CS4450

Computer Networks:
Architecture and Protocols

Lecture 22
TCP and Congestion Control

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Recap: WHYs behind TCP design

- Started from first principles
    - Correctness condition for reliable transport
- … to understanding **why feedback from receiver is necessary** (sol-v1)
- … to understanding **why timers may be needed** (sol-v2)
- … to understanding **why window-based design may be needed** (sol-v3)
- … to understanding **why cumulative ACKs may be a good idea**
    - Very close to modern TCP

Lets learn today's reliable transport: TCP

Transport layer

- Transport layer offer a “pipe” abstraction to applications
- Data goes in one end of the pipe and emerges from other
- **Pipes are between processes, not hosts**
- There are two basic pipe abstractions

Two Pipe Abstractions

- **Unreliable packet** delivery (UDP)
  - Unreliable (application responsible for resending)
  - Messages limited to single packet

- **Reliable byte stream** delivery
  - Bytes inserted into pipe by sender
  - They emerge, in order at receiver (to the app)

- What features must transport protocol implement to support these abstractions?

Transmission Control Protocol (TCP)

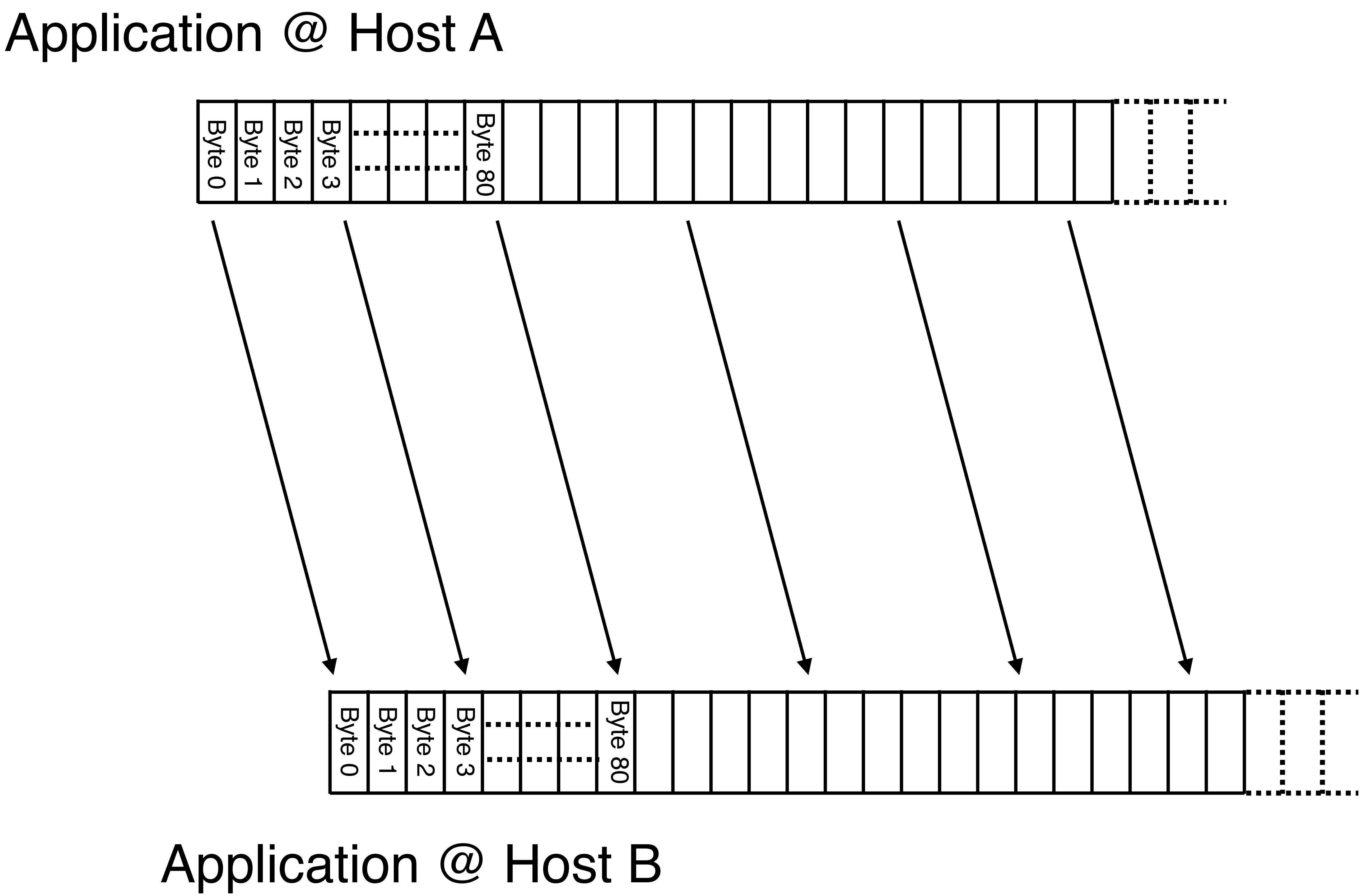
- Reliable, in-order delivery
  - Ensures byte stream (eventually) arrives intact
  - In the presence of corruption, delays, reordering, loss
- Connection oriented
  - Explicit set-up and tear-down of TCP session
- Full duplex stream of **byte service**
  - **Sends and receives stream of bytes**
- **Flow control**
  - Ensures the sender does not overwhelm the receiver
- **Congestion control**
  - Dynamic adaptation to network path's capacity

Basic Components of TCP

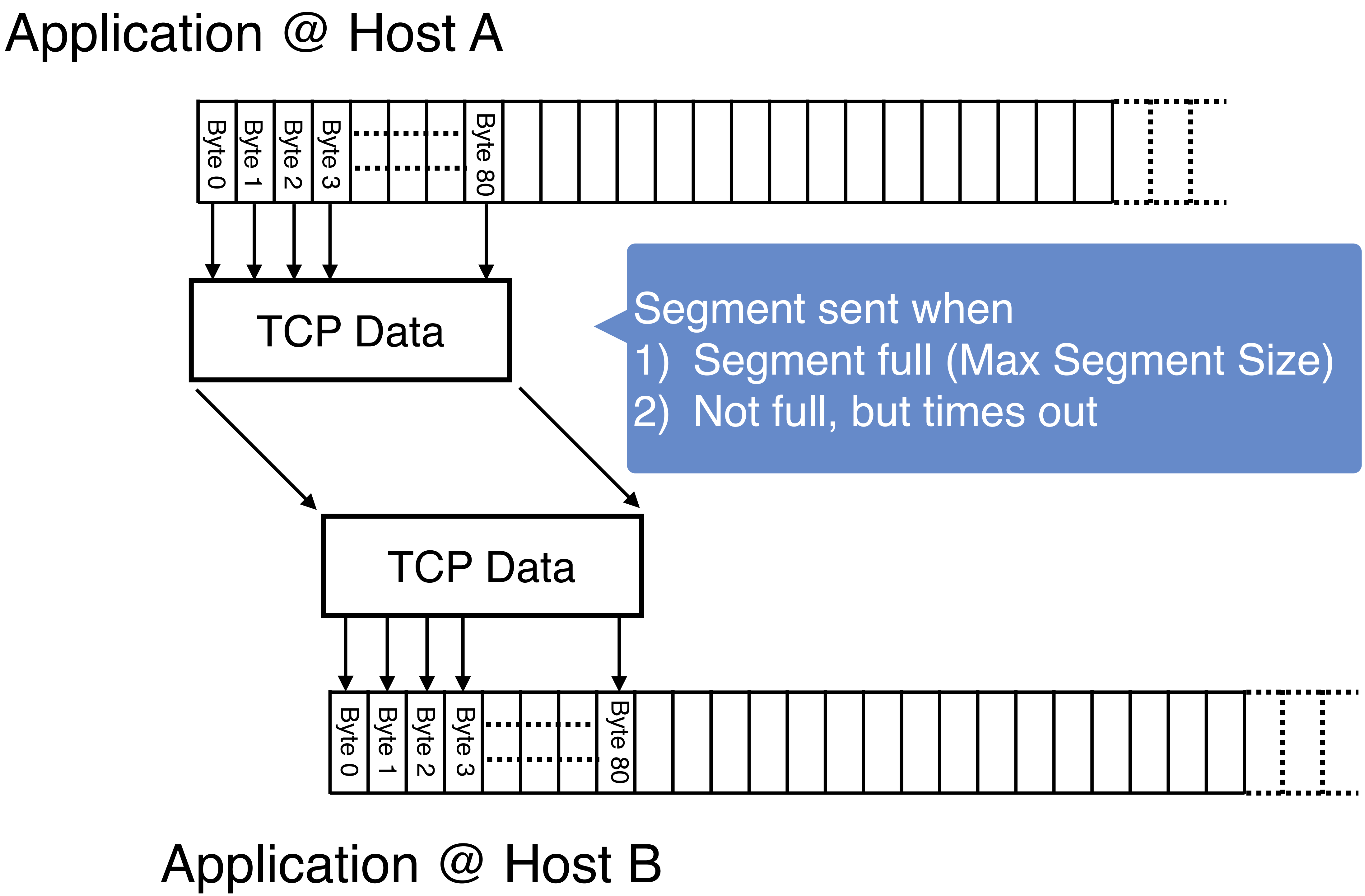
- **Segments, Sequence numbers, ACKs**
    - TCP uses byte sequence numbers to identify payloads
    - ACKs referred to sequence numbers
    - Window sizes expressed in terms of # of bytes
- **Retransmissions**
    - Can’t be correct without retransmitting lost/corrupted data
    - TCP retransmits based on timeouts and duplicate ACKs
        - Timeouts based on estimate of RTT
- **Flow Control**
- **Congestion Control**

Segments, Sequence Numbers and ACKs

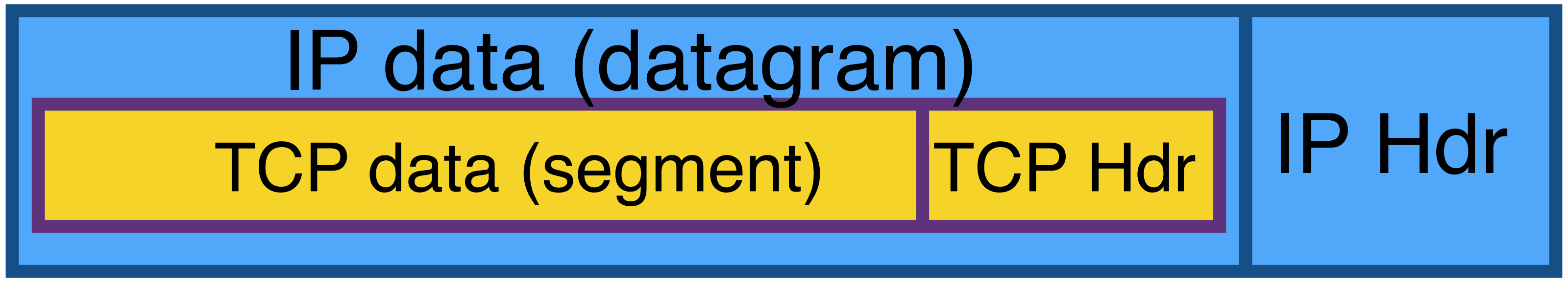
TCP "Stream of Bytes" Service



TCP "Stream of Bytes" Service



TCP Segment



- IP Packet
  - No bigger than Maximum Transmission Unit (MTU)
  - E.g., up to 1500 bytes with Ethernet
- TCP Packet
  - IP packet with a TCP header and data inside
  - TCP header >= 20 bytes long
- TCP Segment
  - No more than MSS (Maximum Segment Size) bytes
  - E.g., up to 1460 consecutive bytes from the stream
  - MSS = MTU - IP header - TCP header

ACKing and Sequence Numbers

- Sender sends segments (byte stream)
  - Data starts with Initial Sequence Number (ISN): X
    - See backup slides on how this is set
  - Packet contains B bytes
    - X, X+1, X+2, …, X+B-1

- Upon receipt of a segment, receiver sends an ACK
  - If all data prior to X already received:
    - ACK acknowledges X+B (because that is next expected byte)
  - If highest **contiguous** byte received is smaller value Y
    - ACK acknowledges Y+1
    - Even if this has been ACKed before

Any Questions?

