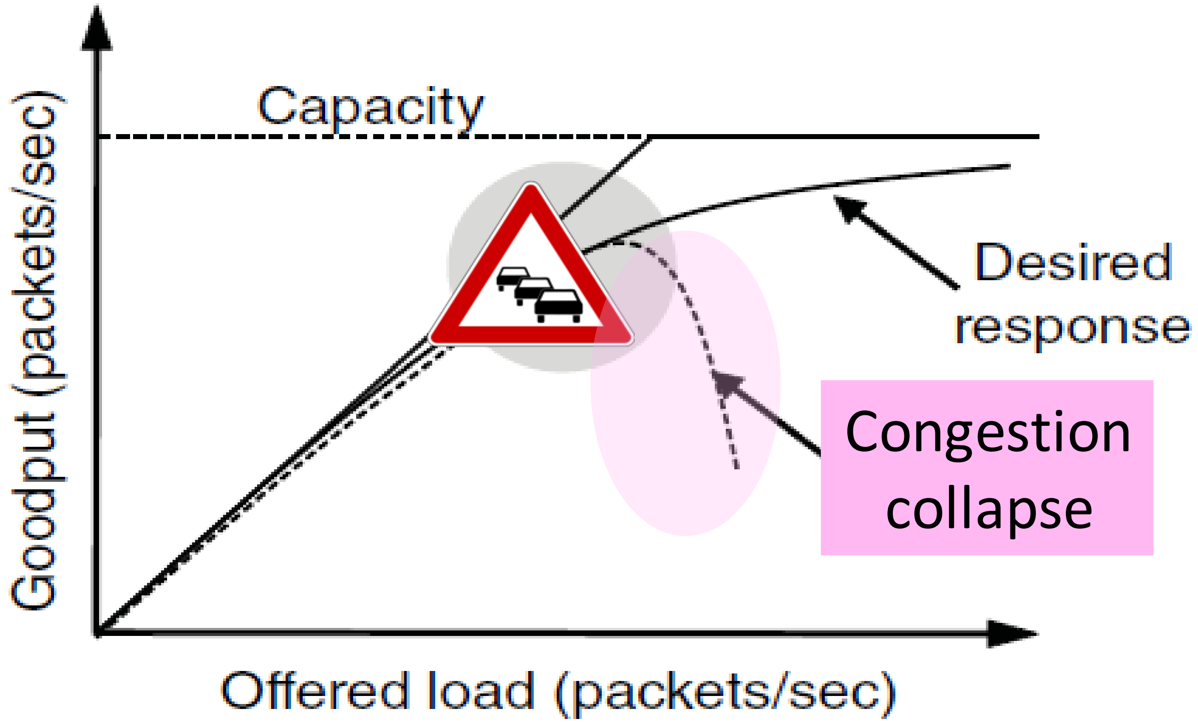
Congestion Collapse

Congestion Collapse in the 1980s

- Early TCP used fixed size window (e.g., 8 packets)
  - Initially fine for reliability
- But something 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 <u>goodput</u> fell



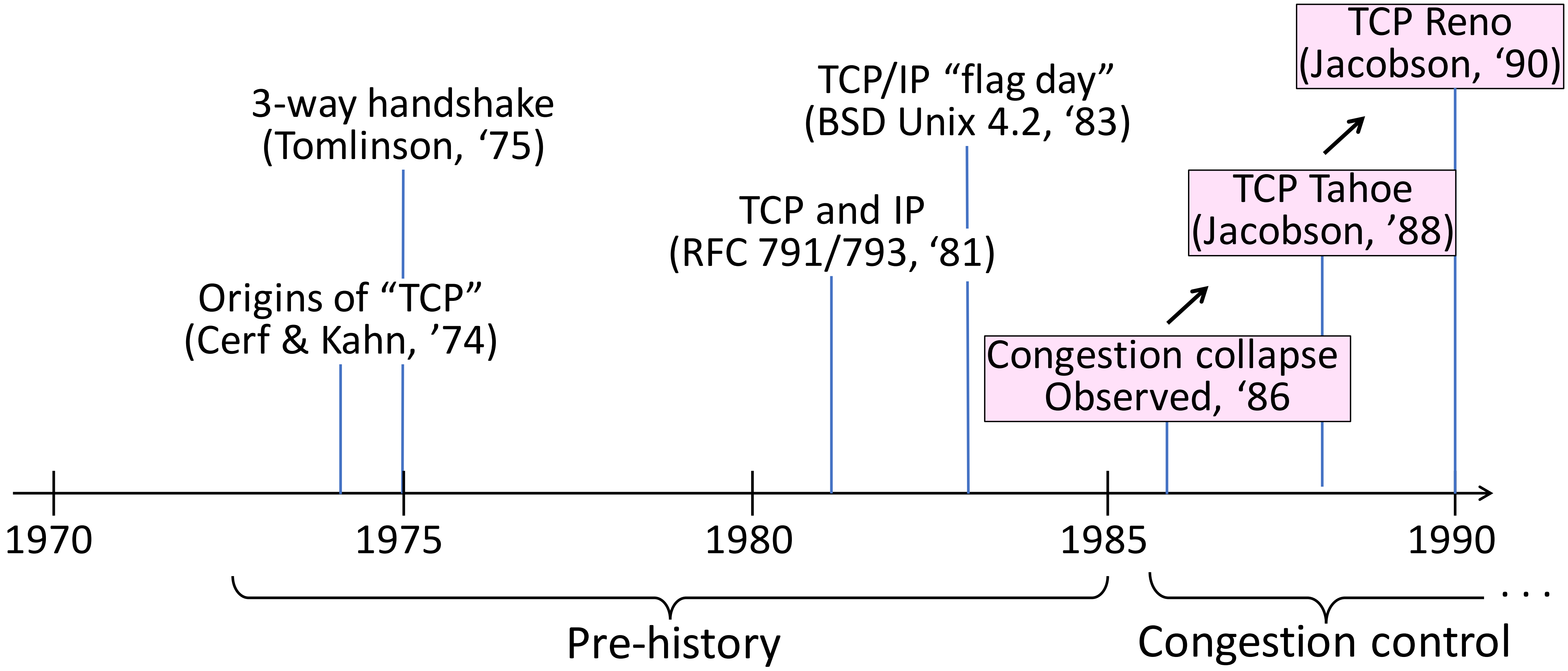
Van Jacobson (1950—)

