Computer Networks

Lecture 11: Transport Layer

Based on slides from D. Choffnes Northeastern U. and P. Gill from StonyBrook University
Revised Autumn 2015 by S. Laki

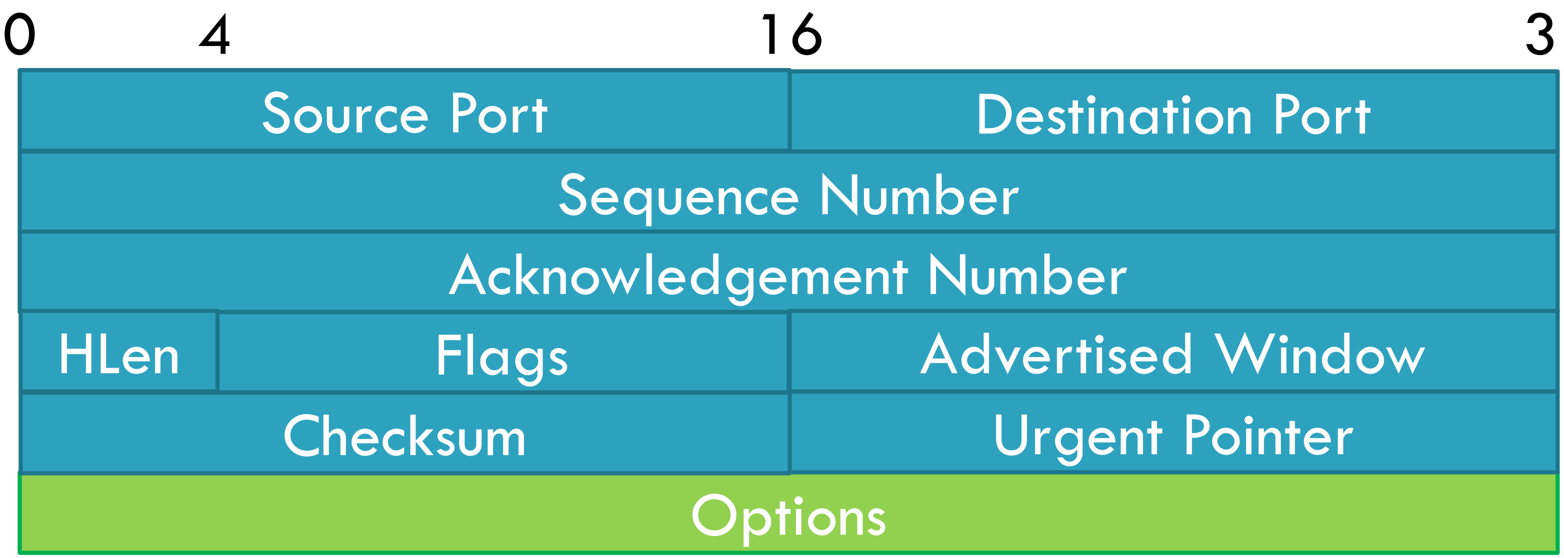
Outline

- UDP – already discussed
- TCP
- Congestion Control
- Evolution of TCP
- Problems with TCP

Transmission Control Protocol

- Reliable, in-order, bi-directional byte streams
  - Port numbers for demultiplexing
  - Virtual circuits (connections)
  - Flow control
  - Congestion control, approximate fairness

