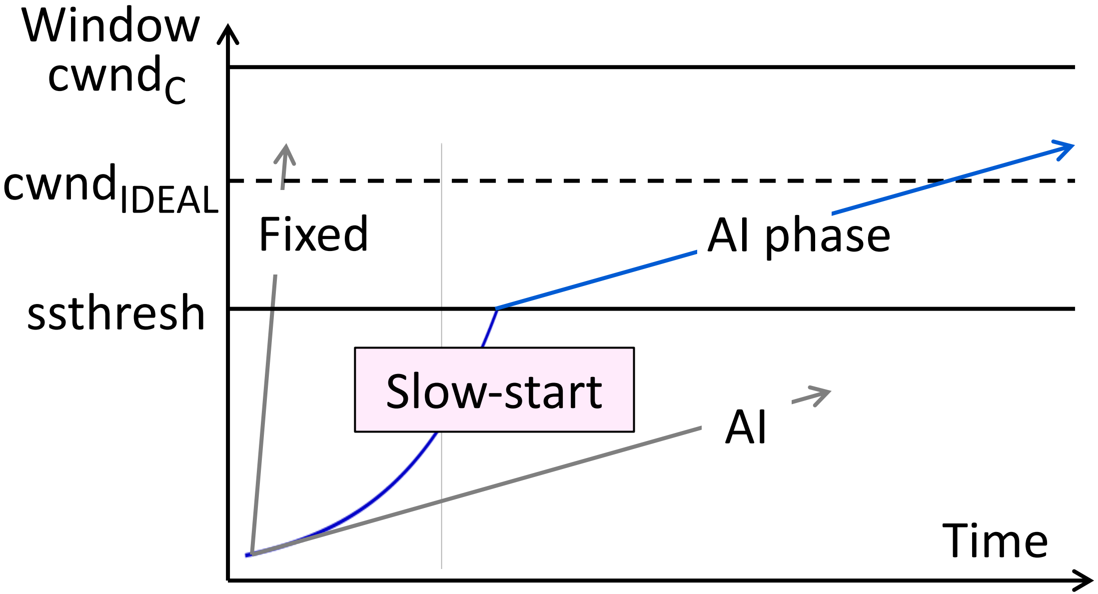


Slow-Start Solution (2)

- Eventually packet loss will occur when the network is congested
  - Loss timeout tells us cwnd is too large
  - Next time, switch to AI beforehand
  - Slowly adapt cwnd near right value
- In terms of cwnd:
  - Expect loss for $\text{cwnd}_C \approx 2\text{BD}+\text{queue}$
  - Use ssthresh = $\text{cwnd}_C/2$ to switch to AI