- 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

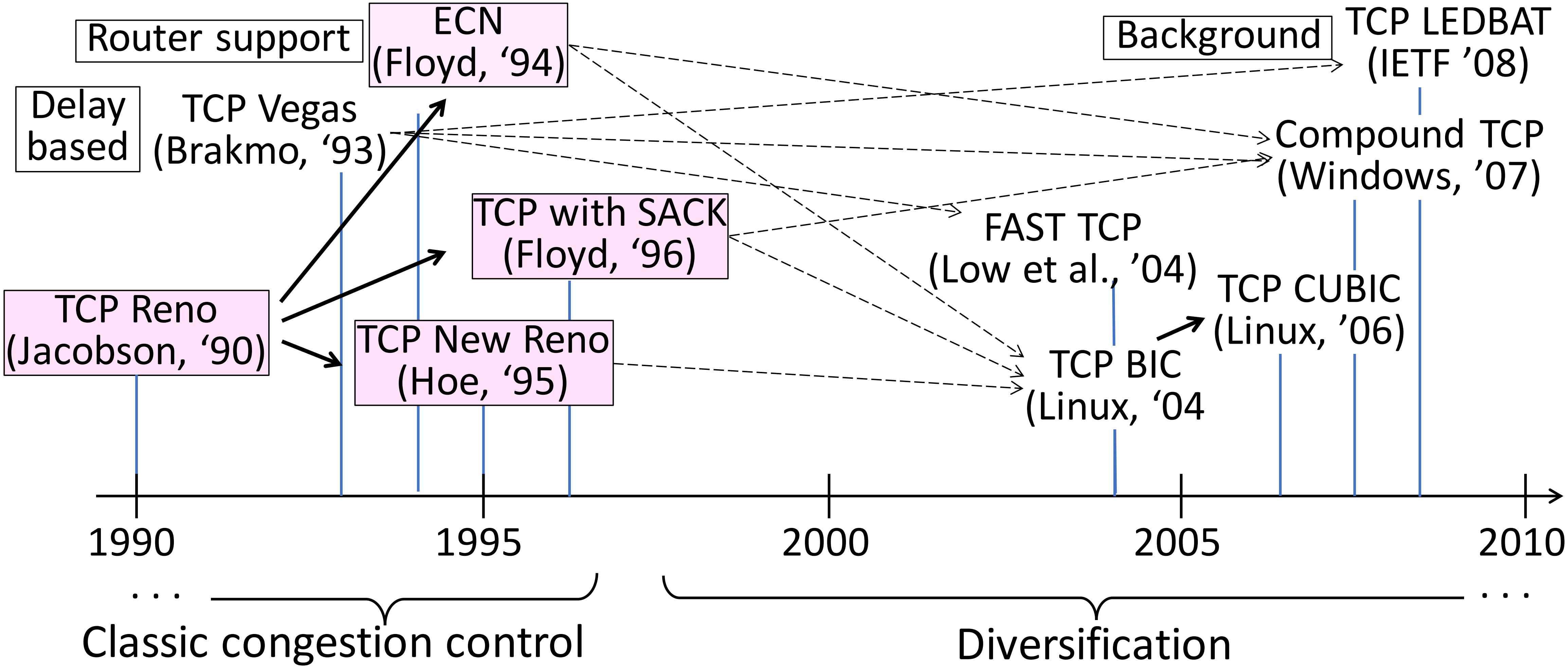
TCP Tahoe/Reno

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

TCP Timeline



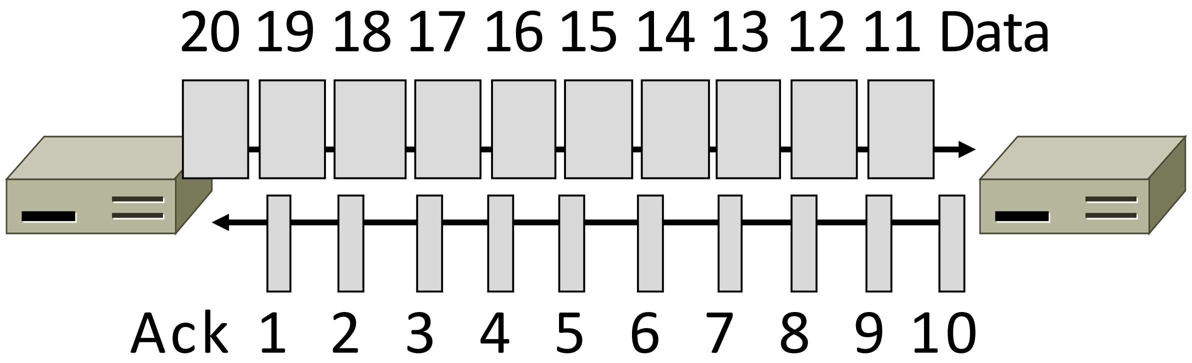
TCP Timeline



ACK Clocking

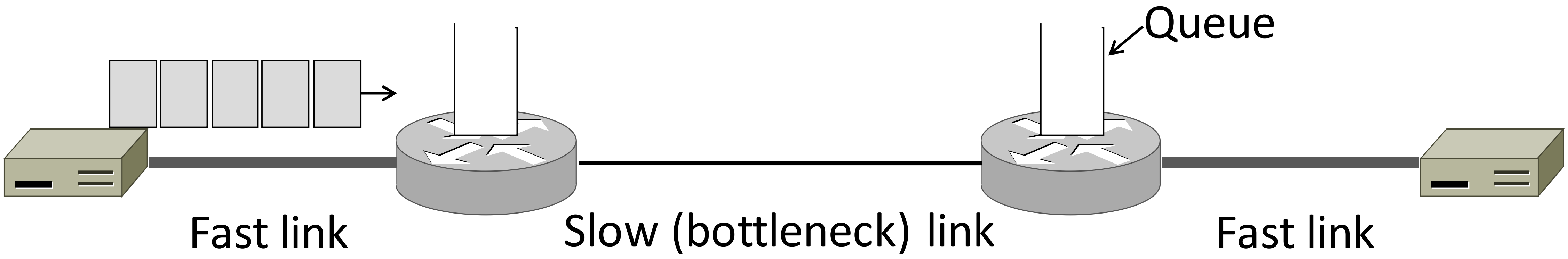
Sliding Window ACK Clock

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



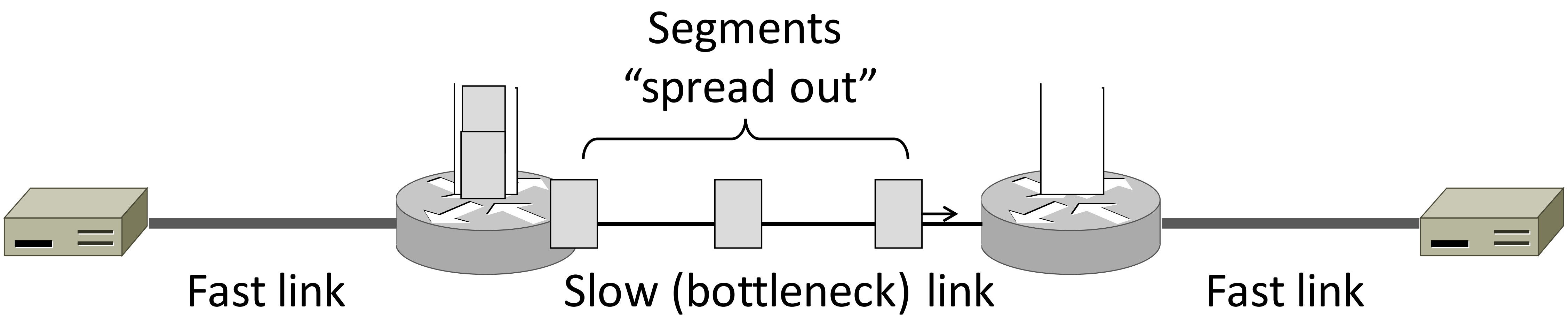
Benefit of ACK Clocking

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



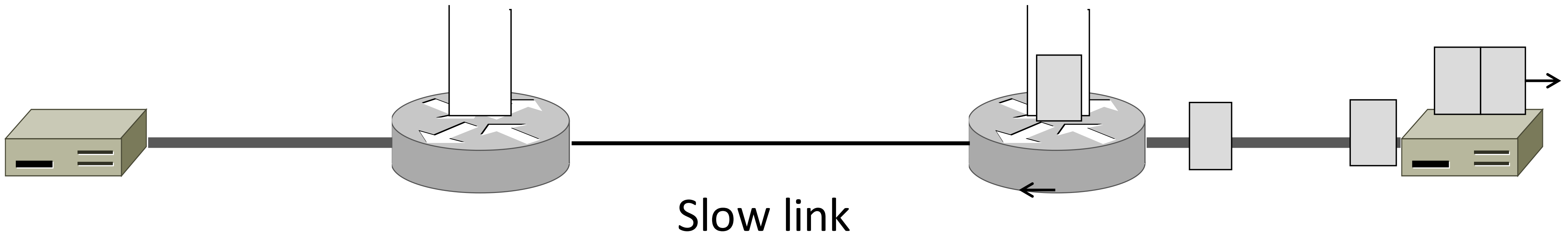
Benefit of ACK Clocking

- Segments are buffered and spread out on slow link



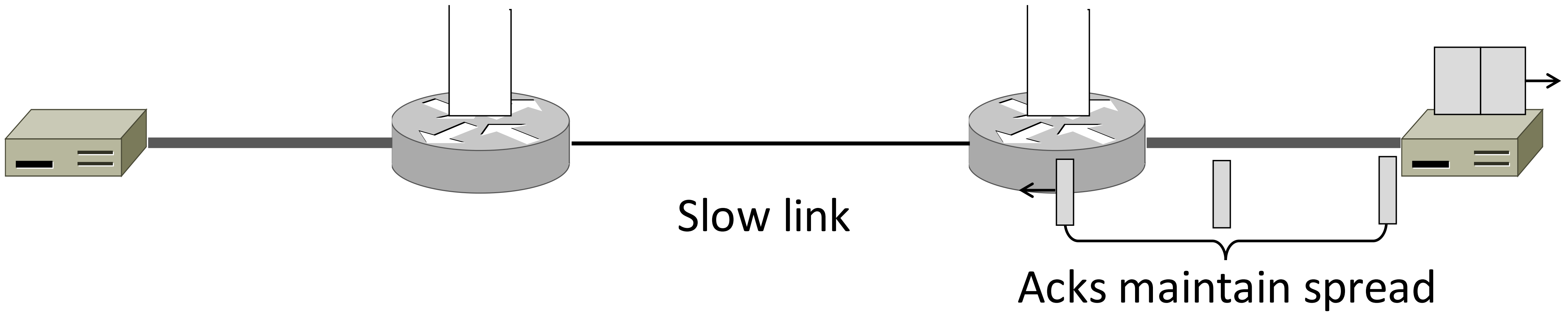
Benefit of ACK Clocking

- Segments maintain the spread up to the destination



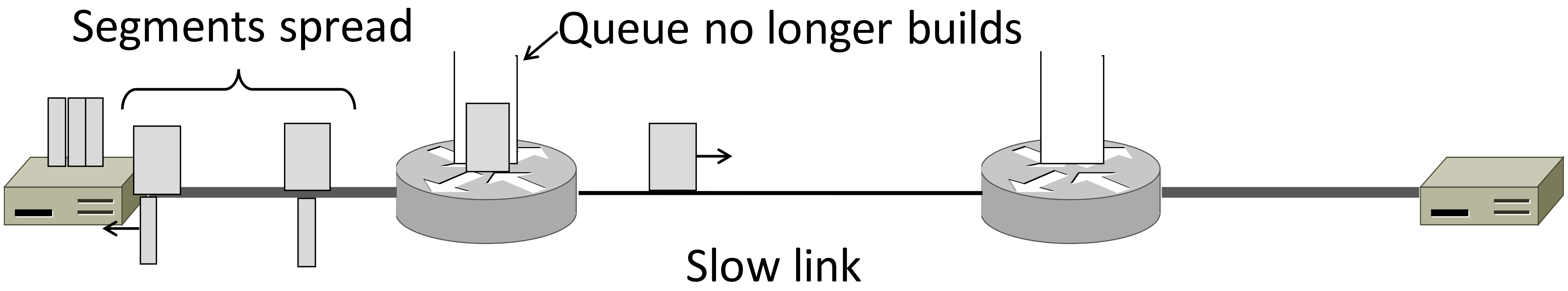
Benefit of ACK Clocking

- ACKs repeat the spread back to the sender



Benefit of ACK Clocking

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