TCP Retransmission

Two Mechanisms for Retransmissions

- Duplicate ACKs

- Timeouts

Loss with Cumulative ACKs

- Sender sends packets with 100B and seqnos
    - 100, 200, 300, 400, 500, 600, 700, 800, 900
- Assume 5th packet (seqno 500) is lost, but no others
- Stream of ACKs will be
    - 200, 300, 400, 500, 500, 500, 500, 500

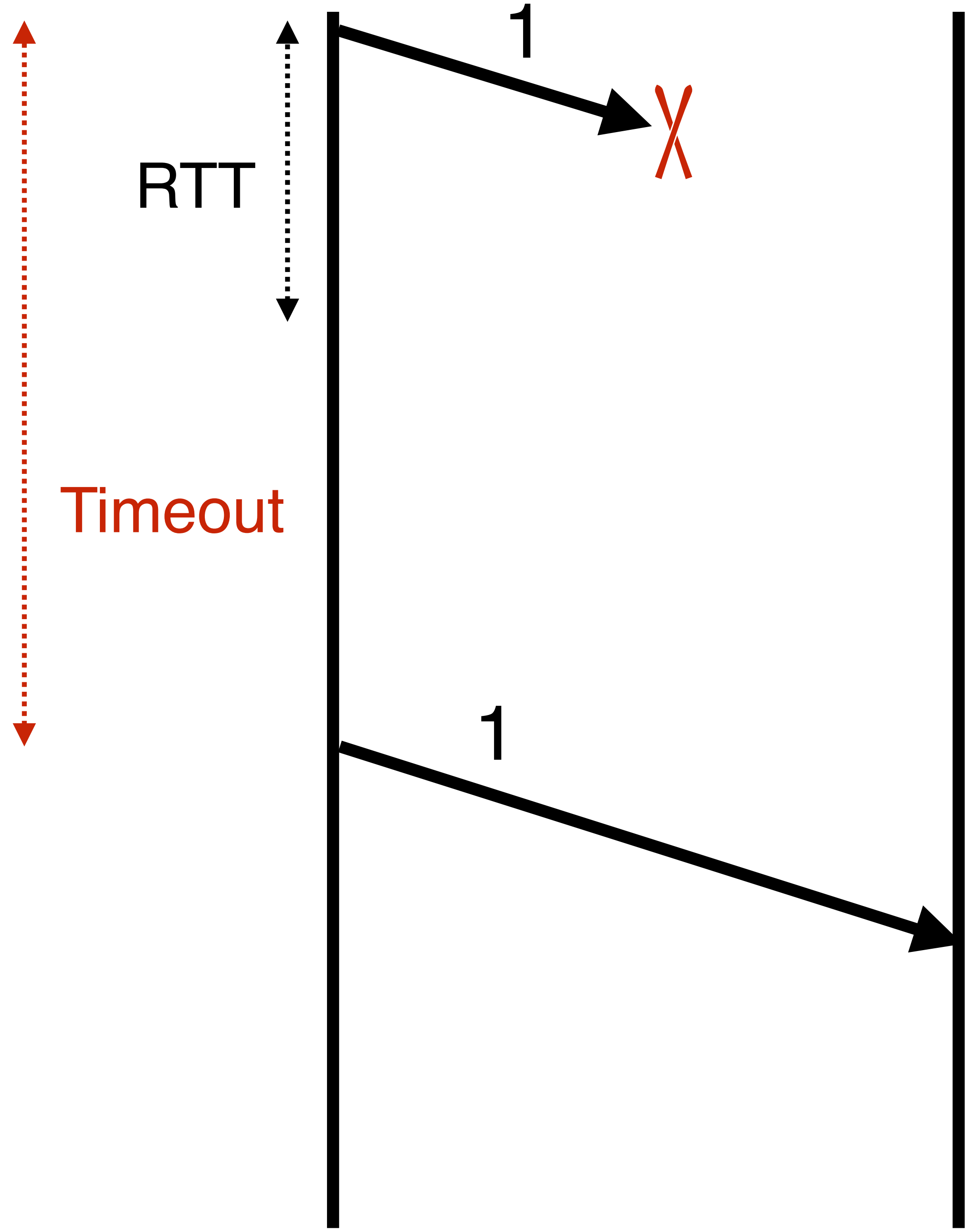
Loss with Cumulative ACKs

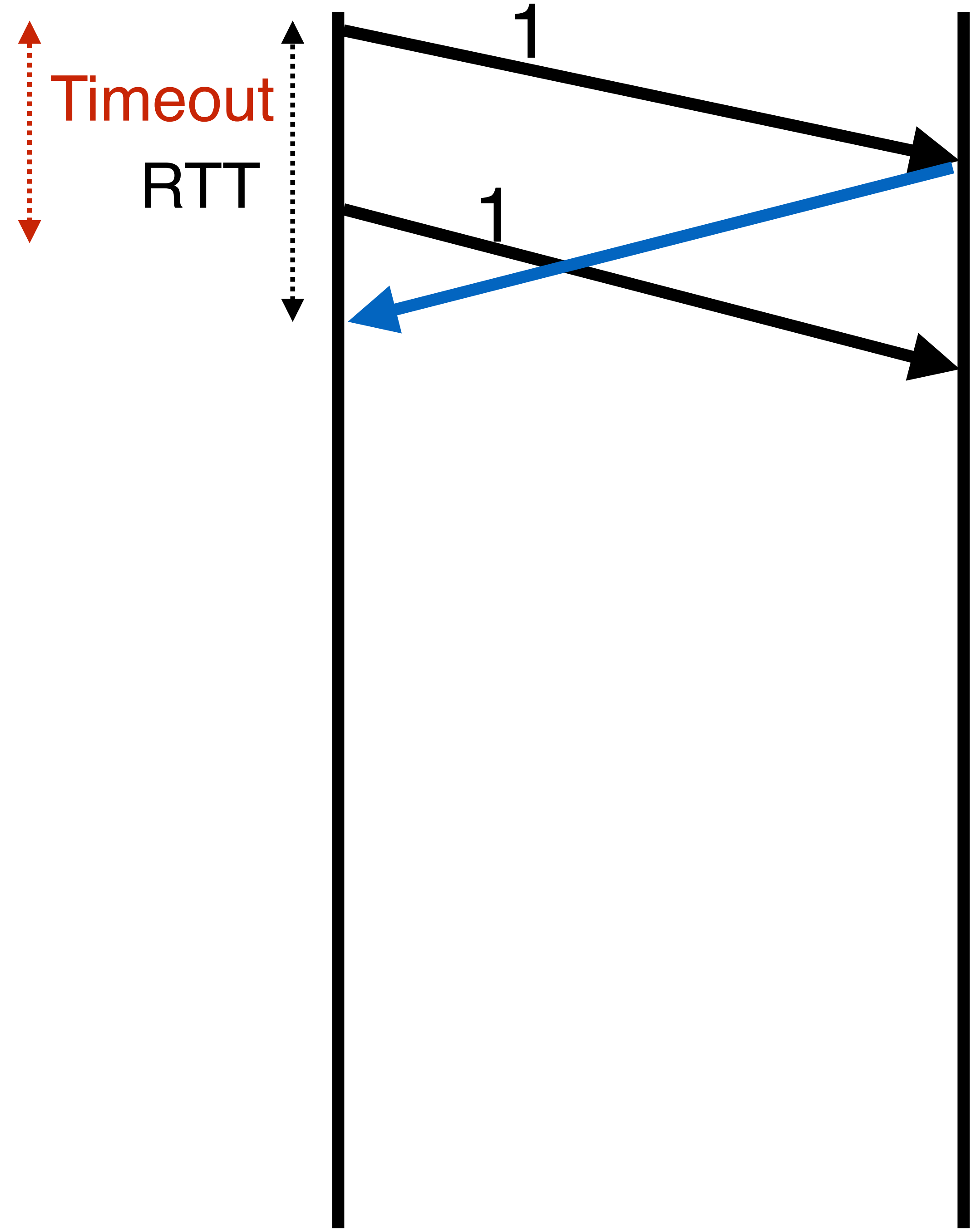
- Duplicate ACKs are a sign of an isolated loss
  - The lack of ACK progress means 500 hasn’t been delivered
  - Stream of ACKs means some packets are being delivered
- Therefore, could trigger resend upon receiving k duplicate ACKs
  - TCP uses k = 3
- We will revisit this in congestion control

Timeouts and Retransmissions

- TCP only has a single timer
- TCP resets timer whenever new data is ACKed
- RTO (Retransmit Time Out) is the basic timeout value

Setting the Timeout Value (RTO)





Timeout too long -> inefficient

Timeout too short -> duplicate packets

TCP Flow Control

Flow Control (Sliding Window)

- Advertised Window: W
  - Can send W bytes beyond the next expected byte
- Receiver uses W to prevent sender from overflowing buffer
- Limits number of bytes sender can have in flight

Advertised Window Limits Rate

- Sender can send no faster than W/RTT bytes/sec
- In original TCP, that was the sole protocol mechanism controlling sender’s rate
- What’s missing?
- **Congestion control** about how to adjust W to avoid network congestion

Any Questions?

TCP Congestion Control

TCP congestion control: high-level idea

- End hosts adjust sending rate
- Based on implicit feedback from the network
  - Implicit: router drops packets because its buffer overflows, not because it's trying to send message
- Hosts probe network to test level of congestion
  - Speed up when no congestion (i.e., no packet drops)
  - Slow down when when congestion (i.e., packet drops)
- How to do this efficiently?
  - Extend TCP's existing window-based protocol…
  - Adapt the window size based in response to congestion

All These Windows…

- **Flow control window**: Advertised Window (RWND)
  - How many bytes can be sent without overflowing receivers buffers
  - Determined by the receiver and reported to the sender
- **Congestion Window** (CWND)
  - How many bytes can be sent without overflowing routers
  - Computed by the sender using congestion control algorithm
- **Sender-side window** = minimum{CWND,RWND}
  - Assume for this lecture that RWND >> CWND

Note

- This lecture will talk about CWND in units of MSS
    - Recall MSS: Maximum Segment Size, the amount of payload data in a TCP packet
    - This is only for pedagogical purposes
- Keep in mind that real implementations maintain CWND in bytes

Basics of TCP Congestion

- Congestion Window (CWND)
  - Maximum # of unacknowledged bytes to have in flight
  - Rate ~CWND/RTT
- Adapting the congestion window
  - Increase upon lack of congestion: optimistic exploration
  - Decrease upon detecting congestion
- But how do you detect congestion?

Not All Losses the Same

- **Duplicate ACKs: isolated loss**
  - Still getting ACKs
- **Timeout: possible disaster**
  - Not enough duplicate ACKs
  - Must have suffered several losses

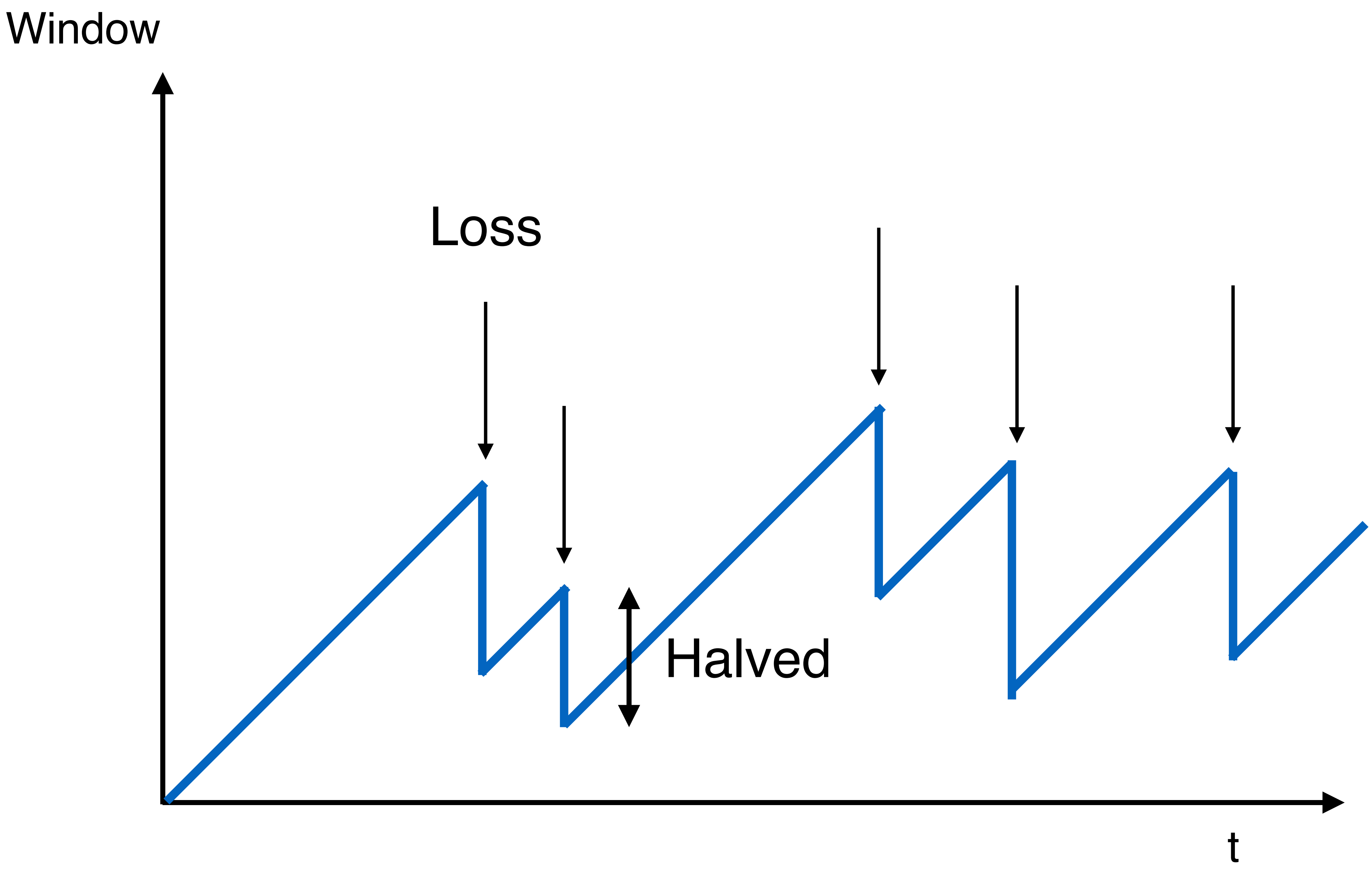
Additive Increase, Multiplicative Decrease (AIMD)