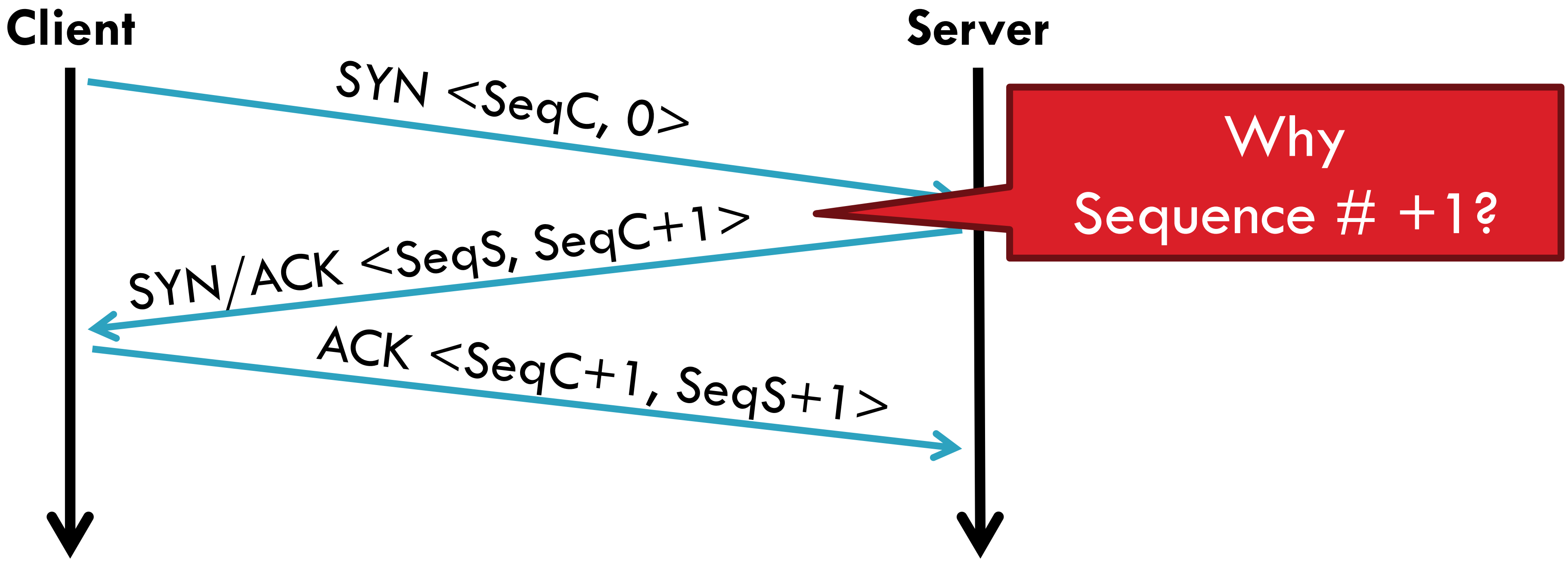
Why these features?



Connection Setup

- Why do we need connection setup?
  - To establish state on both hosts
  - Most important state: sequence numbers
    - Count the number of bytes that have been sent
    - Initial value chosen at random
    - Why?
- Important TCP flags (1 bit each)
  - SYN – synchronization, used for connection setup
  - ACK – acknowledge received data
  - FIN – finish, used to tear down connection