Benefit of ACK Clocking

- Helps run with low levels of loss and delay!
- The network smooths out the burst of data segments
- ACK clock transfers this smooth timing back to sender
  - “just happens”
- Subsequent data segments are not sent in bursts so do not queue up in the network

TCP Uses ACK Clocking

- TCP manages offered load using a sliding window
- Sliding window controls how many segments are inside the network
  - Called the congestion window, or cwnd
  - (As always, rate is roughly cwnd/RTT)
- TCP sends only small bursts of segments to let the network keep the traffic smooth

TCP Slow Start

Practical AIMD

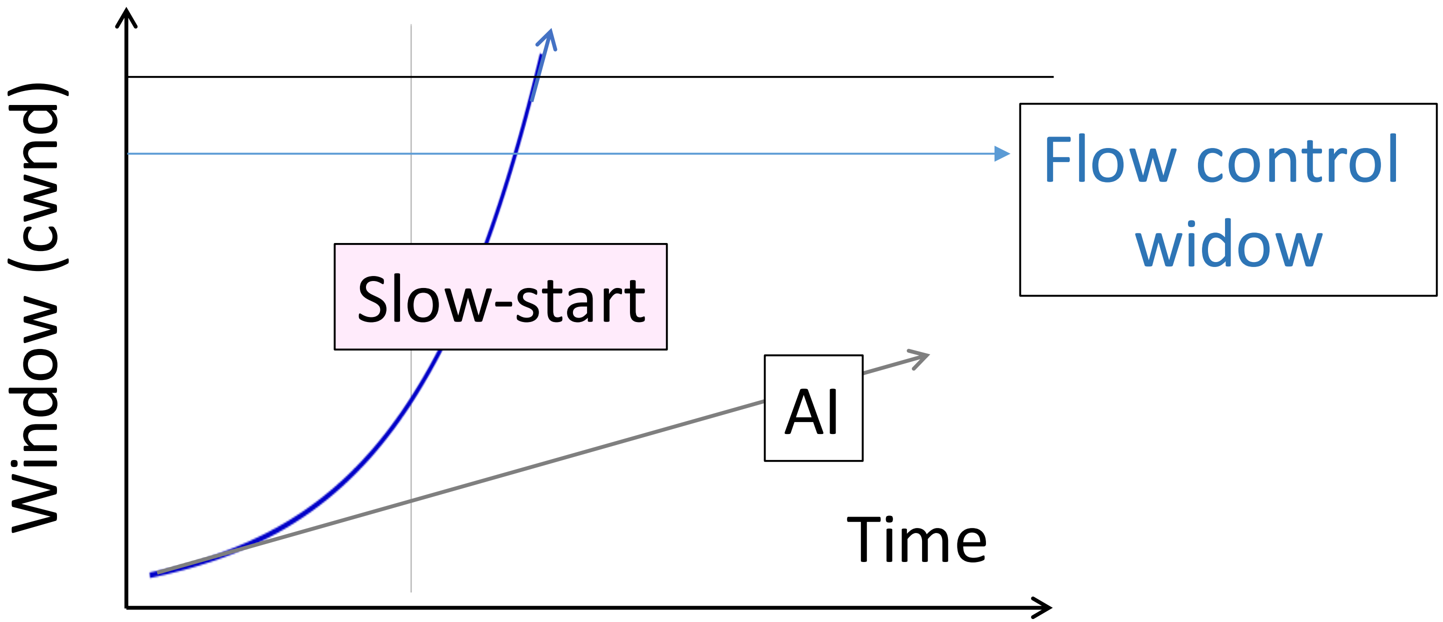
- We want TCP to follow an AIMD control law for a good allocation
- Sender uses a congestion window or cwnd to set its rate (≈cwnd/RTT)
- Sender uses loss as network congestion signal
- Need TCP to work across a very large range of rates and RTTs