- Additive increase
  - On success of last window of data, increase by one MSS
  - If W packets in a row have been ACKed, increase W by one
  - i.e., +1/W per ACK
- Multiplicative decrease
  - On loss of packets by DupACKs, divide congestion window by half
  - Special case: when timeout, reduce congestion window to one MSS

AIMD

- ACK: CWND -> CWND + 1/CWND
  - When CWND is measured in MSS
  - Note: after a full window, CWND increase by 1 MSS
  - Thus, **CWND increases by 1 MSS per RTT**
- 3rd DupACK: CWND -> CWND/2
- Special case of timeout: CWND -> 1 MSS

Leads to the TCP Sawtooth



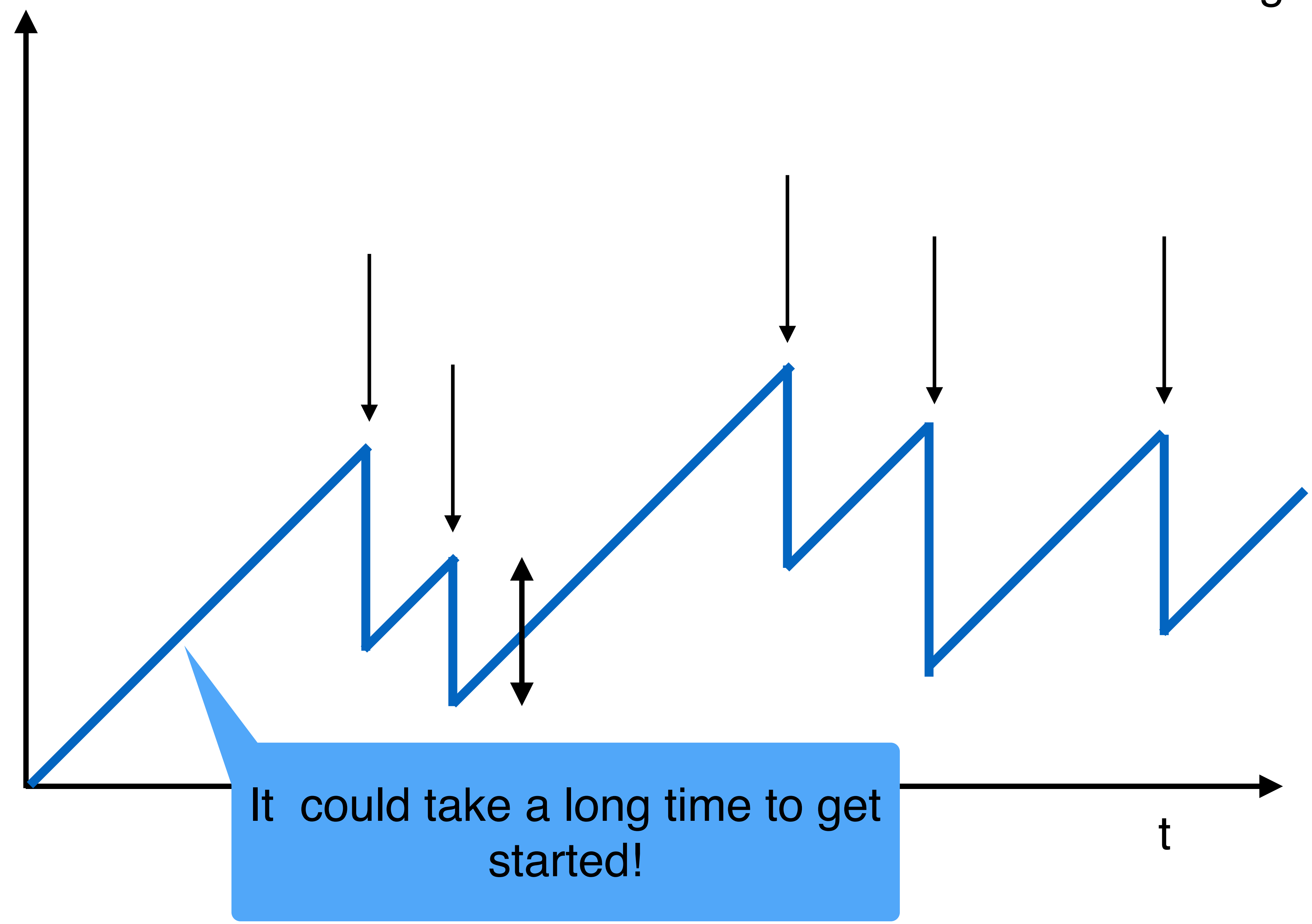
Any Questions?

Slow Start

AIMD Starts Too Slowly

Window

Need to start with a small CWND to avoid overloading the network



Bandwidth Discovery with Slow Start

- Goal: estimate available bandwidth
  - Start slow (for safety)
  - But ramp up quickly (for efficiency)

- Consider
  - RTT = 100ms, MSS=1000bytes
  - Window size to fill 1Mbps of BW = 12.5 MSS
  - Window size to fill 1 Gbps = 12,500 MSS
    - With just AIMD, it takes about 12500 RTTs to get to this window size!
    - ~21 mins

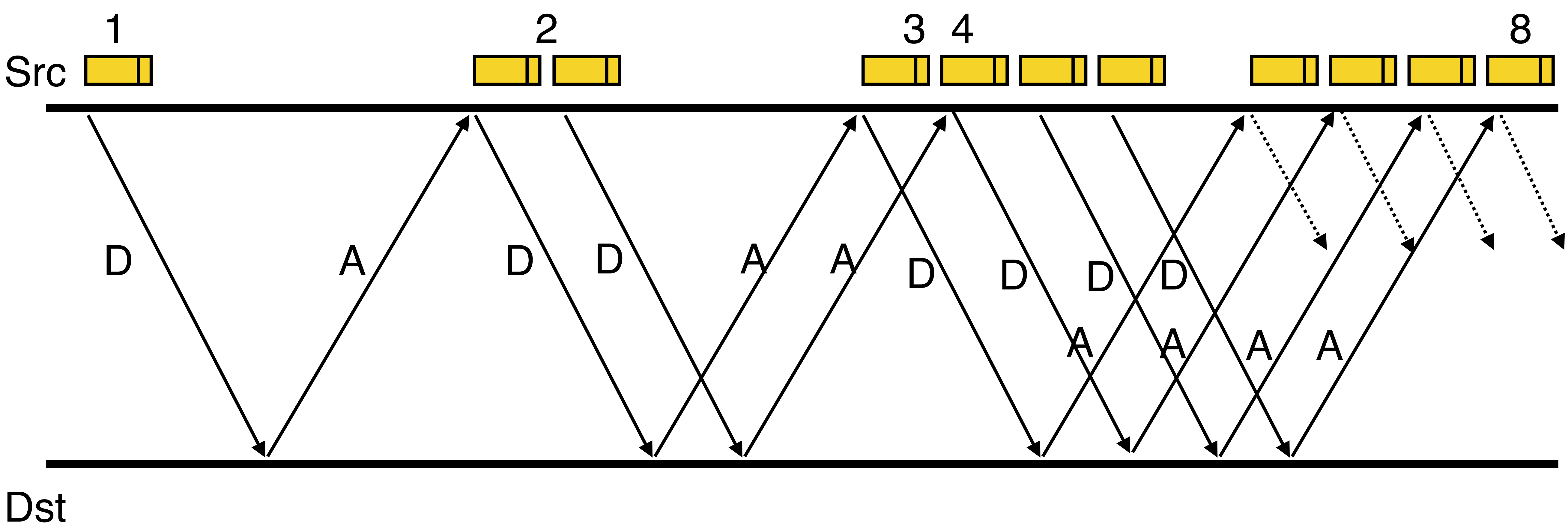
“Slow Start” Phase

- Start with a small congestion window
  - Initially, CWND is 1 MSS
  - So, initial sending rate is MSS/RTT
- That could be pretty wasteful
  - Might be much less than the actual bandwidth
  - Linear increase takes a long time to accelerate
- Slow-start phase (**actually “fast start”**)
  - Sender starts at a slow rate (hence the name)
  - … but increases exponentially until first loss

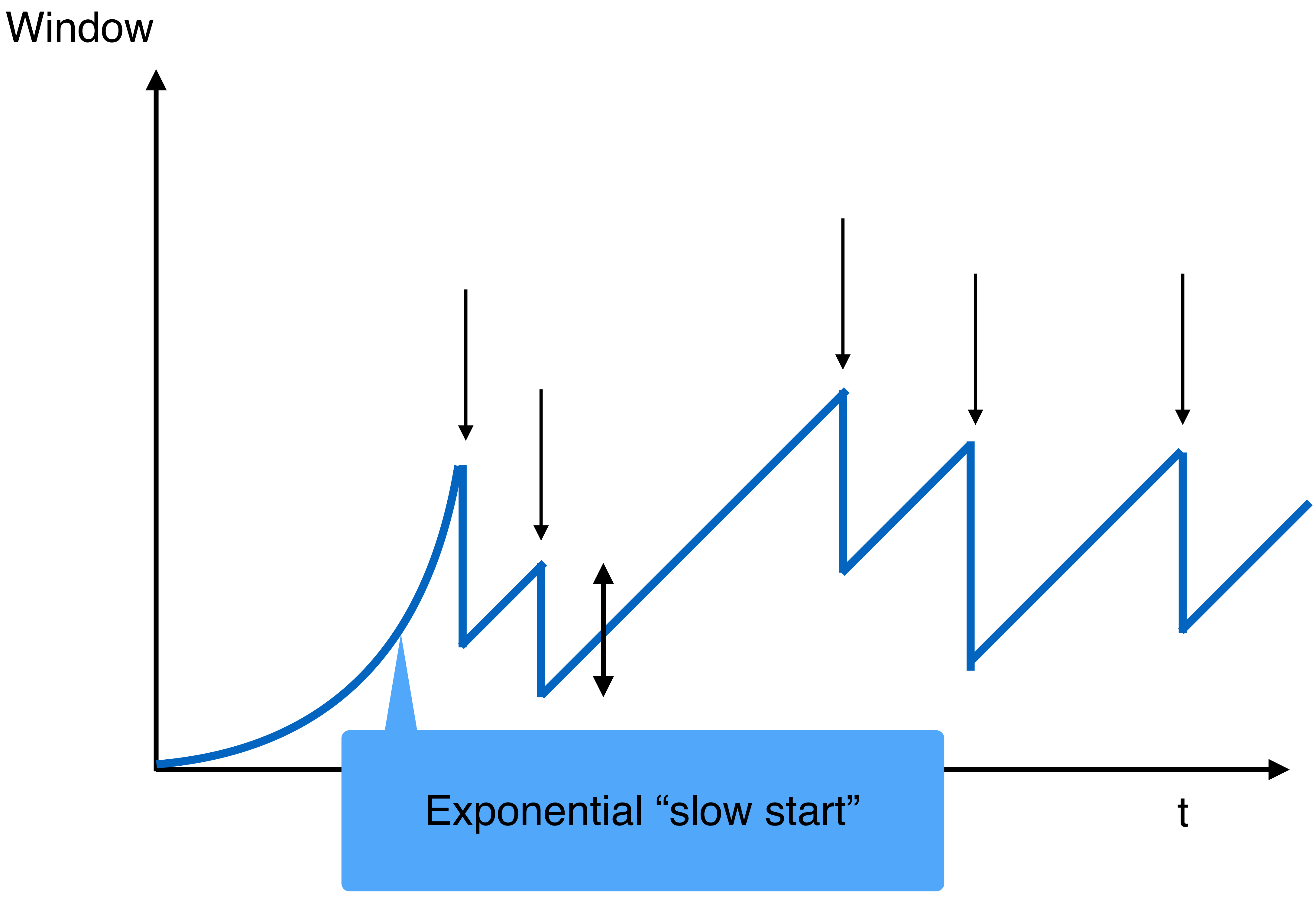
Slow Start in Action

Double CWND per round-trip time

Simple implementation: on each ACK, CWND += MSS



Slow Start and the TCP Sawtooth



Why is it called slow-start? Because TCP originally had no congestion control mechanism. The source would just start by sending a whole window’s worth of data.