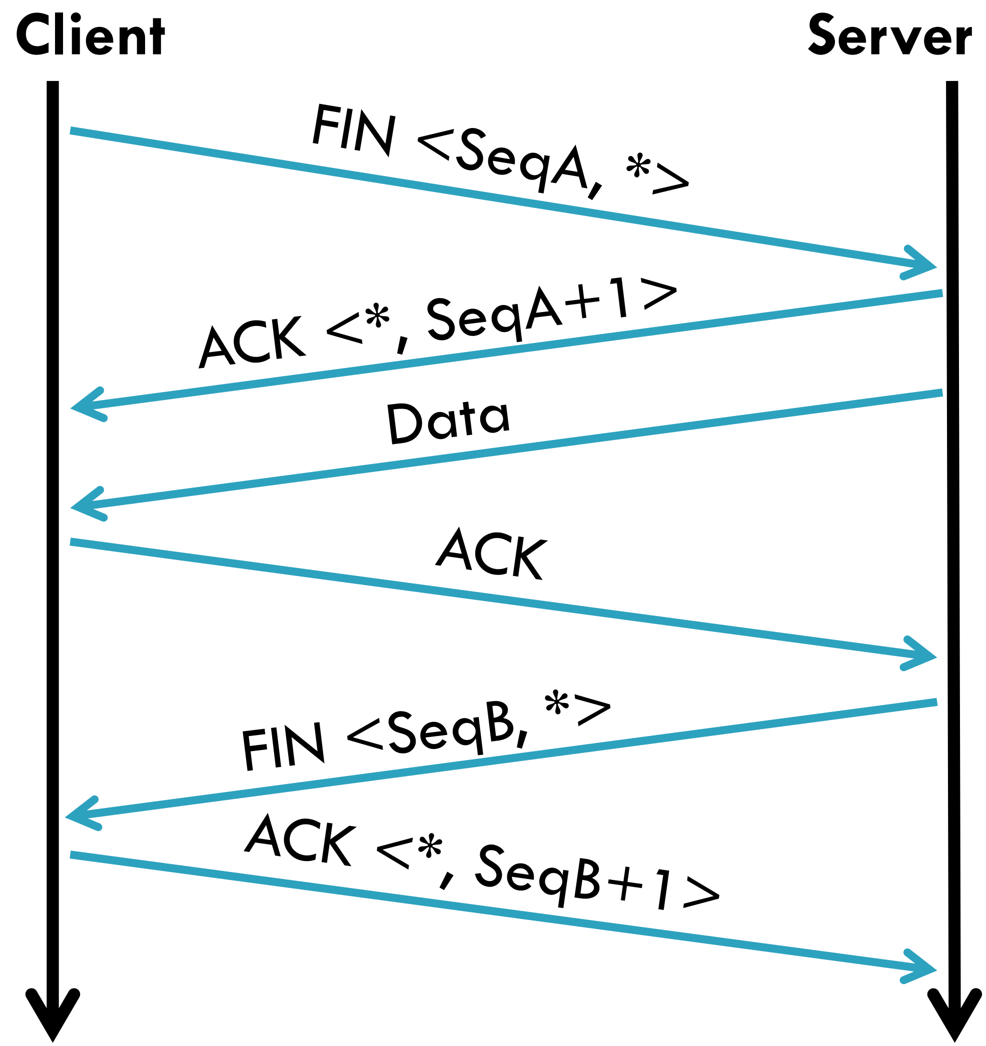
Three Way Handshake



- Each side:
  - Notifies the other of starting sequence number
  - ACKs the other side’s starting sequence number

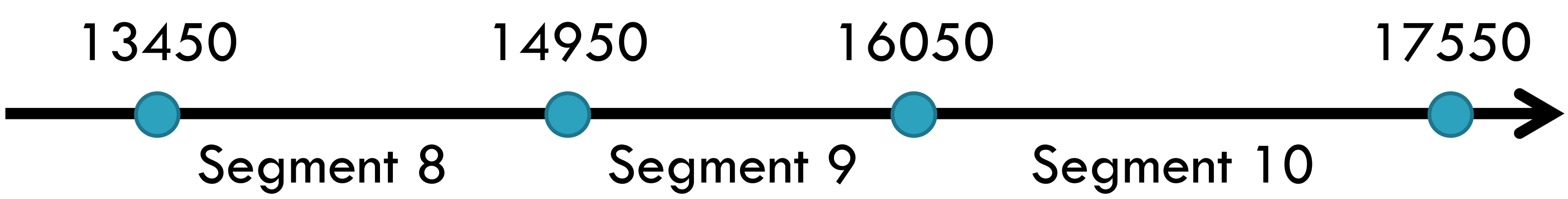
Connection Tear Down

- Either side can initiate tear down
- Other side may continue sending data
  - Half open connection
  - *shutdown()*
- Acknowledge the last FIN
  - Sequence number + 1
- What happens if 2nd FIN is lost?