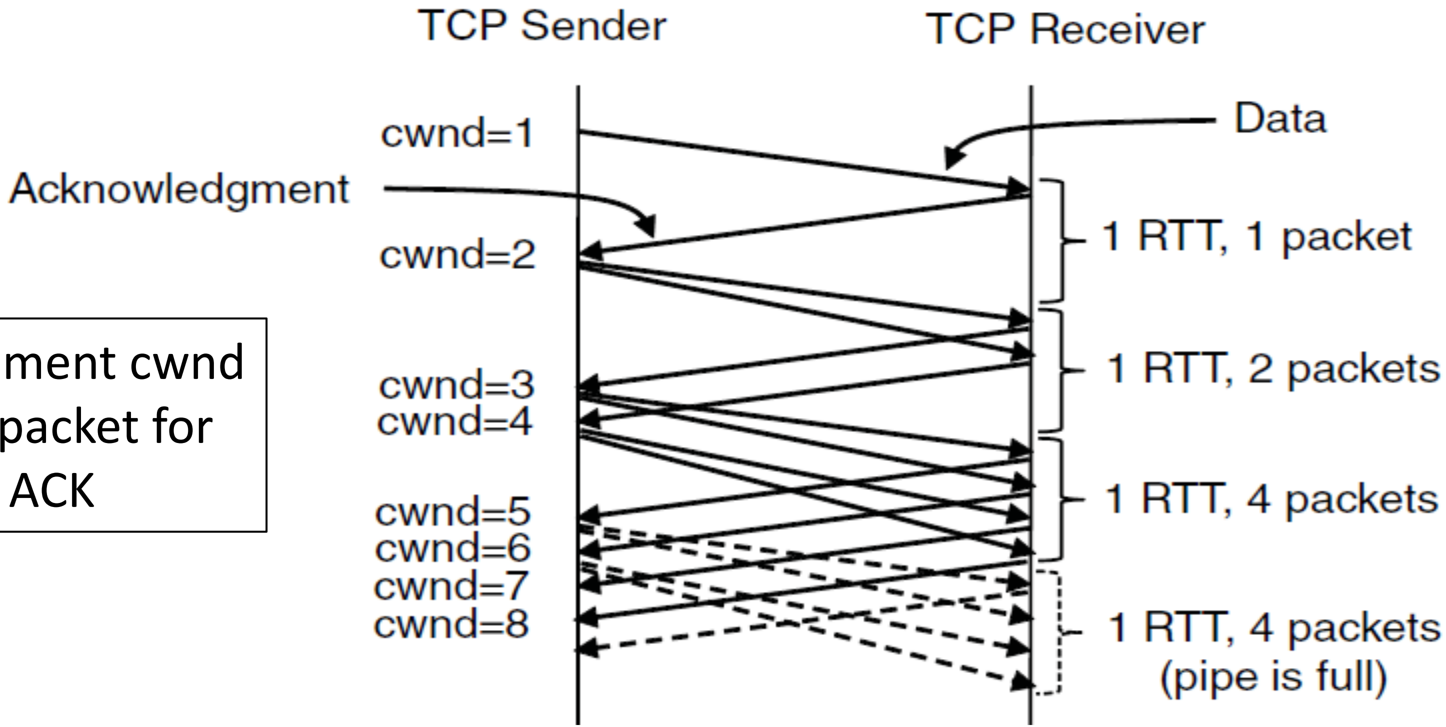
Slow-Start Solution (3)