Slow-Start vs AIMD

- When does a sender stop Slow-Start and start Additive Increase?
- Introduce a "slow start threshold" (ssthresh)
  - Initialized to a large value
  - On timeout, ssthresh = CWND/2
- When CWND > ssthresh, sender switches from slow-start to AIMD-style increase

Timeouts

Loss Detected by Timeout

- Sender starts a timer that runs for RTO seconds
- **Restart timer whenever ACK for new data arrives**
- If timer expires
  - Set SSHTHRESH <- CWND/2 ("Slow Start Threshold")
  - Set CWND <- 1 (MSS)
  - Retransmit **first** lost packet
  - Execute Slow Start until CWND > SSTHRESH
  - After which switch to Additive Increase

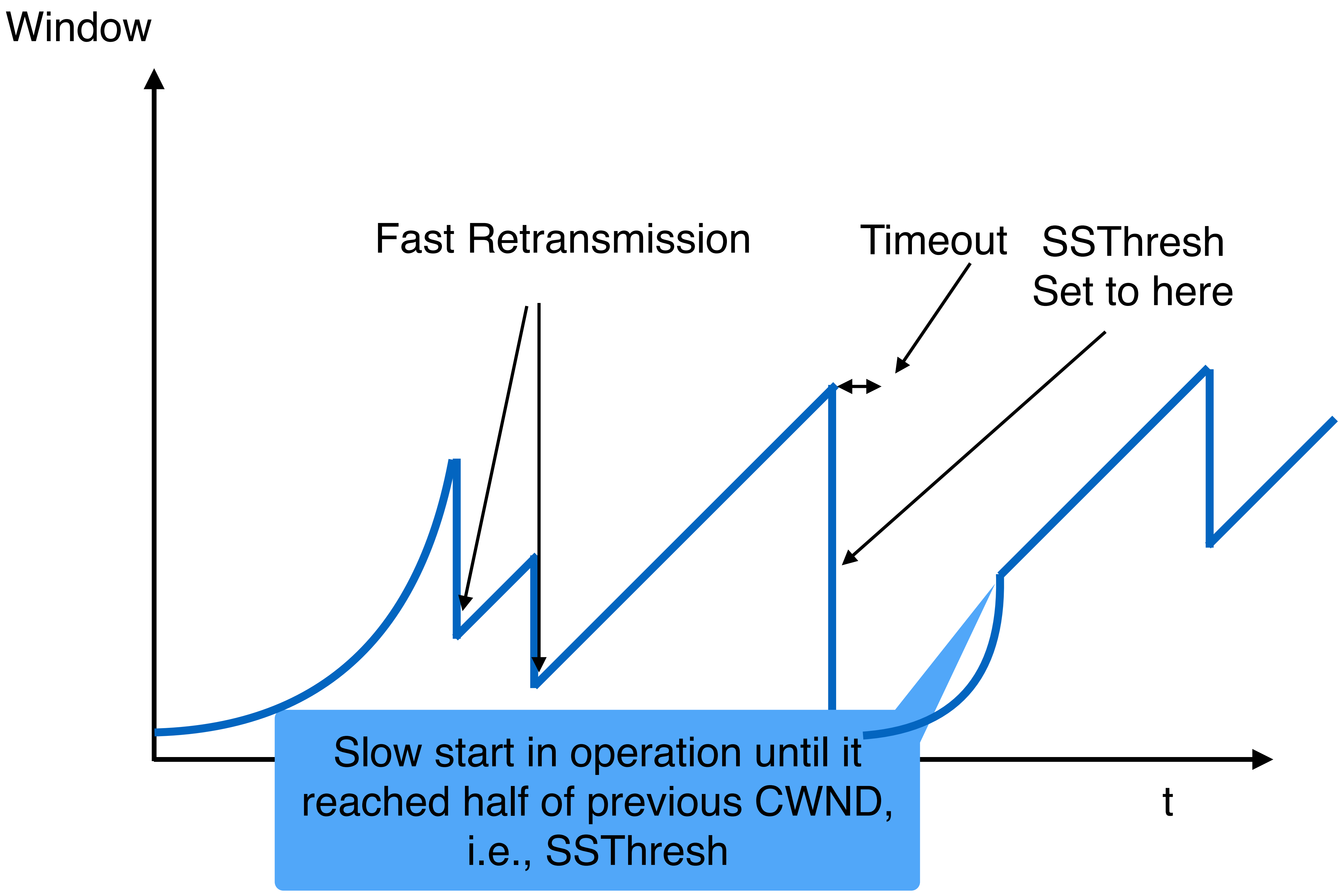
Summary of Increase

- "Slow start": increase CWND by 1 (MSS) for each ACK
  - A factor of 2 per RTT
- Leave slow-start regime when either:
  - CWND > SSTHRESH
  - Packet drop detected by dupacks
- Enter AIMD regime
  - Increase by 1 (MSS) for each window's worth of ACKed data

Summary of Decrease

- Cut CWND half on loss detected by dupacks
  - **Fast retransmit to avoid overreacting**
- Cut CWND all the way to 1 (MSS) on **timeout**
  - Set ssthresh to CWND/2
- Never drop CWND below 1 (MSS)
  - Our correctness condition: always try to make progress

Time Diagram



Slow-start restart: Go back to CWND of 1 MSS, but take advantage of knowing the previous value of CWND.

Done!

Next lecture: TCP properties and critical analysis

TCP Congestion Control Details

TCP Back up slides

Implementation

- State at sender
  - CWND (initialized to a small constant)
  - ssthresh (initialized to a large constant)
  - dupACKcount
  - Timer, as before
- Events at sender
  - ACK (new data)
  - dupACK (duplicate ACK for old data)
  - Timeout
- What about receiver? Just send ACKs upon arrival

Event: ACK (new data)

- If in slow start
    - CWND += 1

CWND packets per RTT
Hence after one RTT with no drops:
CWND = 2 x CWND

Event: ACK (new data)

- If CWND <= ssthresh
  - CWND += 1

Slow Start Phase

- Else
  - CWND = CWND + 1/CWND

Congestion Avoidance Phase

(additive increase)

CWND packets per RTT
Hence after one RTT with no drops:
CWND = CWND + 1

Event: Timeout

- On Timeout
  - ssthresh <- CWND/2
  - CWND <- 1

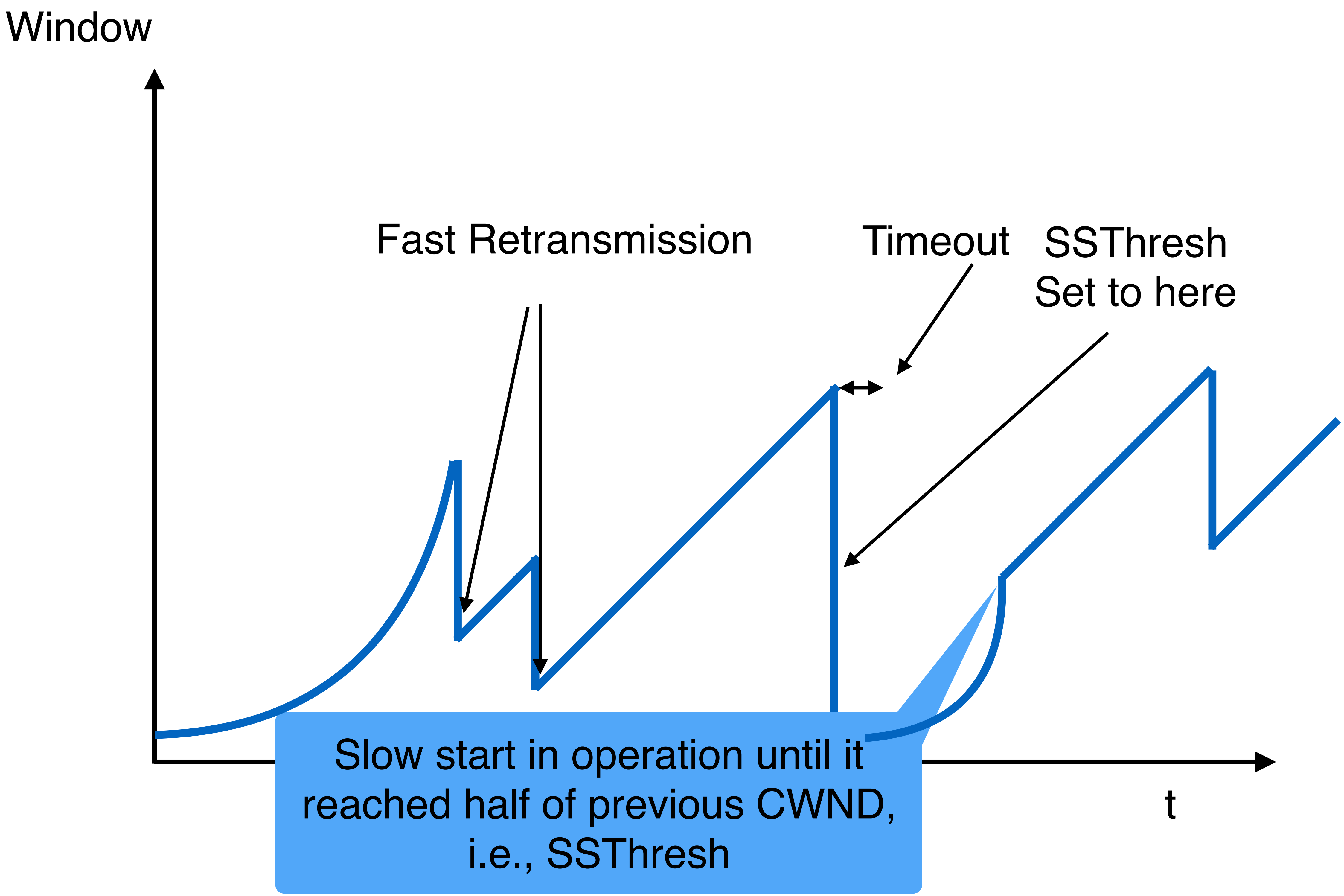
Event: dupACK

- dupACKcount++

- If dupACKcount = 3 /* fast retransmit */
  - ssthresh <- CWND/2
  - CWND <- CWND/2

Remains in congestion avoidance after fast retransmission

Time Diagram



Slow-start restart: Go back to CWND of 1 MSS, but take advantage of knowing the previous value of CWND.

TCP Flavors

- TCP Tahoe
    - CWND = 1 on triple dupACK
- TCP Reno
    - CWND = 1 on timeout
    - CWND = CWND/2 on triple dupACK

Our default assumption

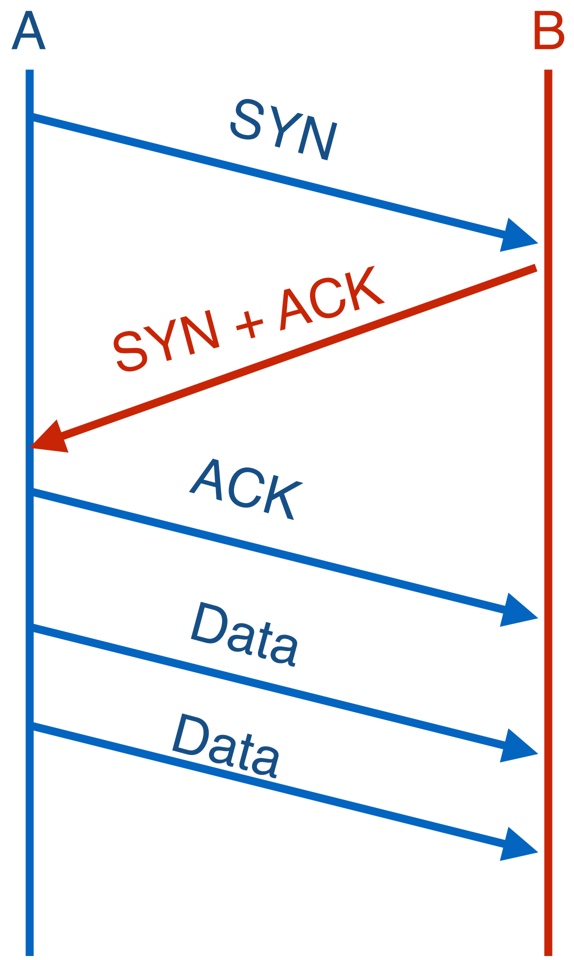
- TCP-newReno
    - TCP-Reno + improved fast recovery
- TCP-SACK
    - Incorporates selective acknowledgements

TCP Connection Establishment and Initial Sequence Numbers

Initial Sequence Number (ISN)

- Sequence number for the very first byte
    - E.g., Why not just use ISN = 0?
- Practical issue
    - IP addresses and port #s uniquely identify a connection
    - Eventually, though, these port #s do get **used again**
    - … small chance an old packet is **still in flight**
- TCP therefore requires changing ISN
    - Set from 32-bit clock that ticks every 4 microseconds
    - … only wraps around once every 4.55 hours
- To establish a connection, hosts exchange ISNs
    - How does this help?