TCP Startup Problem

- We want to quickly near the right rate, $cwnd_{IDEAL}$, but it varies greatly
  - Fixed sliding window doesn’t adapt and is rough on the network (loss!)
  - Additive Increase 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

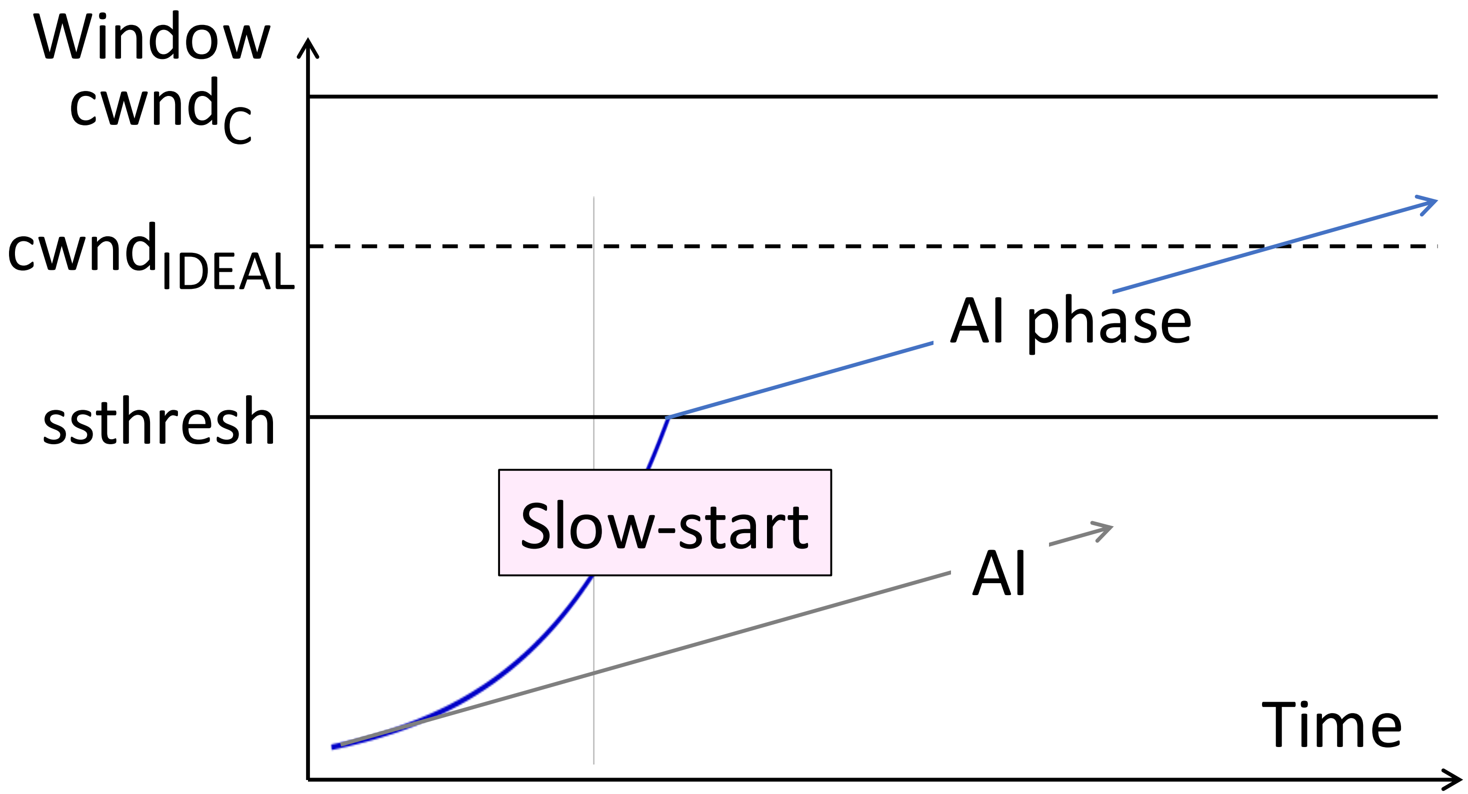


Slow-Start Solution

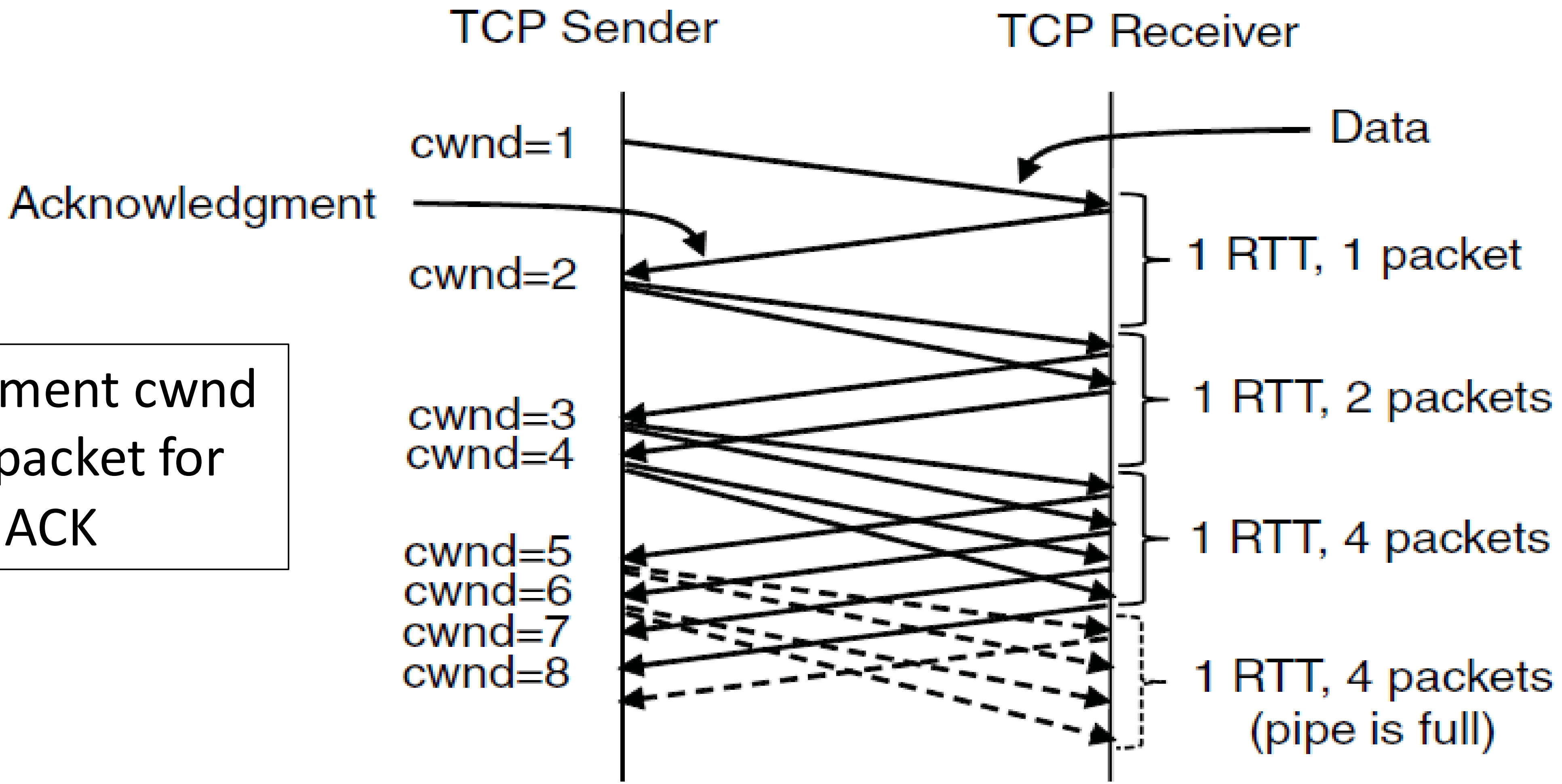
- Start very conservatively and ramp up quickly
- Eventually packet loss will occur when the network is congested
  - Loss timeout tells us cwnd is too large
  - Next time, switch to AI beforehand
  - Slowly adapt cwnd near right value
- In terms of cwnd:
  - Expect loss for $cwnd_C \approx 2BD$+queue
  - Use ssthresh = $cwnd_C/2$ to switch to AI