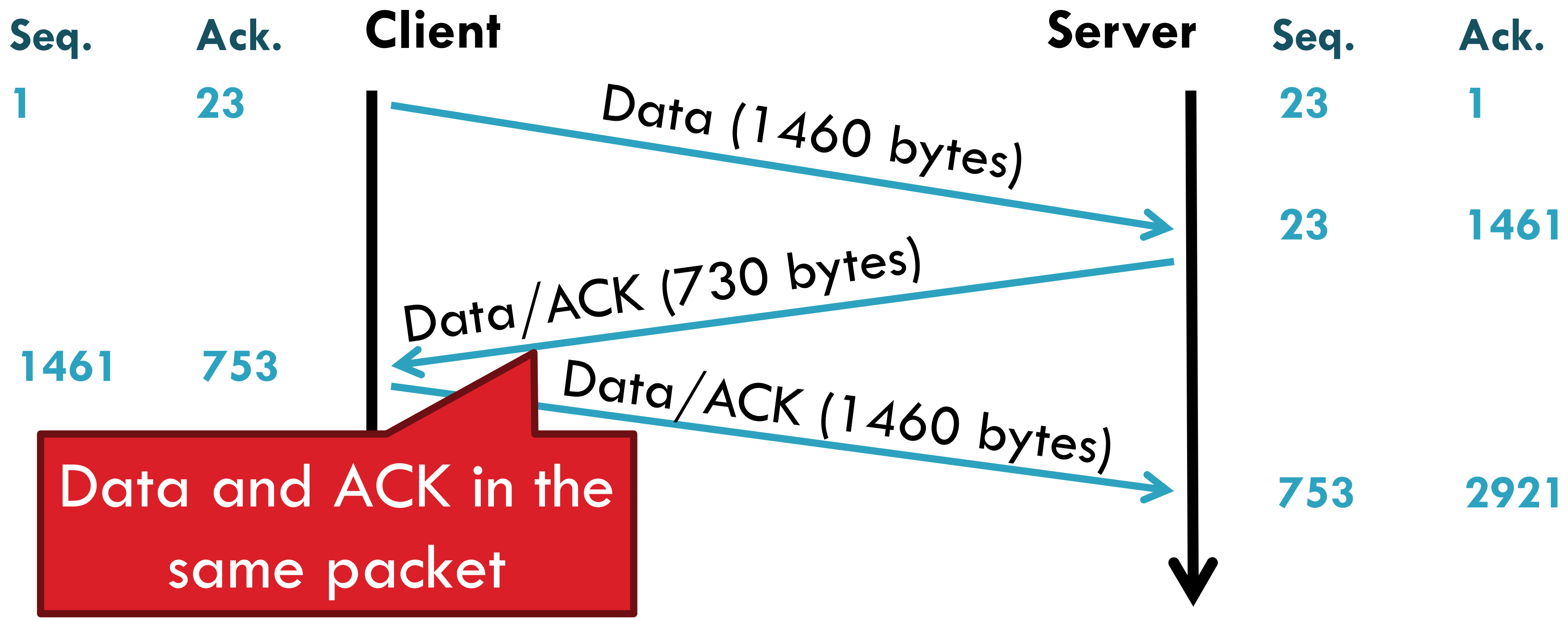


Sequence Number Space

- TCP uses a byte stream abstraction
  - Each byte in each stream is numbered
  - 32-bit value, wraps around
  - Initial, random values selected during setup. Why?
- Byte stream broken down into segments (packets)
  - Size limited by the Maximum Segment Size (MSS)
  - Set to limit fragmentation
- Each segment has a sequence number