Establishing a TCP Connection

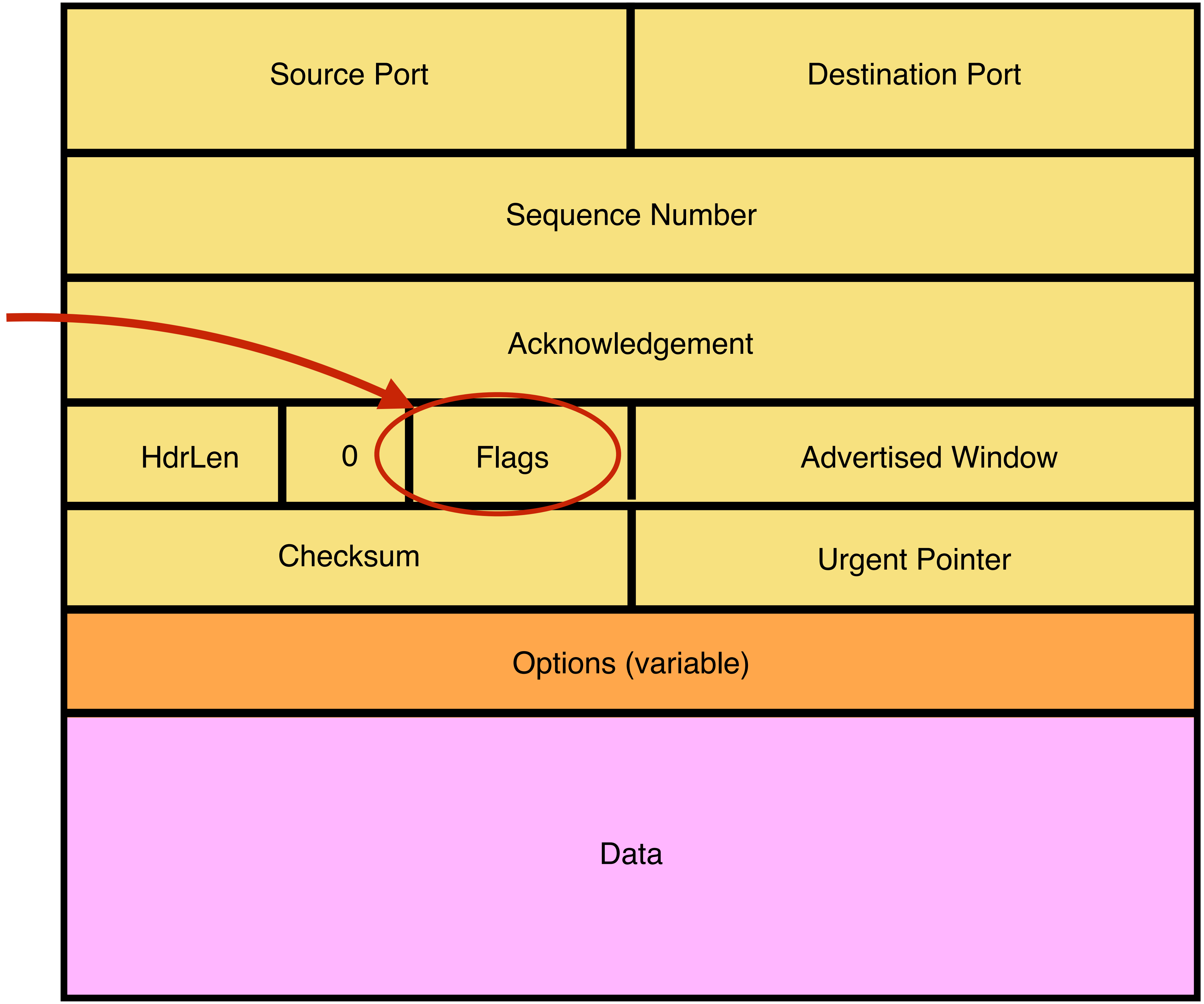


Each host tells its ISN to the other host.

- Three-way handshake to establish connection
  - Host A sends a **SYN** (open; “synchronize sequence numbers”) to host B
  - Host B returns a SYN acknowledgement (**SYN ACK**)
  - Host sends an **ACK** to acknowledge the SYN ACK

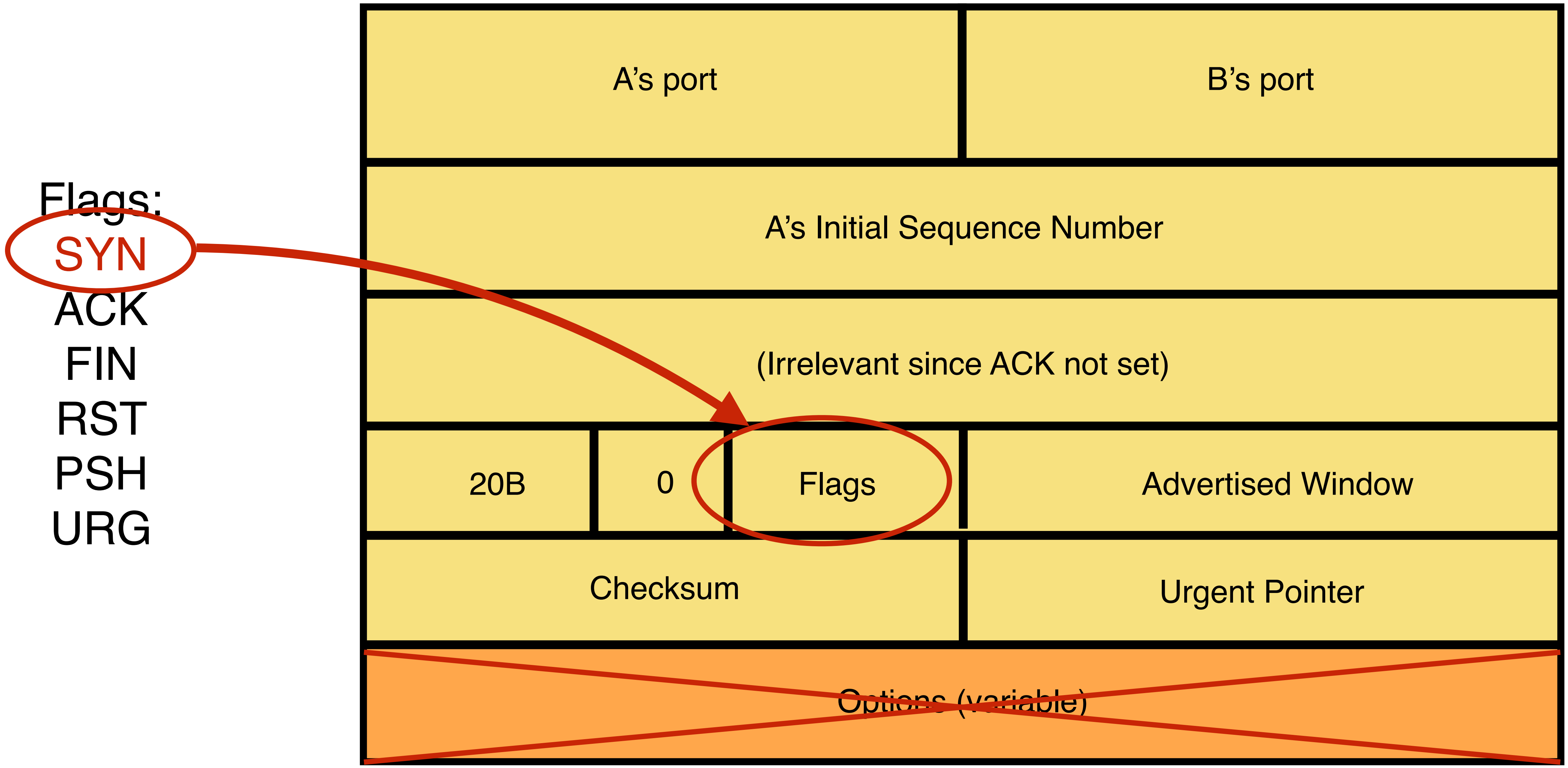
TCP Header

Flags:
SYN
ACK
FIN
RST
PSH
URG



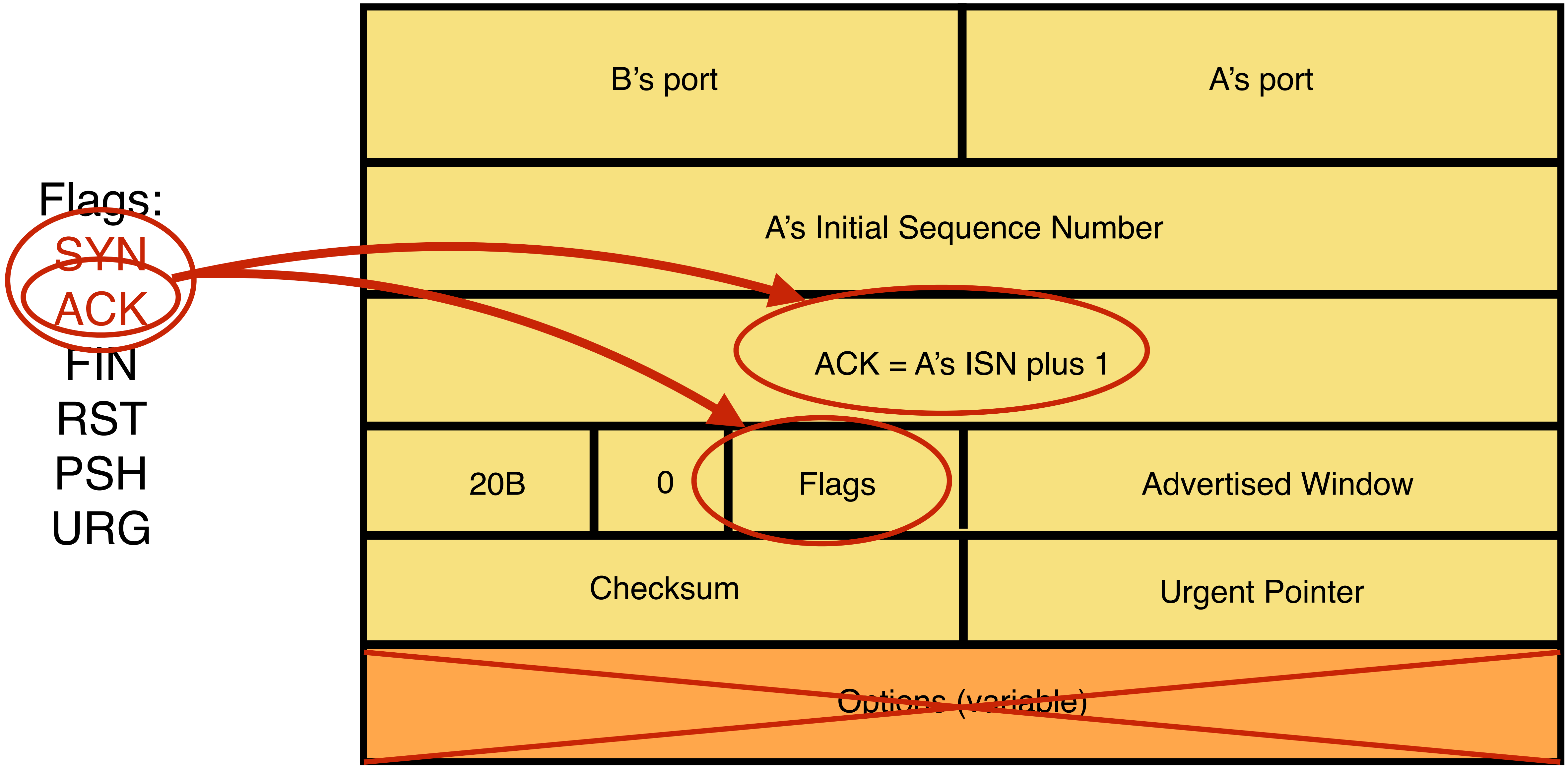
See /usr/include/netinet/tcp.h on Unix Systems