Slow-Start Solution (3)

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

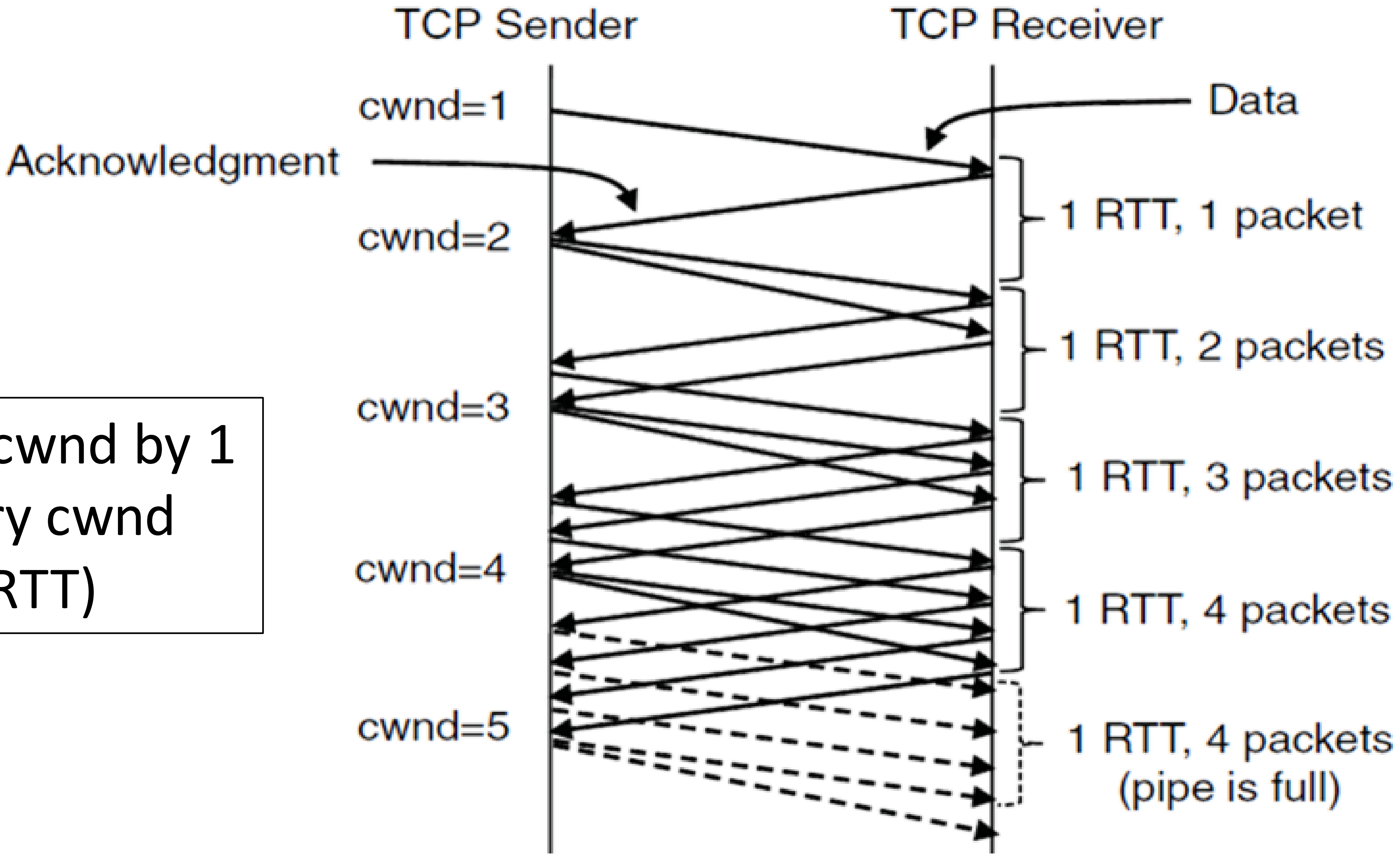


Slow-Start (Doubling) Timeline



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

TCP Fast Recovery

Practical AIMD

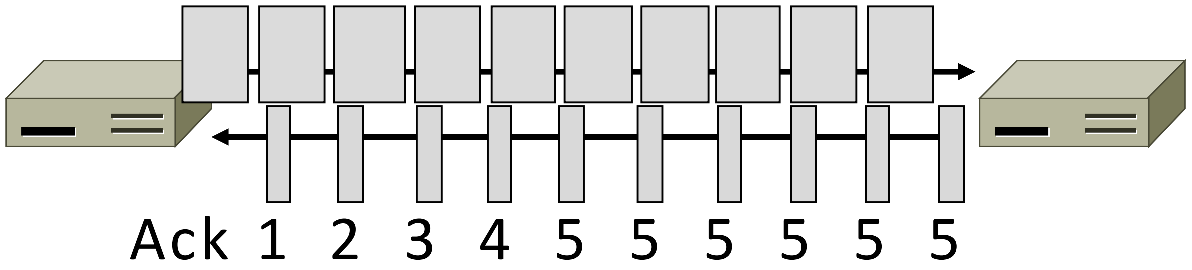
- We want TCP to follow an AIMD control law for a good allocation
- Sender uses a congestion window or cwnd to set its rate (≈cwnd/RTT)
- Sender uses slow-start to ramp up the ACK clock, followed by Additive Increase
- But after a timeout, sender slow-starts again with cwnd=1 (as it has no ACK clock)