- Combined behavior, after first time
  - Most time spend near right value

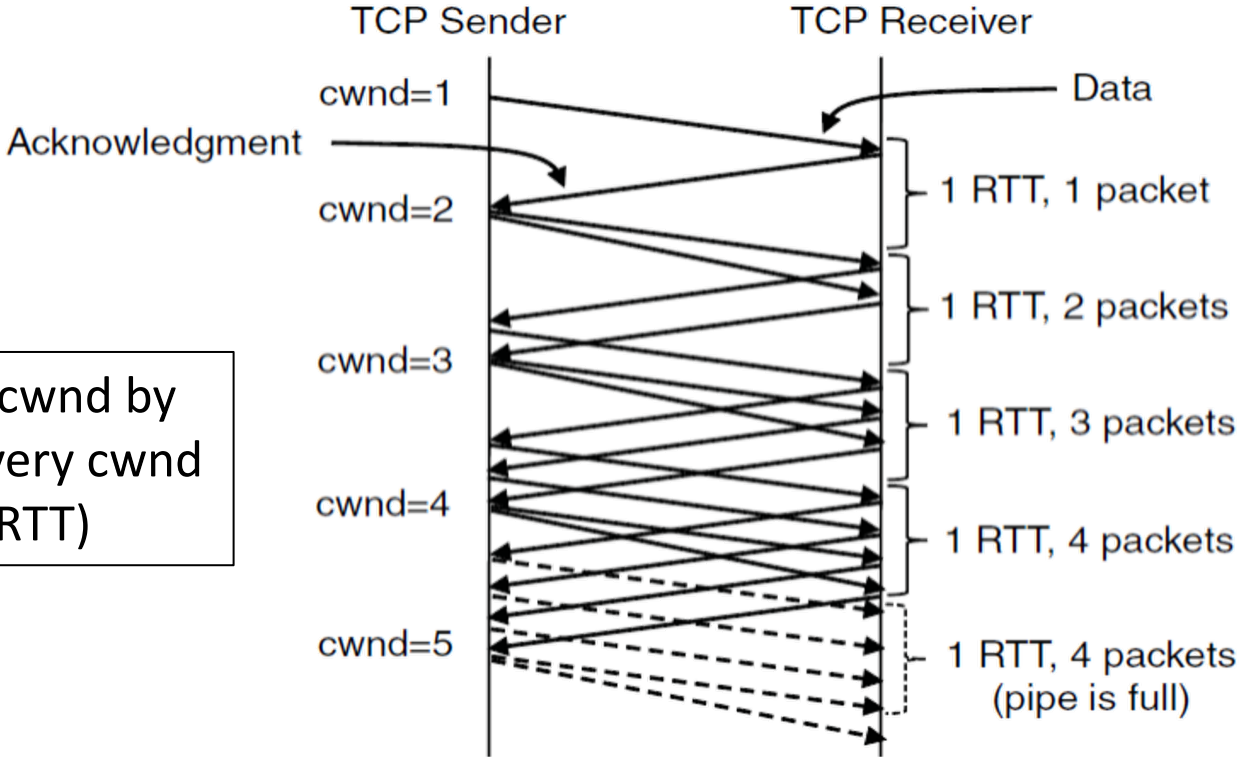


Slow-Start (Doubling) Timeline

Increment cwnd by 1 packet for each ACK



Additive Increase Timeline



Increment cwnd by 1 packet every cwnd ACKs (or 1 RTT)

TCP Tahoe (Implementation)

- Initial slow-start (doubling) phase
  - Start with cwnd = 1 (or small value)
  - cwnd += 1 packet per ACK

- Later Additive Increase phase
  - cwnd += 1/cwnd packets per ACK
  - Roughly adds 1 packet per RTT

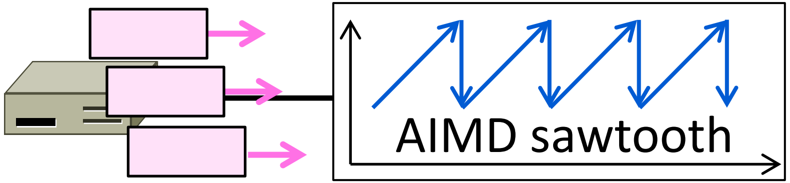
- Switching threshold (initially infinity)
  - Switch to AI when cwnd > ssthresh
  - Set ssthresh = cwnd/2 after loss
  - Begin with slow-start after timeout

Timeout Misfortunes

- Why do a slow-start after timeout?
  - Instead of MD cwnd (for AIMD)
- Timeouts are sufficiently long that the ACK clock will have run down
  - Slow-start ramps up the ACK clock
- We need to detect loss before a timeout to get to full AIMD
  - Done in TCP Reno (next time)

Topic

- How TCP implements AIMD, part 2
  - "Fast retransmit" and "fast recovery" are the MD portion of AIMD



Recall

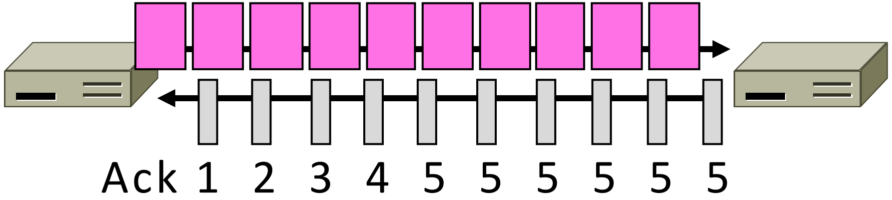
- We want TCP to follow an AIMD control law for a good allocation
- Sender uses a congestion window or cwnd to set its rate (≈cwnd/RTT)
- Sender uses slow-start to ramp up the ACK clock, followed by Additive Increase
- But after a timeout, sender slow-starts again with cwnd=1 (as it no ACK clock)