Bidirectional Communication

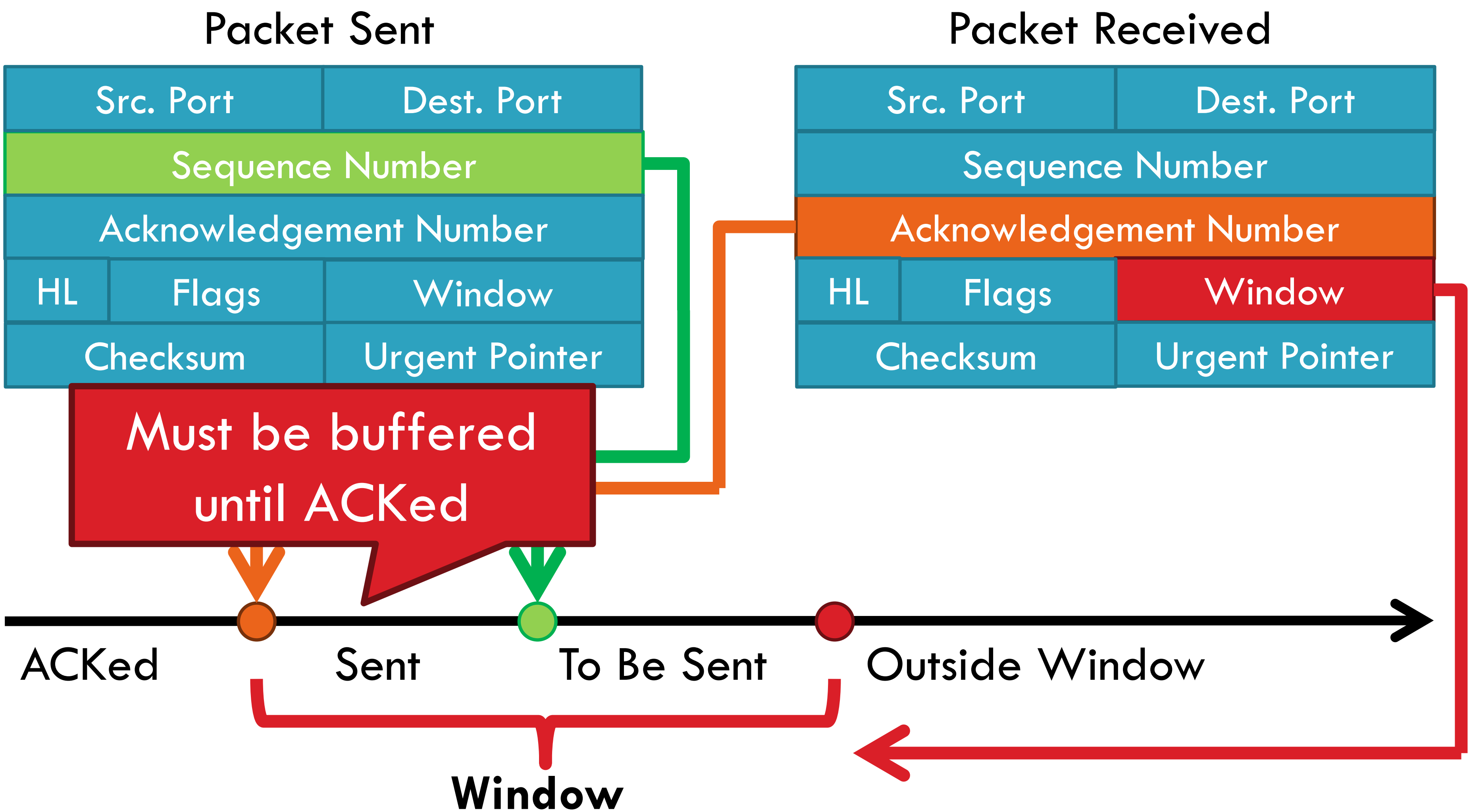


- Each side of the connection can send and receive
  - Different sequence numbers for each direction