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

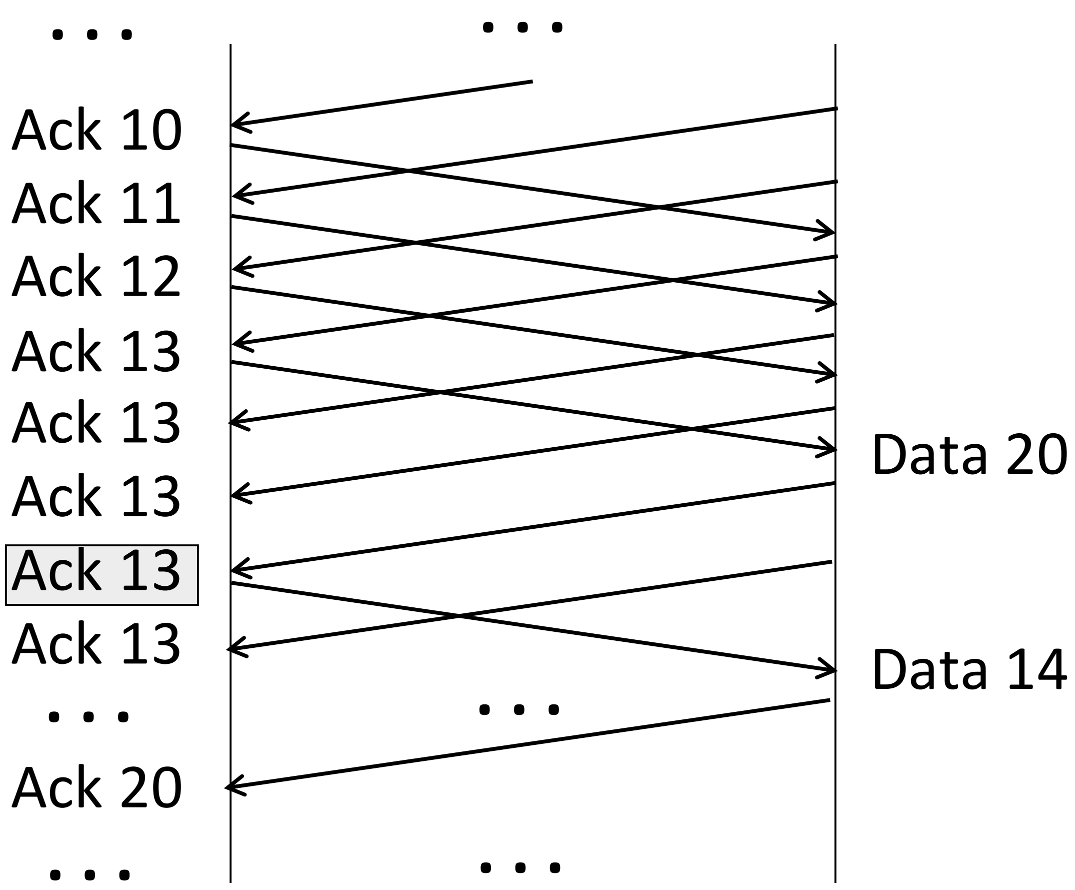
- Treat three duplicate ACKs as a loss signal
  - Retransmit next expected segment
  - Some repetition allows for reordering, but still detects loss quickly



Fast Retransmit

Third duplicate ACK, so send 14

ACK jumps after loss is repaired



Data 14 was lost earlier, but got 15 to 20

Retransmission fills in the hole at 14

Fast Retransmit

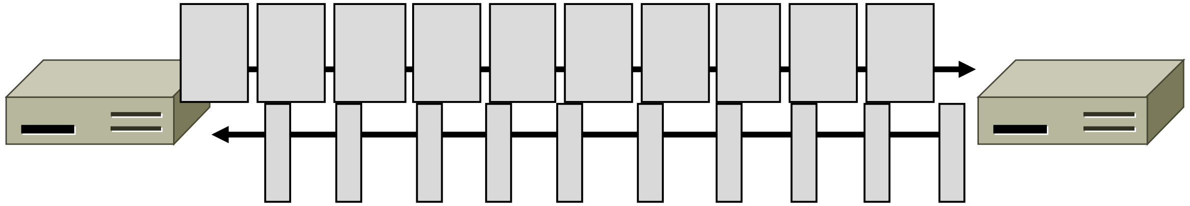
- It can repair single segment loss quickly, typically before a timeout
- However, we have quiet time at the sender/receiver while waiting for the ACK to jump
- And we still need to MD cwnd …

Inferring Non-Loss from ACKs

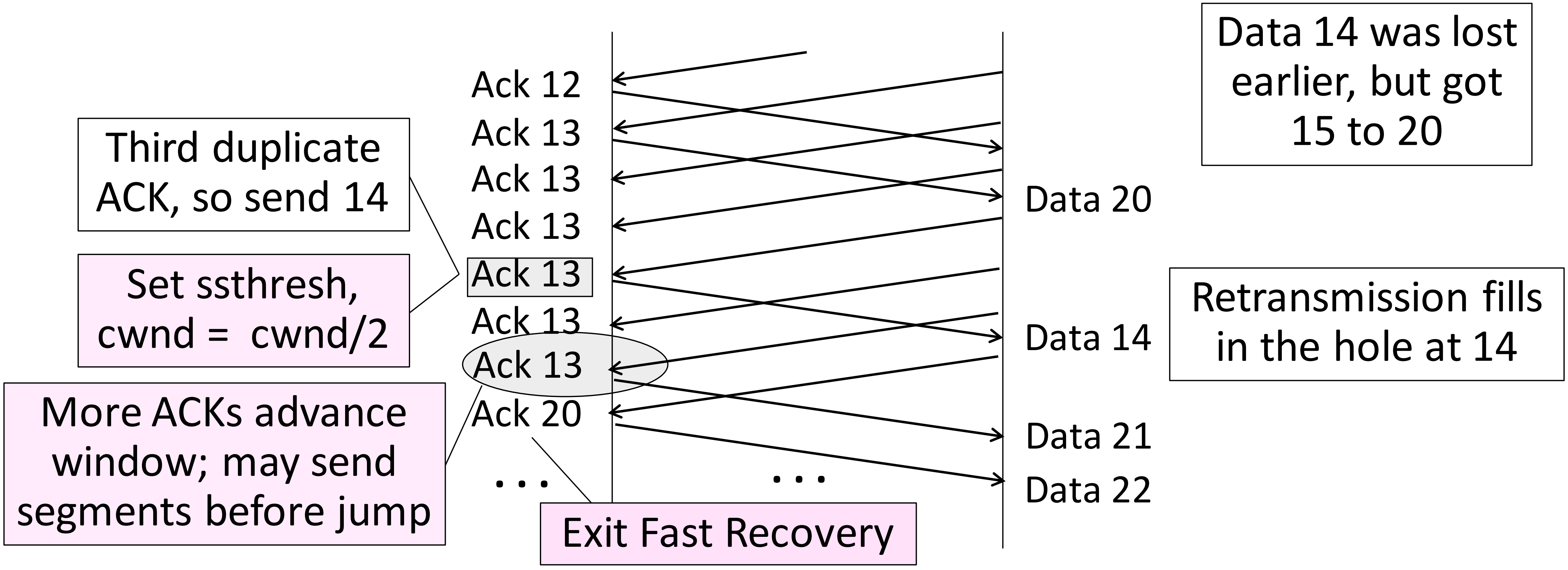
- Duplicate ACKs also give us hints about what data has arrived
  - Each new duplicate ACK means that some new segment has arrived
  - It will be the segments after the loss
  - Thus advancing the sliding window will not increase the number of segments in transit in the network