Inferring Loss from ACKs

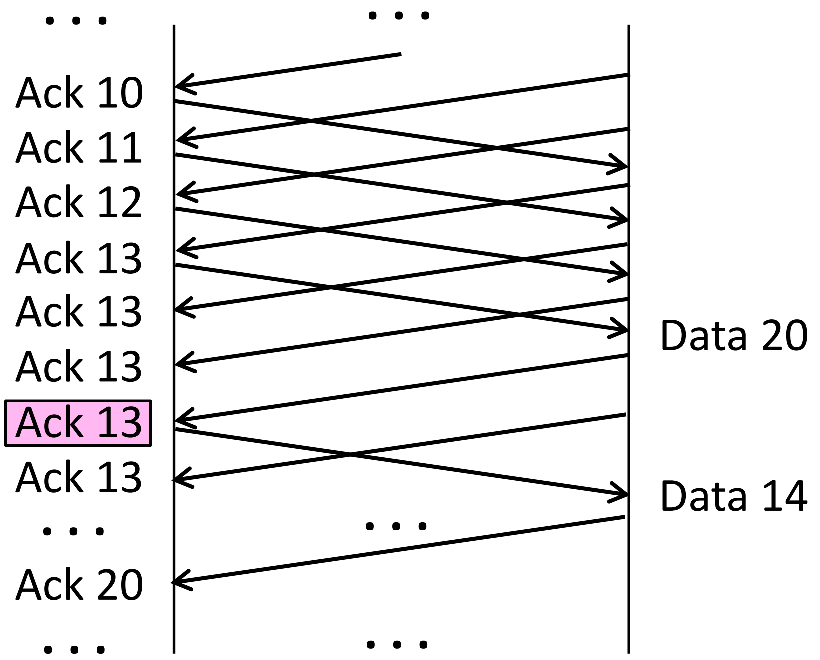
- TCP uses a cumulative ACK
  - Carries highest in-order seq. number
  - Normally a steady advance
- Duplicate ACKs give us hints about what data hasn't arrived
  - Tell us some new data did arrive, but it was not next segment
  - Thus the next segment may be lost

Fast Retransmit

- Treat three duplicate ACKs as a loss
  - Retransmit next expected segment
  - Some repetition allows for reordering, but still detects loss quickly



Fast Retransmit (2)



Data 14 was lost earlier, but got 15 to 20

Third duplicate ACK, so send 14

Retransmission fills in the hole at 14

ACK jumps after loss is repaired

Fast Retransmit (3)

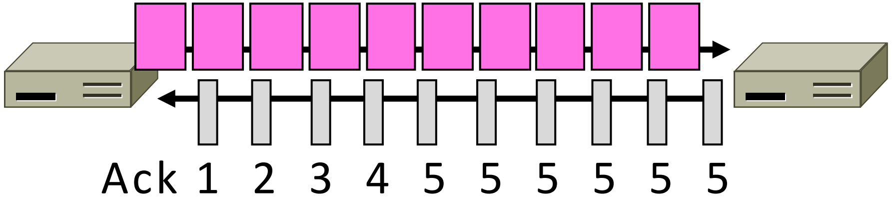
- It can repair single segment loss quickly, typically before a timeout
- However, we have quiet time at the sender/receiver while waiting for the ACK to jump
- And we still need to MD cwnd …