Step 1: A's Initial SYN Packet



A tells B it wants to open a connection…

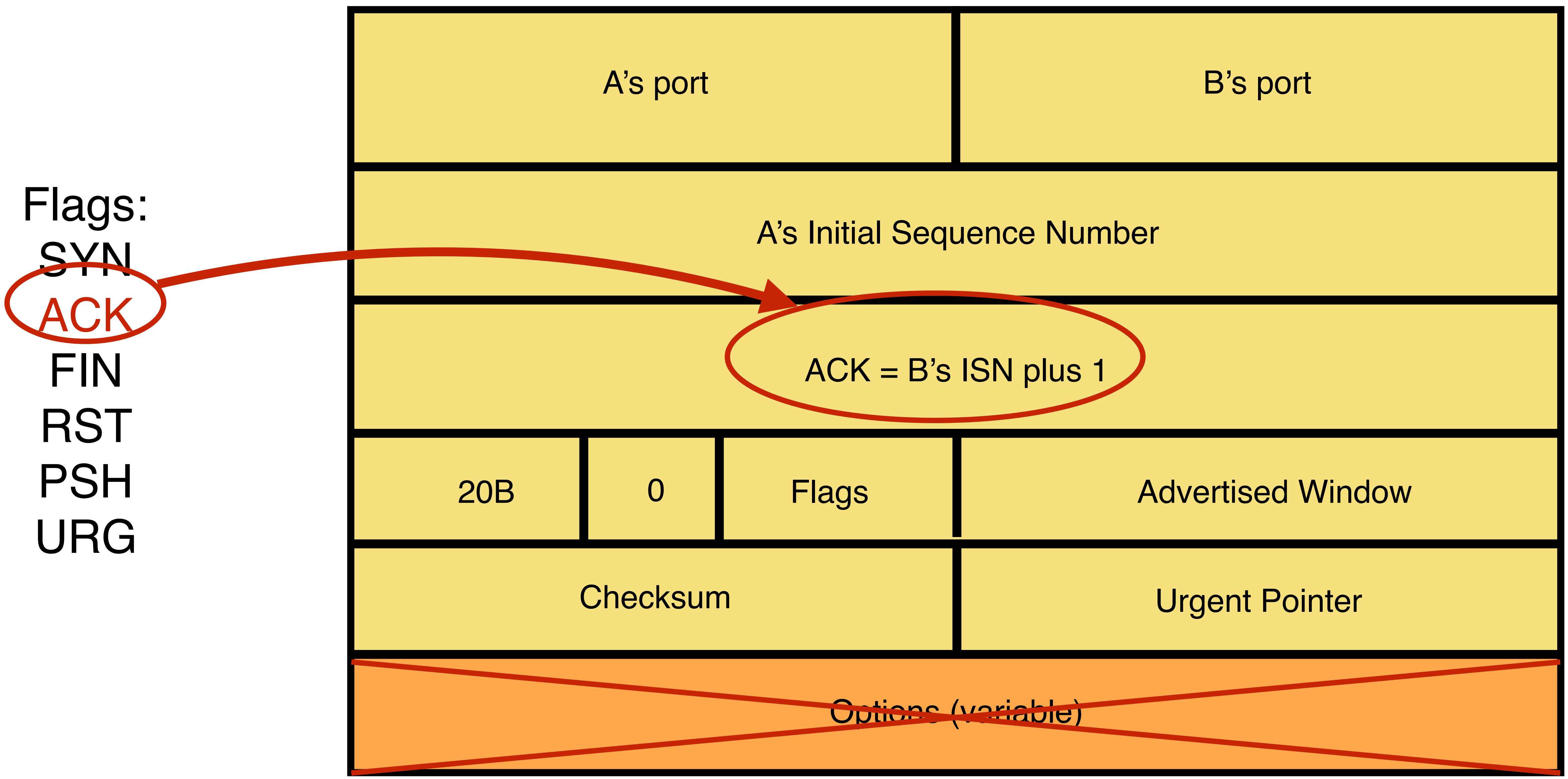
Step 2: B's SYN-ACK Packet



B tells A it accepts and is ready to hear the next byte…

… upon receiving this packet, A can start sending data

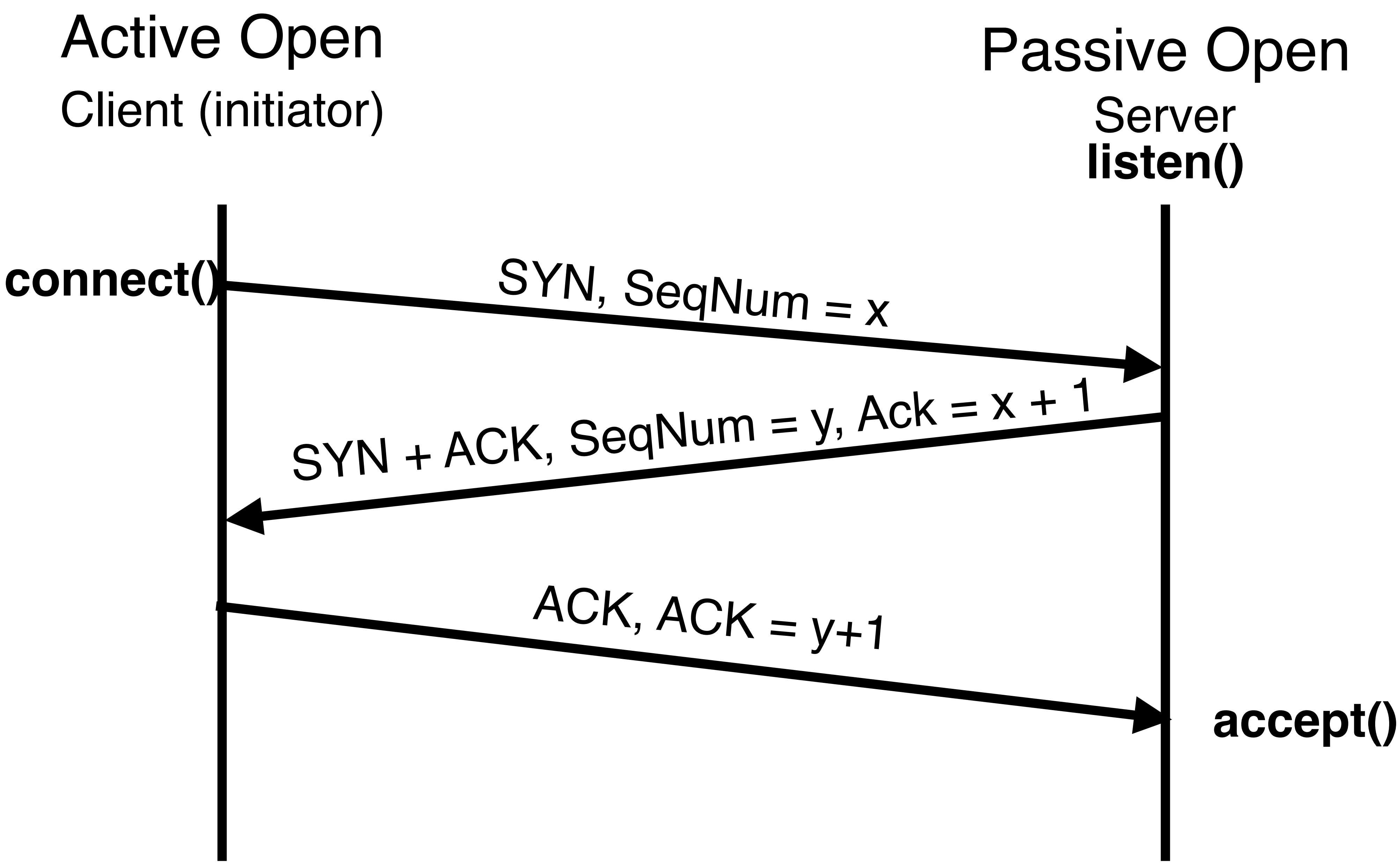
Step 3: A's ACK of the SYN-ACK



A tells B it's likewise okay to start sending

… upon receiving this packet, B can start sending data

Timing Diagram: 3-Way Handshaking



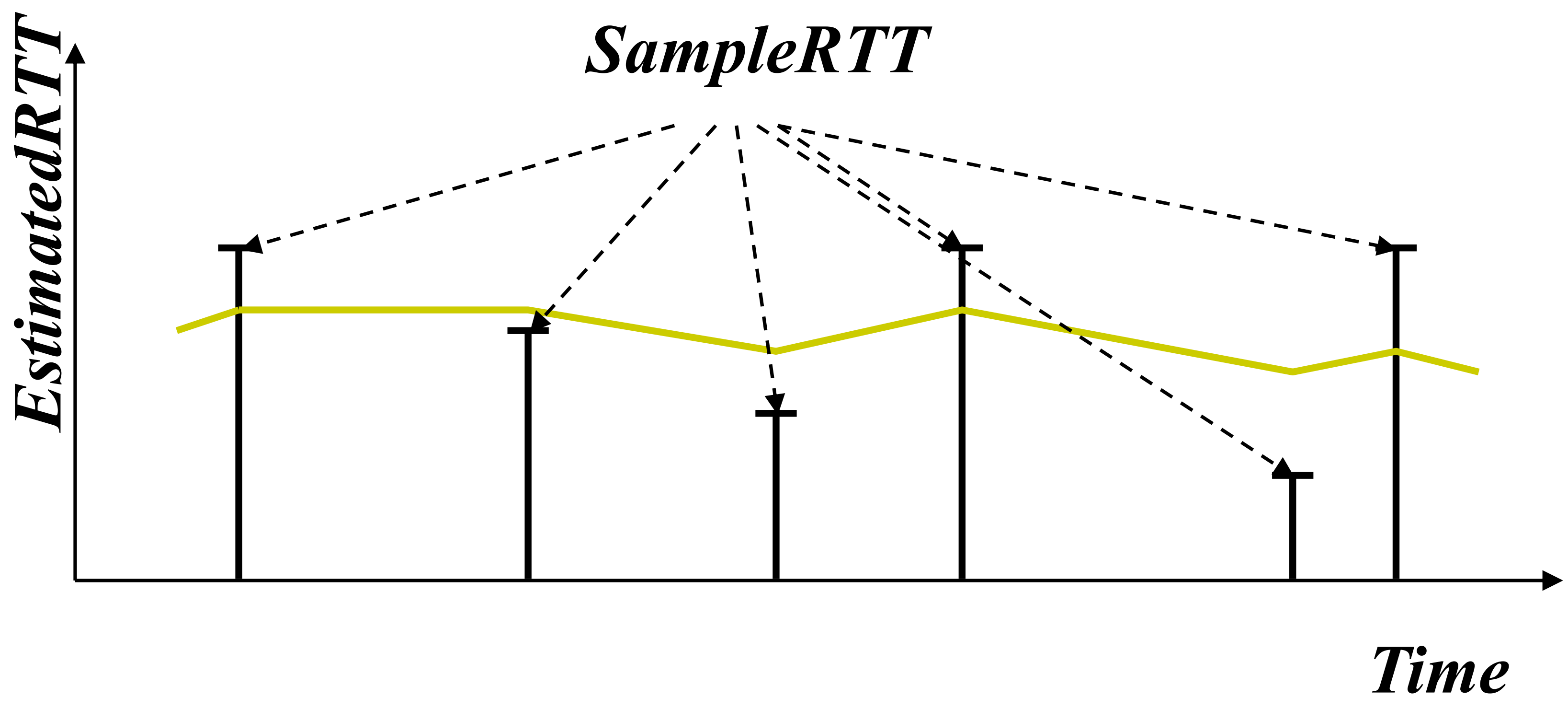
Could Base RTO on RTT Estimation

- Use exponential averaging if RTT samples

SampleRTT = AckRcvdTime - SendPktTime

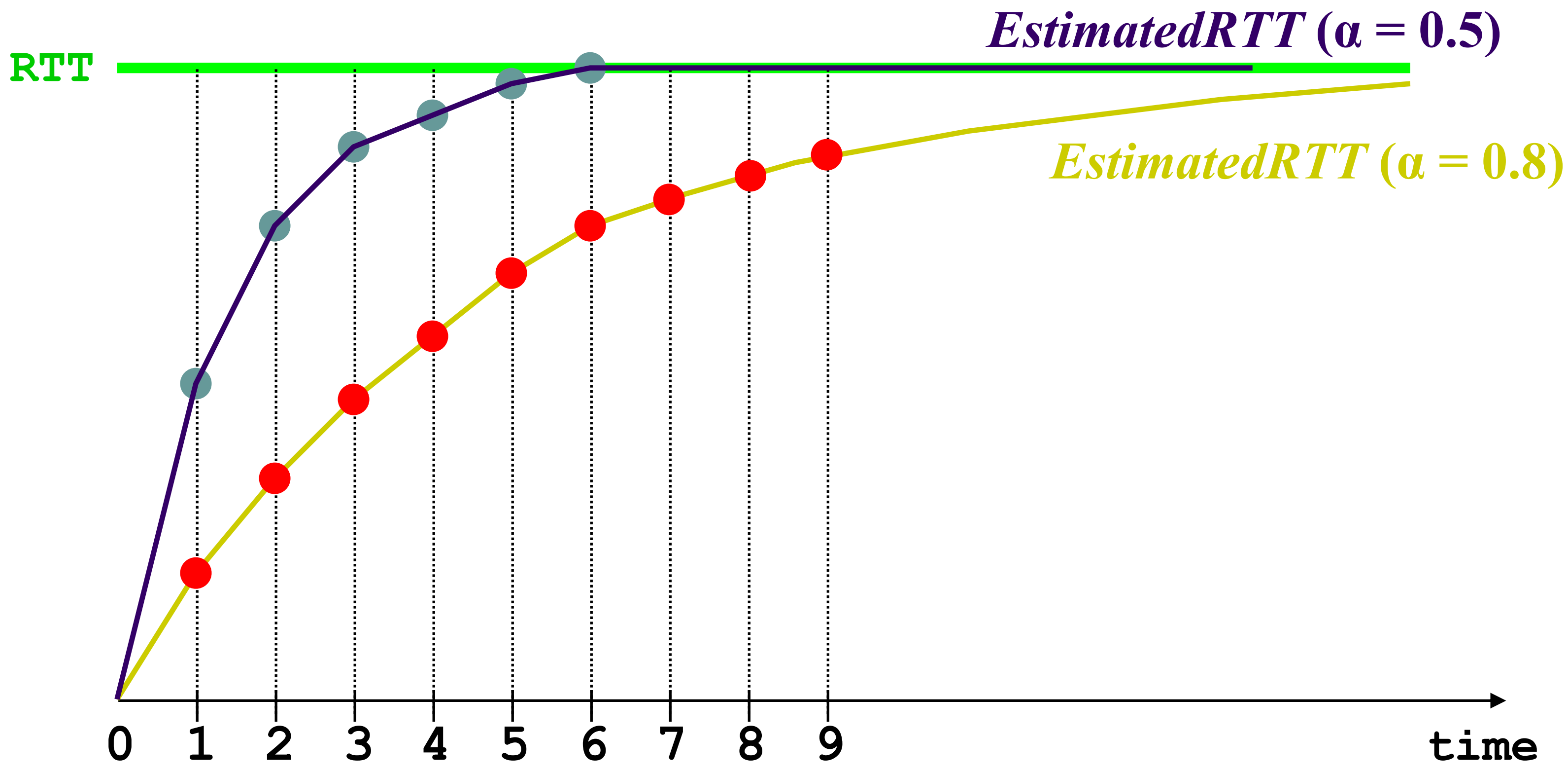
EstimatedRTT = α x EstimatedRTT + (1-α) x SampledRTT