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



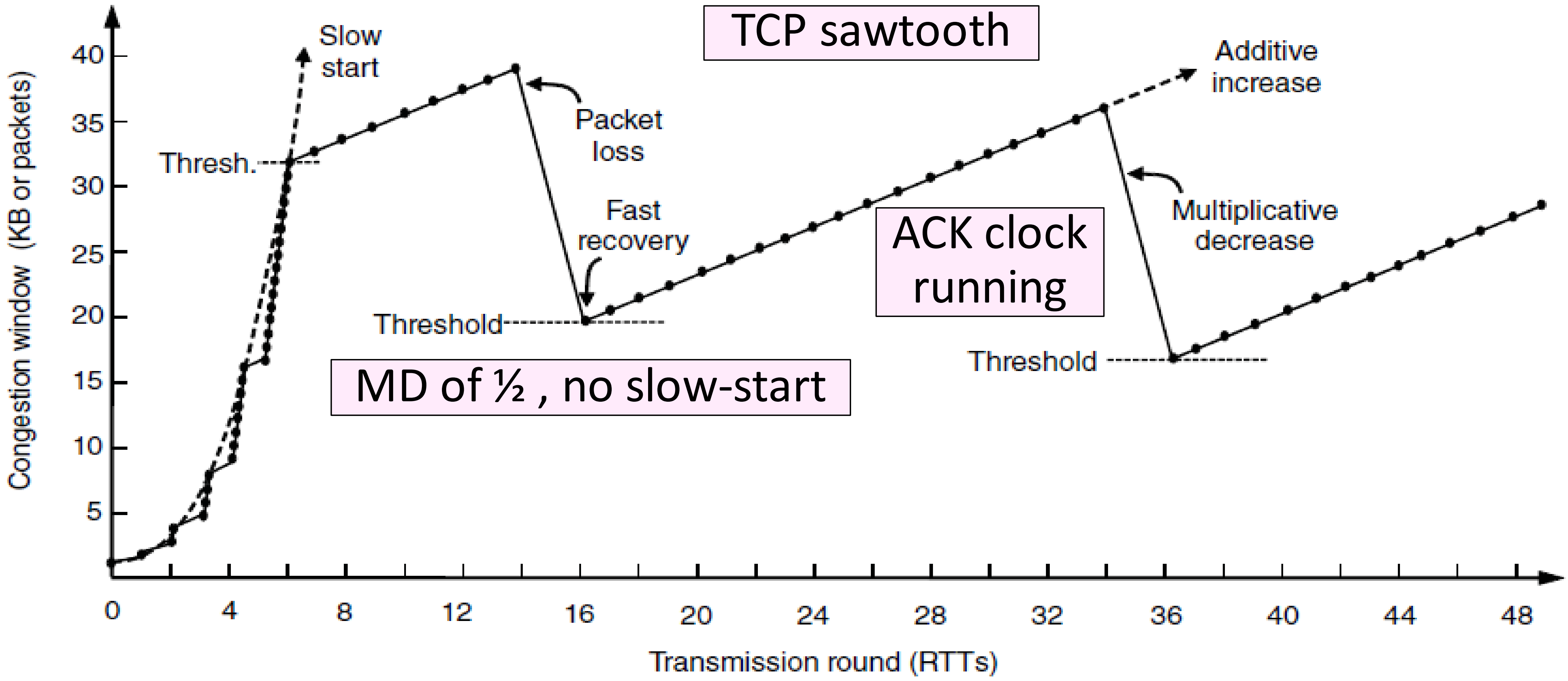
Fast Recovery



Fast Recovery

- 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
Slow start
Thresh.
Packet loss
Fast recovery
Threshold
Additive increase
Multiplicative decrease
Threshold
Congestion window (KB or packets)
Transmission round (RTTs)

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
- Selective ACK (SACK) is a better idea
  - Receiver sends ACK ranges so sender can retransmit without guesswork

Network-Side Congestion Control

Congestion Avoidance vs. Control

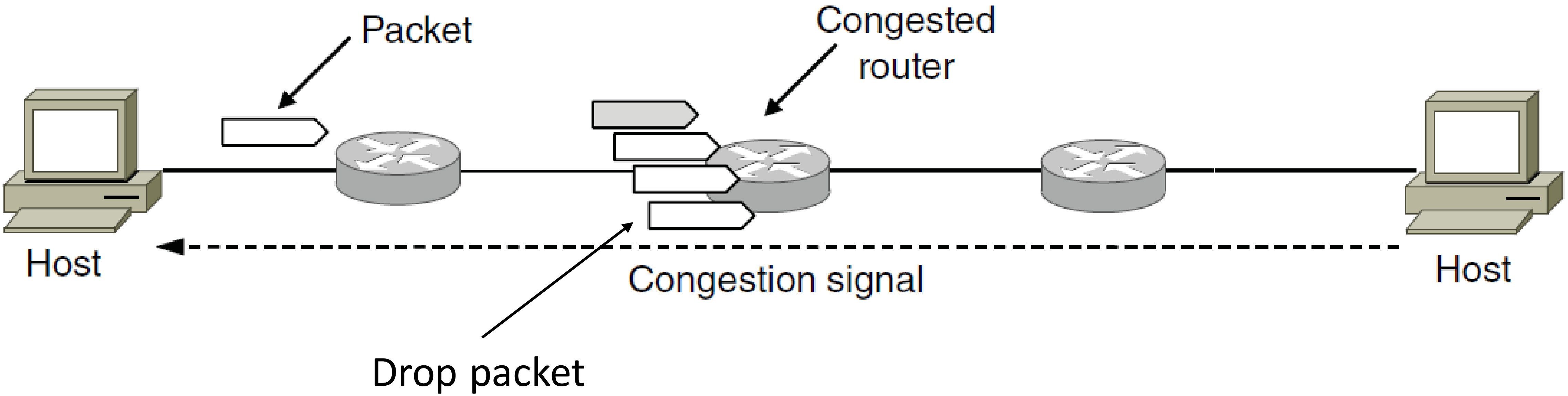
- Classic TCP drives the network into congestion and then recovers
  - Needs to see loss to slow down
- Would be better to use the network but avoid congestion altogether!
  - Reduces loss and delay
- But how can we do this?

Random Early Detection (RED)

- When router's buffer is filling, drop TCP packets at random
- TCP flow takes the dropped packet as a loss and slows down
  - Note this scheme relies only on TCP characteristics
  - Don't have to modify headers or require that all routers support it
- Drop at random, depending on queue size
  - If queue empty, accept packet always
  - If queue full, always drop
  - As queue approaches full, increase likelihood of packet drop
    - Example: 1 queue slot left, 10 packets expected, 90% chance of drop
- When you pick a packet at random to drop, which flow is it most likely to belong to?