Inferring Non-Loss from ACKs

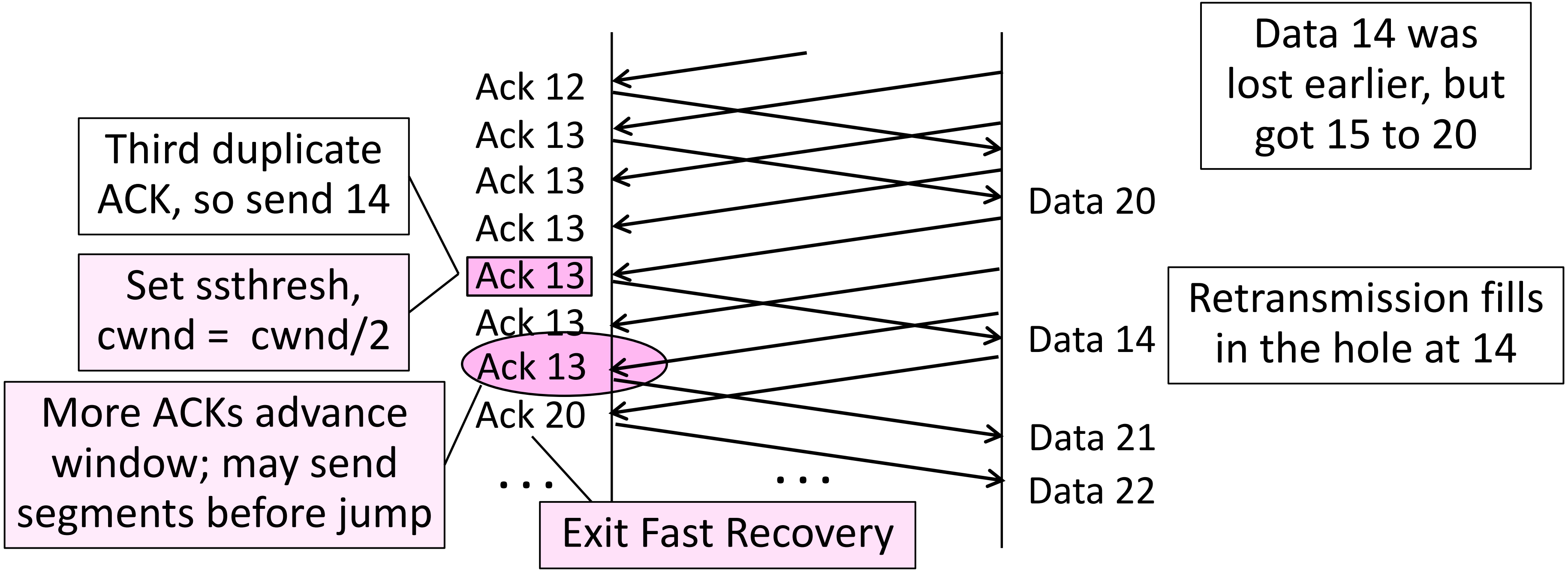
- Duplicate ACKs also give us hints about what data has arrived
  - Each new duplicate ACK means that some new segment has arrived
  - It will be the segments after the loss
  - Thus advancing the sliding window will not increase the number of segments stored in the network

Fast Recovery

- First fast retransmit, and MD cwnd
- Then pretend further duplicate ACKs are the expected ACKs
  - Lets new segments be sent for ACKs
  - Reconcile views when the ACK jumps



Fast Recovery (2)