0 < α <= 1



Exponential Averaging Example

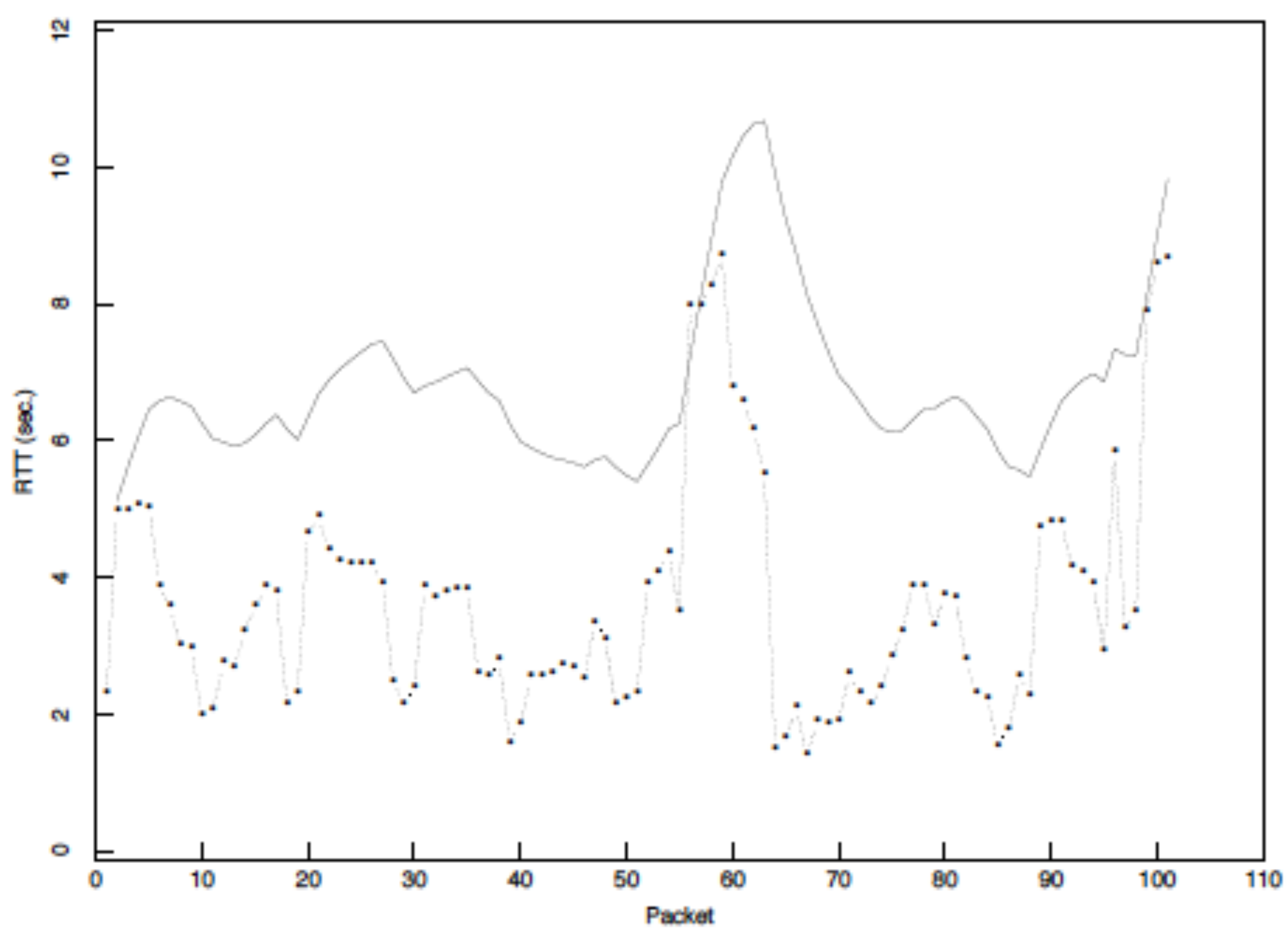
EstimatedRTT = α x EstimatedRTT + (1-α) x SampledRTT
(Assume RTT is constant => SampleRTT = RTT)



Exponential Averaging in Action

Set Timeout Estimate (ETO) = 2 x EstimatedRTT

Figure 5: Performance of an RFC793 retransmit timer



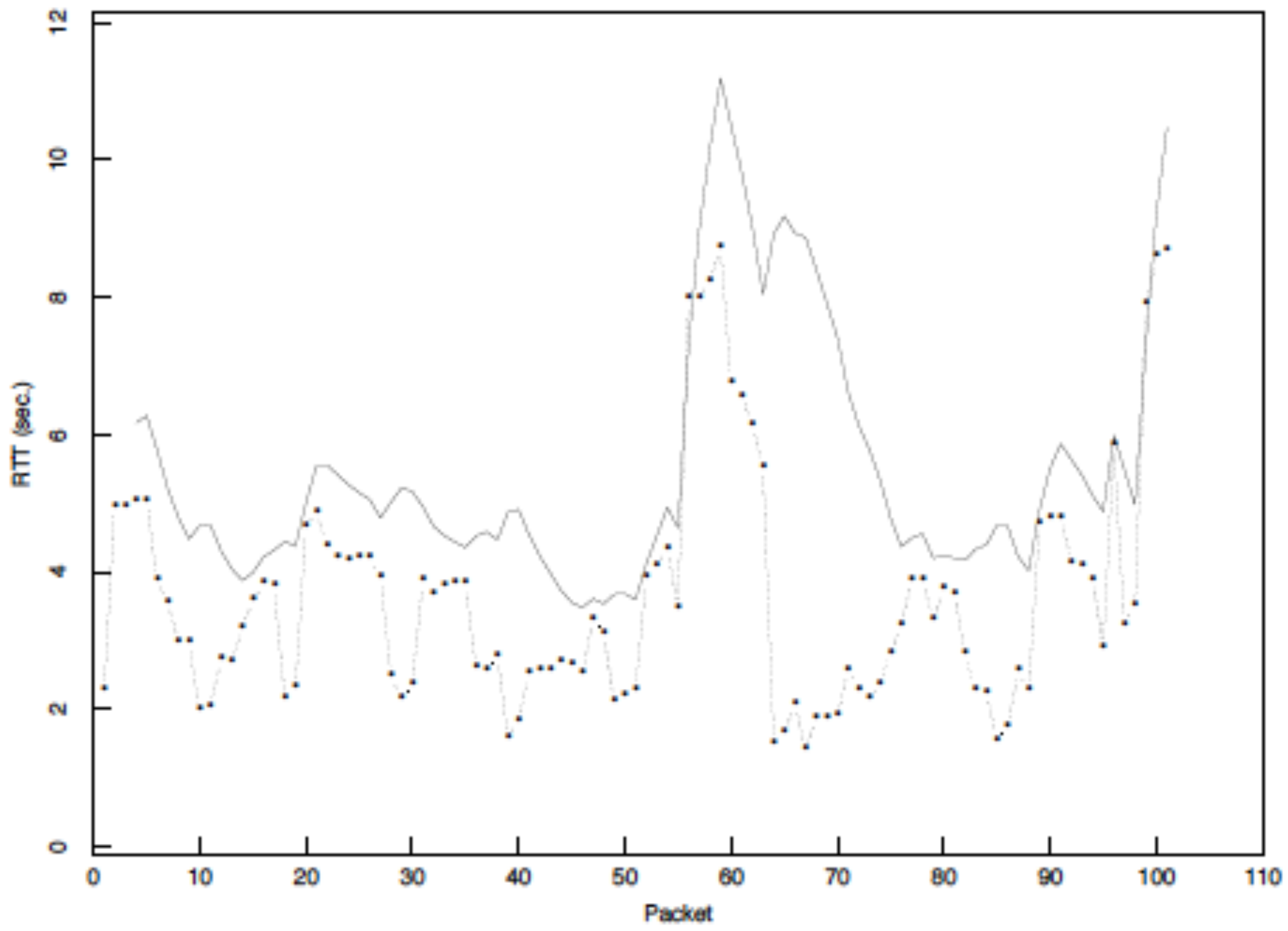
From Jacobson and Karels, SIGCOMM 1988

Jacobson/Karels Algorithm

- Problem: need to better capture variability in RTT
  - Directly measure deviation
- Deviation = | SampleRTT - EstimatedRTT|
- Estimated Deviation: exponential average of Deviation
- ETO = EstimatedRTT + 4 x EstimatedDeviation

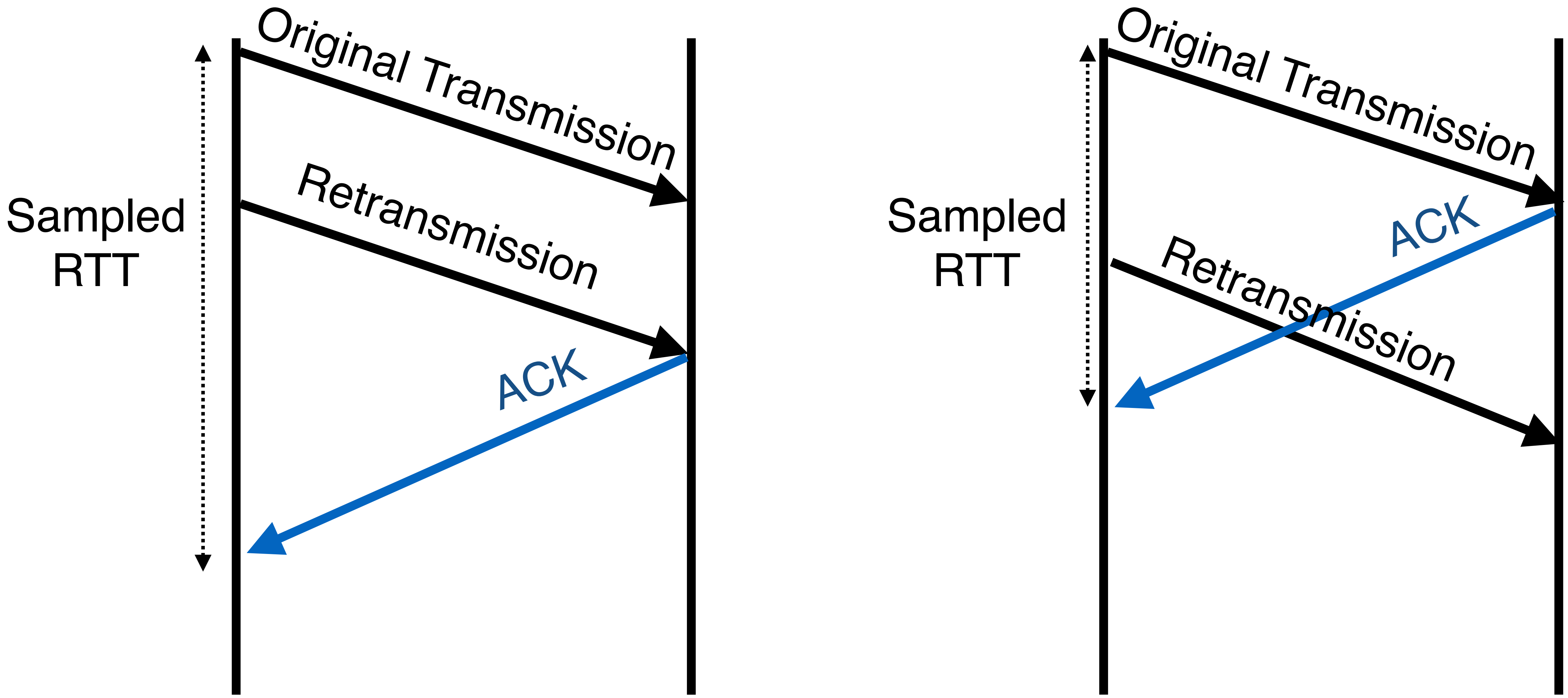
With Jacobson/Karels

Figure 6: Performance of a Mean+Variance retransmit timer



Problem: Ambiguous Measurements

- How do we differentiate between the real ACK, and ACK of the retransmitted packet?