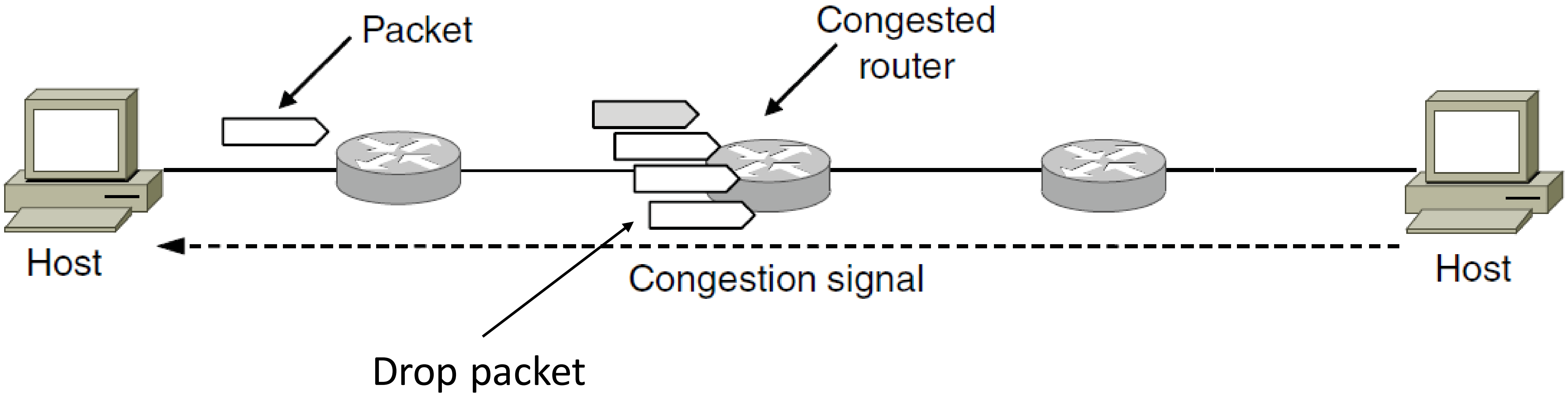
RED (Random Early Detection)

- Router detects the onset of congestion via its queue
  - Prior to congestion, drop a packet to signal



RED (Random Early Detection)

- Sender enters MD, slows packet flow
  - We shed load, everyone is happy



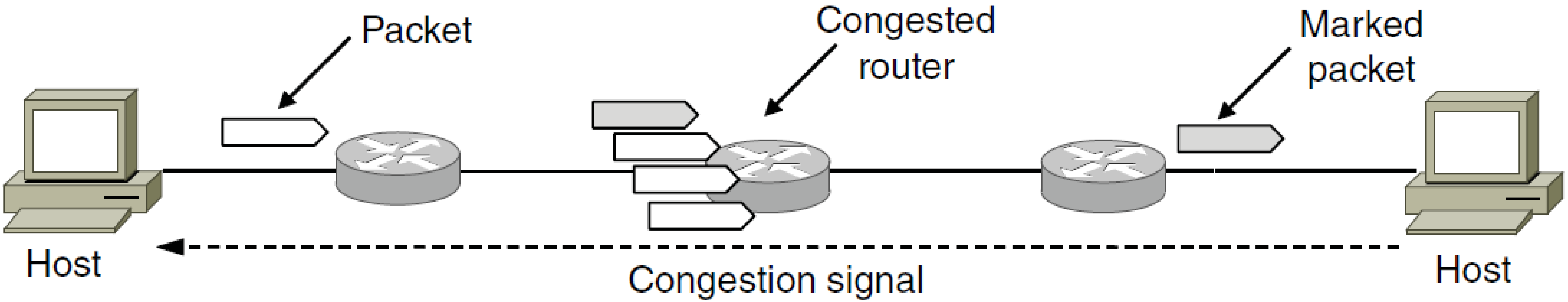
ECN (Explicit Congestion Notification)

- Idea: to send feedback to sender, RED drops a packet
  - Why not deliver the packet, but “set a bit” in it indicating that the packet has encountered a congested router?

- The problems:
  - What bit?
  - The packet is headed to the receiver, but notification needs to go to the sender

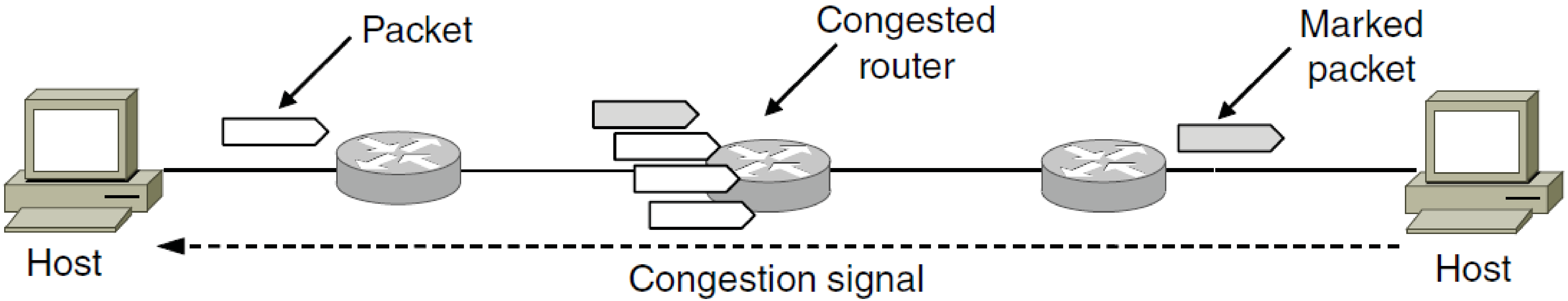
ECN (Explicit Congestion Notification)

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



ECN

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



ECN

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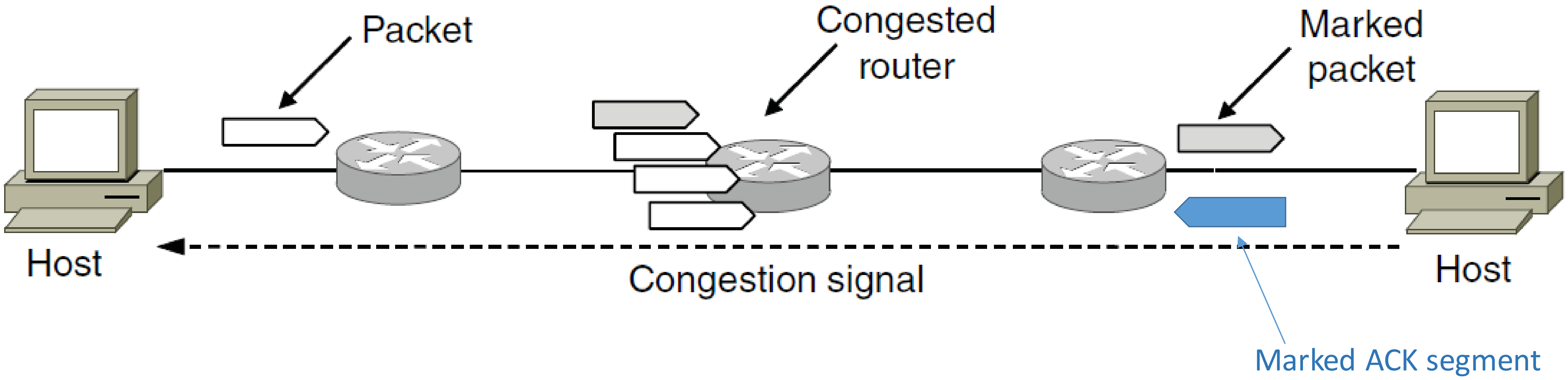

ECN

- 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