Data 14 was lost earlier, but got 15 to 20

Third duplicate ACK, so send 14

Set ssthresh, cwnd = cwnd/2

Retransmission fills in the hole at 14

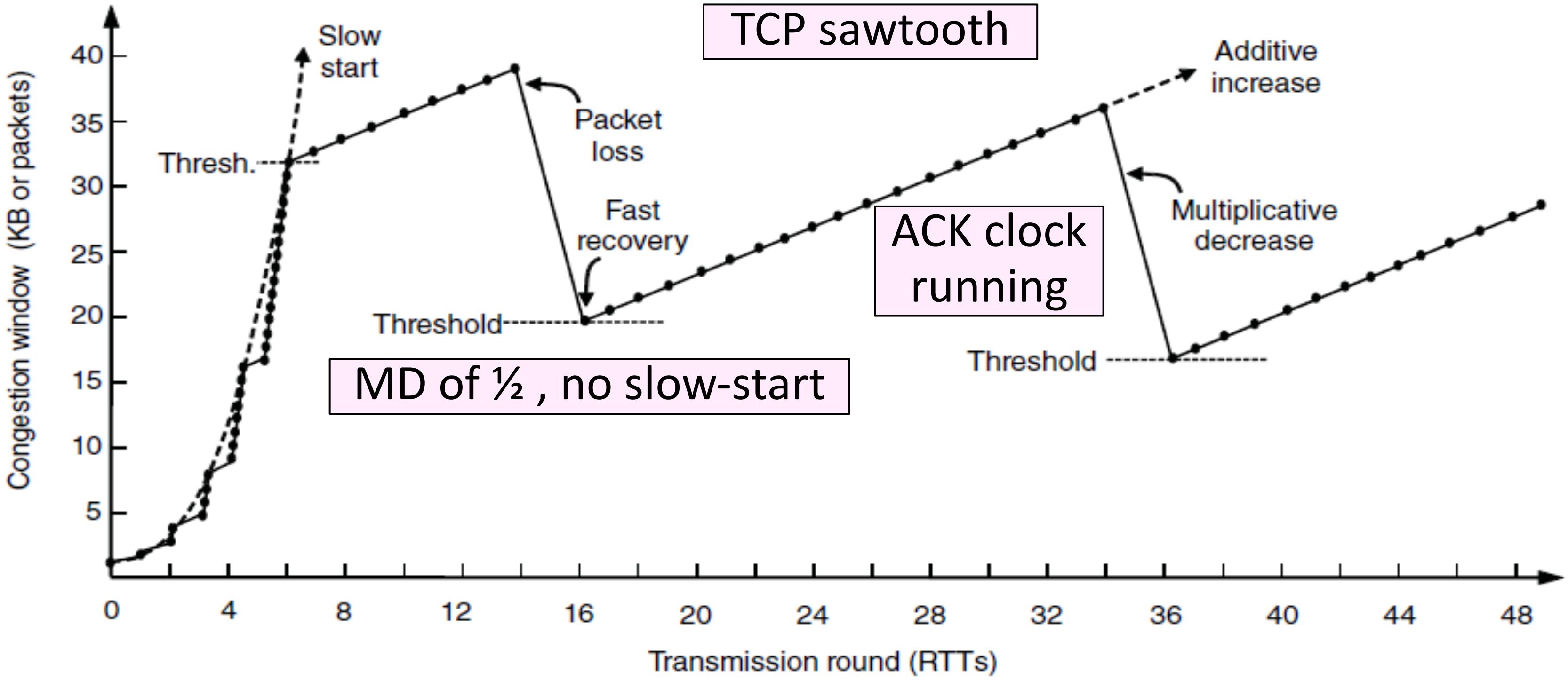
More ACKs advance window; may send segments before jump

Exit Fast Recovery

Fast Recovery (3)

- With fast retransmit, it repairs a single segment loss quickly and keeps the ACK clock running
- This allows us to realize AIMD
  - No timeouts or slow-start after loss, just continue with a smaller cwnd
- TCP Reno combines slow-start, fast retransmit and fast recovery
  - Multiplicative Decrease is ½

TCP Reno



TCP sawtooth

ACK clock running

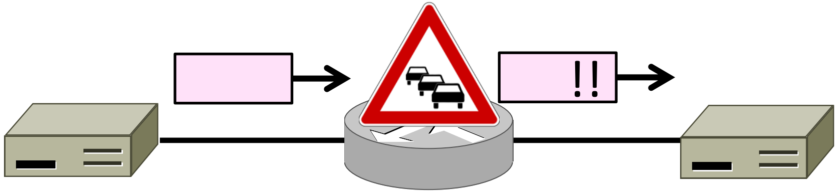
MD of ½ , no slow-start

TCP Reno, NewReno, and SACK

- Reno can repair one loss per RTT
  - Multiple losses cause a timeout
- NewReno further refines ACK heuristics
  - Repairs multiple losses without timeout
- SACK is a better idea
  - Receiver sends ACK ranges so sender can retransmit without guesswork

Topic

- How routers can help hosts to avoid congestion
  - Explicit Congestion Notification



Congestion Avoidance vs. Control

- Classic TCP drives the network into congestion and then recovers
  - Needs to see loss to slow down
- Would be better to use the network but avoid congestion altogether!
  - Reduces loss and delay
- But how can we do this?