TCP Timers

- Two important quantities
  - RTO: value you set timer to for timeouts
  - ETO: current estimate of appropriate “raw” timeout
- Use exponential averaging to estimate
  - RTT
  - Deviation = | Estimated RTT - Sample RTT |
- ETO = Estimated RTT + 4 x Estimated Deviation

Use Only “Clean” Samples for ETO

- Only update ETO when you get a clean sample
- Where clean means ACK includes no retransmitted segments

Example

- Send 100, 200, 300
    - 100 means packet whose first byte is 100, last byte is 199
- Receive A200
    - A200 means bytes up to 199 rep'd, expecting 200 next
    - Clean sample
- 200 times out, resend 200, receive A300
    - No clean samples
- Send 400, 500, receive A600
    - Clean samples

Setting RTO

- Every time RTO timer expires, set RTO <- 2.RTO
    - Upto maximum >= 60 sec
- Every time clean sample arrives set RTO to ETO

Example

- First arriving ACK expects 100 (adv. window=500)
  - Initialize ETP; RTO = ETO
  - Restart timer for RTO seconds (new data ACK’ed)
    - Remember TCP only has one timer, not timer per packet
  - Send packets 100, 200, 300, 400 and 500

- Arriving ACK expects 300 (A300)
  - Update ETO; RTO = ETO
  - Restart timer for RTO seconds (new data ACKed)
  - Send packets 600, 700

- Arriving ACK expects 300 (A300)

Example (cont'd)

- Timer goes off
    - RTO = 2*RTO (back off timer)
    - Restart timer for RTO seconds (it had expired)
    - Resend packet 300

- Arriving ACK expects 800
    - Don't update ETO (ACK includes a retransmission)
    - Restart timer for RTO seconds (new data ACKed)
    - Send packets 800, 900, 1000, 1100, 1200

Example (cont'd)

- Arriving ACK expects 1000
  - Updates ETO; RTO = ETO
  - Restart timer for RTO seconds (new data ACKed)
  - Send packets 1300, 1400
- … Connection continues…

Example

- Consider a TCP connection with:
    - CWND = 10 packets
    - Last ACK was for packet # 101
        - i.e., receiver expecting next packet to have seq no 101
- 10 packets [101, 102, 103, …, 110] are in flight
    - Packet 101 is dropped
    - What ACKs do they generate?
    - And how does the sender respond?

Timeline

- ACK 101(due to 102) CWND = 10 dupACK #1 (no xmit)
- ACK 101(due to 103) CWND = 10 dupACK #2 (no xmit)
- ACK 101(due to 104) CWND = 10 dupACK #3 (no xmit)
- RETRANSMIT 101 ssthresh = 5 CWND = 5
- ACK 101 (due to 105) CWND=5 (no xmit)
- ACK 101 (due to 106) CWND=5 (no xmit)
- ACK 101 (due to 107) CWND=5 (no xmit)
- ACK 101 (due to 108) CWND=5 (no xmit)
- ACK 101 (due to 109) CWND=5 (no xmit)
- ACK 101 (due to 110) CWND=5 (no xmit)
- ACK 111 (due to 101)<- only now can we transmit new packets
- Plus no packets in flight so no ACKs for another RTT

Note that you do not restart dupACKcounter on same packet!

Solution: Fast Recovery

- Idea: Grant the sender temporary "credit" for each dupACK so as to keep packets in flight (each ACK due to arriving pkt)
- If dupACKcount = 3
  - ssthresh = CWND / 2
  - CWND = ssthresh + 3
- While in fast recovery
  - CWND = CWND + 1 for each additional duplicate pet
- Exit fast recovery after receiving new ACK
  - Set CWND = ssthresh (which had been set to CWND/2 after loss)

Example

- Consider a TCP connection with:
    - CWND = 10 packets
    - Last ACK was for packet # 101
        - i.e., receiver expecting next packet to have seq no 101
- 10 packets [101, 102, 103, …, 110] are in flight
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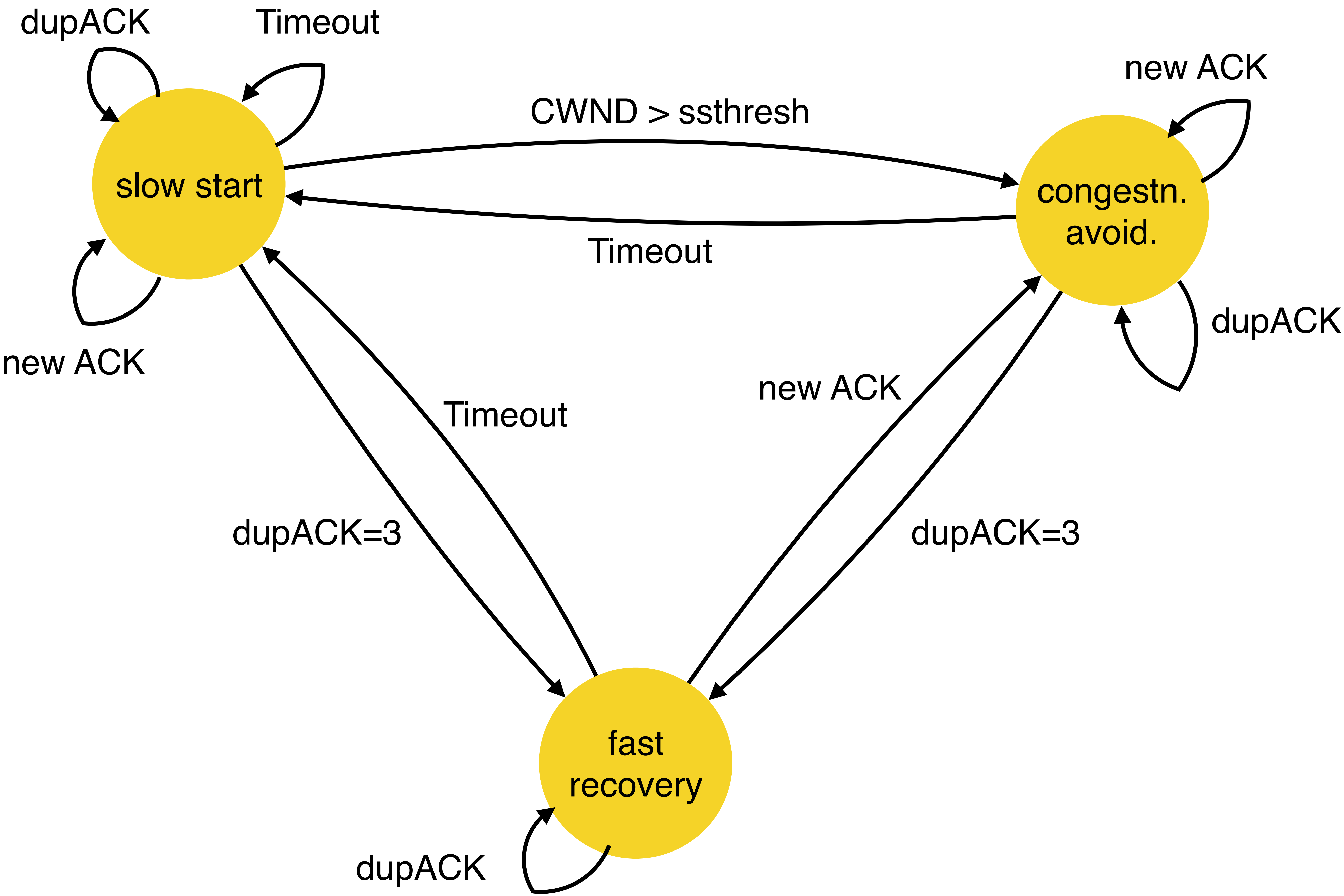
Timeline

- ACK 101(due to 102) CWND = 10 dupACK #1 (no xmit)
- ACK 101(due to 103) CWND = 10 dupACK #2 (no xmit)
- ACK 101(due to 104) CWND = 10 dupACK #3 (no xmit)
- RETRANSMIT 101 ssthresh = 5 CWND = 8 (5 + 3)
- ACK 101 (due to 105) CWND=9 (no xmit)
- ACK 101 (due to 106) CWND=10 (no xmit)
- ACK 101 (due to 107) CWND=11 (xmit 111)
- ACK 101 (due to 108) CWND=12 (xmit 112)
- ACK 101 (due to 109) CWND=13 (xmit 113)
- ACK 101 (due to 110) CWND=14 (xmit 114)
- ACK 111 (due to 101) CWND = 5 (xmit 115) <- exiting fast recovery
- Packets 111-114 already in flight (and not sending 115)
- ACK 112 (due to 111) CWND = 5 + 1/5 <- back to congestion avoidance

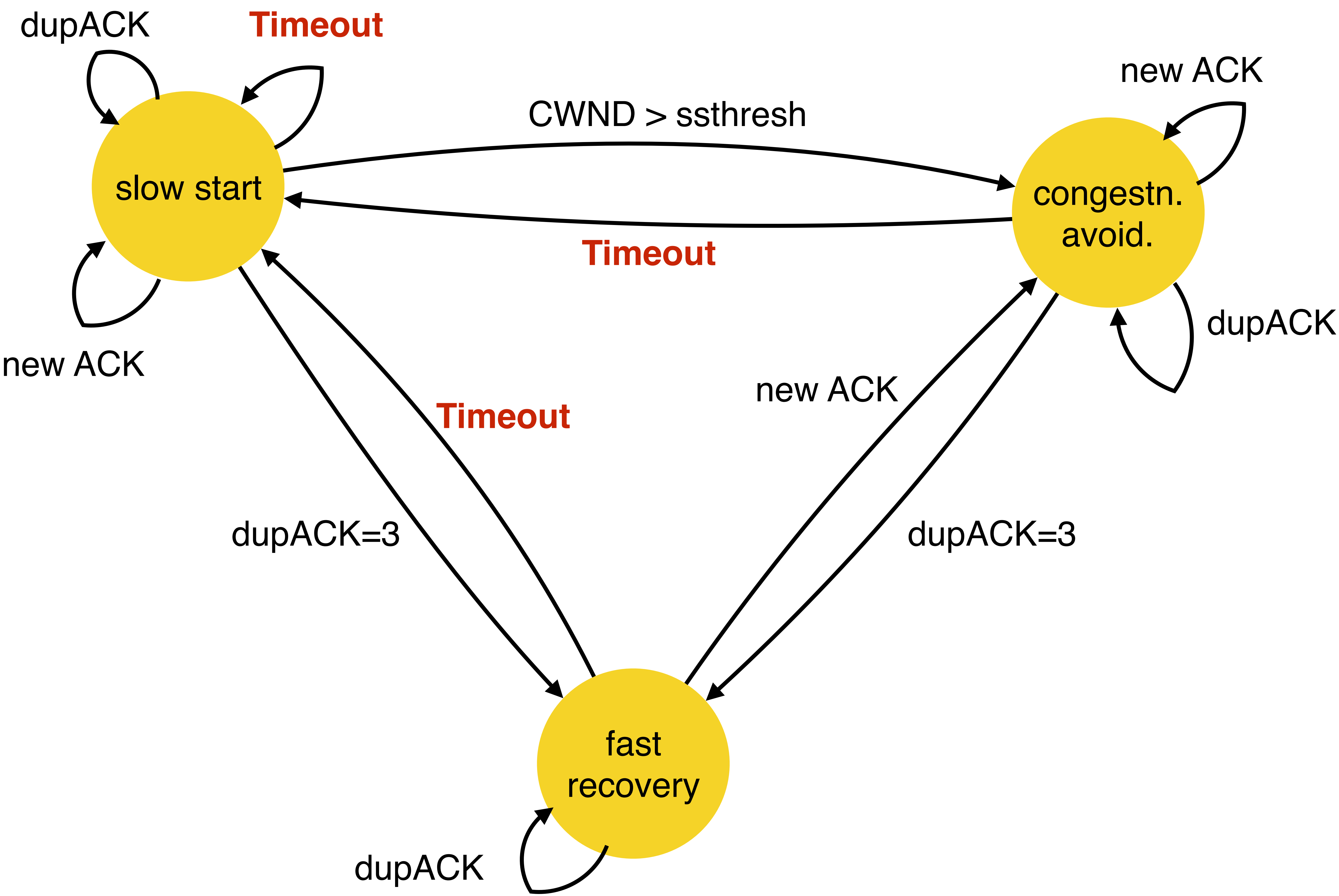
TCP “Phases”

- Slow-start
    - Enter on timeout
    - Leave when CWND > ssthresh (to Cong. Avoid.)
        - The > only applies here…
- Congestion Avoidance
    - Leave when timeout
- Fast recovery
    - Enter when dupACK=3
    - Leave when New ACK or Timeout

TCP State Machine



TCP State Machine



TCP State Machine