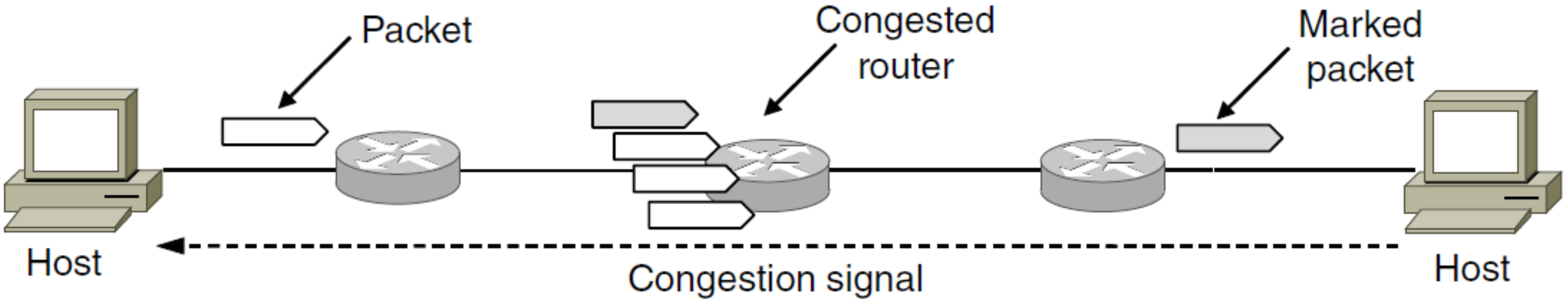
Feedback Signals

- Delay and router signals can let us avoid congestion

Signal	Example Protocol	Pros / Cons
Packet loss	Classic TCP Cubic TCP (Linux)	Hard to get wrong Hear about congestion late
Packet delay	Compound TCP (Windows)	Hear about congestion early Need to infer congestion
Router indication	TCPs with Explicit Congestion Notification	Hear about congestion early Require router support

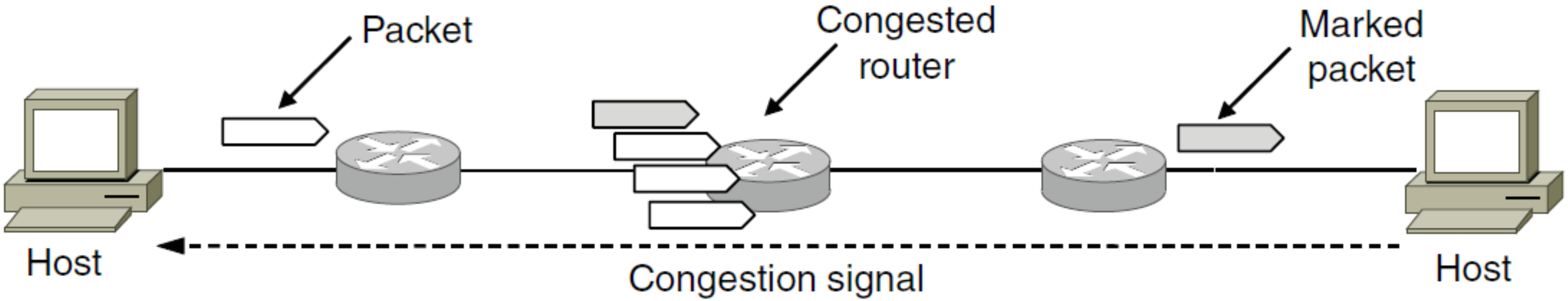
ECN (Explicit Congestion Notification)

- Router detects the onset of congestion via its queue
  - When congested, it marks affected packets (IP header)



ECN (2)

- Marked packets arrive at receiver; treated as loss
  - TCP receiver reliably informs TCP sender of the congestion



ECN (3)

- Advantages:
  - Routers deliver clear signal to hosts
  - Congestion is detected early, no loss
  - No extra packets need to be sent
- Disadvantages:
  - Routers and hosts must be upgraded