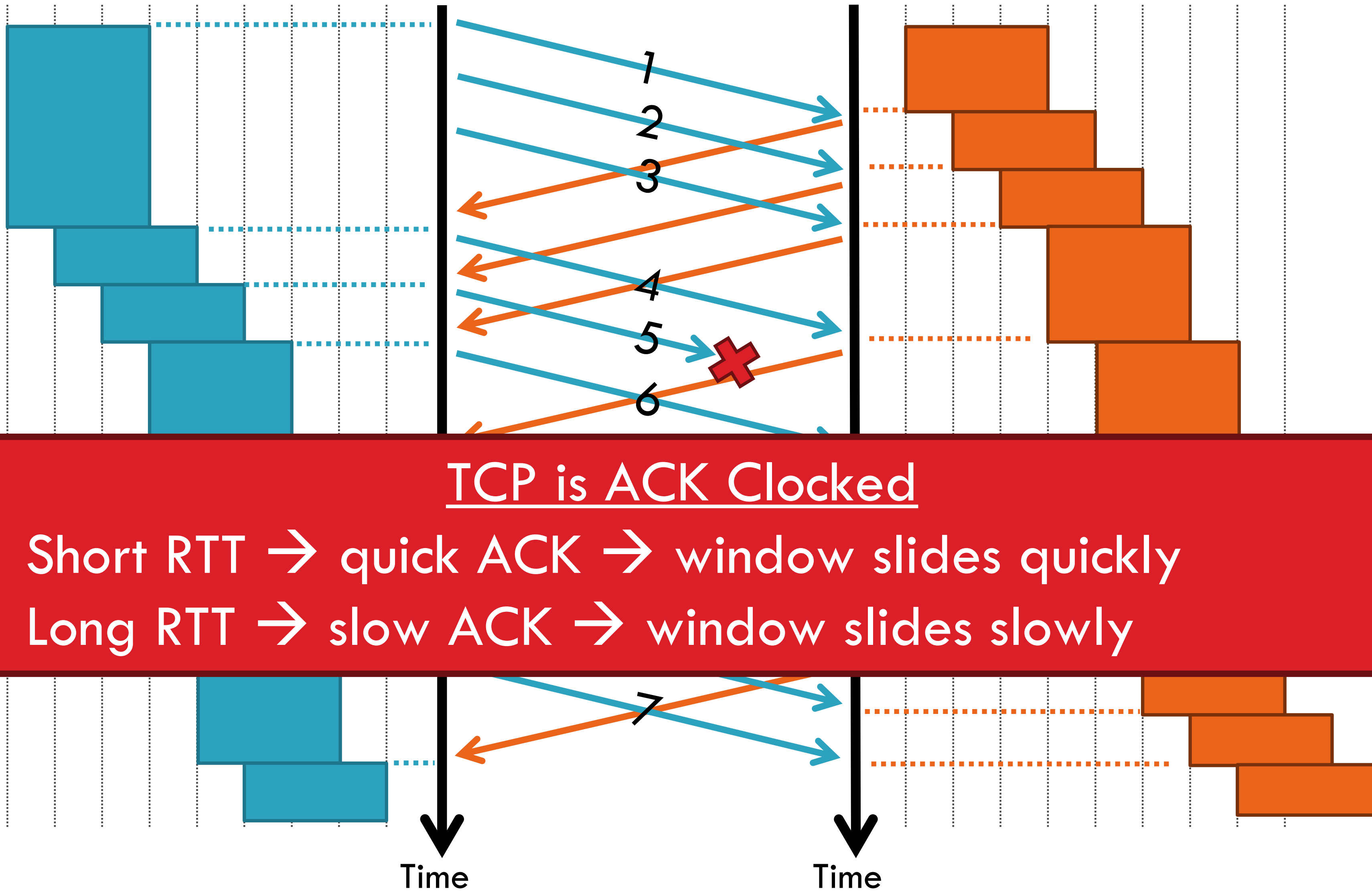
Flow Control

- Problem: how many packets should a sender transmit?
  - Too many packets may overwhelm the receiver
  - Size of the receivers buffers may change over time
- Solution: sliding window
  - Receiver tells the sender how big their buffer is
  - Called the advertised window
  - For window size *n*, sender may transmit *n* bytes without receiving an ACK
  - After each ACK, the window slides forward
- Window may go to zero!

Flow Control: Sender Side



Sliding Window Example



TCP is ACK Clocked

- Short RTT → quick ACK → window slides quickly
- Long RTT → slow ACK → window slides slowly

Observations

- Throughput is ~ w/RTT
- Sender has to buffer all unacknowledges packets, because they may require retransmission
- Receiver may be able to accept out-of-order packets, but only up to buffer limits

What Should the Receiver ACK?

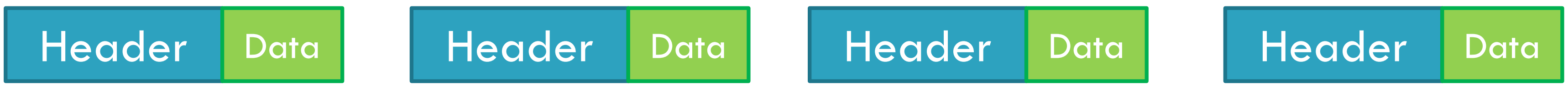
1. ACK every packet
2. Use *cumulative ACK*, where an ACK for sequence n implies ACKS for all $k < n$
3. Use *negative ACKs* (NACKs), indicating which packet did not arrive
4. Use *selective ACKs* (SACKs), indicating those that did arrive, even if not in order
   - SACK is an actual TCP extension

Sequence Numbers, Revisited

- 32 bits, unsigned
  - Why so big?
- For the sliding window you need…
  - |Sequence # Space| > 2 * |Sending Window Size|
  - $2^{32} > 2 * 2^{16}$
- Guard against stray packets
  - IP packets have a maximum segment lifetime (MSL) of 120 seconds
    - i.e. a packet can linger in the network for 2 minutes

Silly Window Syndrome

- Problem: what if the window size is very small?
  - Multiple, small packets, headers dominate data



- Equivalent problem: sender transmits packets one byte at a time

```
for (int x = 0; x < strlen(data); ++x)
  write(socket, data + x, 1);
```

Nagle's Algorithm

1. If the window >= MSS and available data >= MSS:
   Send the data
2. Elif there is unACKed data:
   Enqueue data in a buffer until an ACK is received
3. Else: send the data

Send a full packet

Send a non-full packet if nothing else is happening

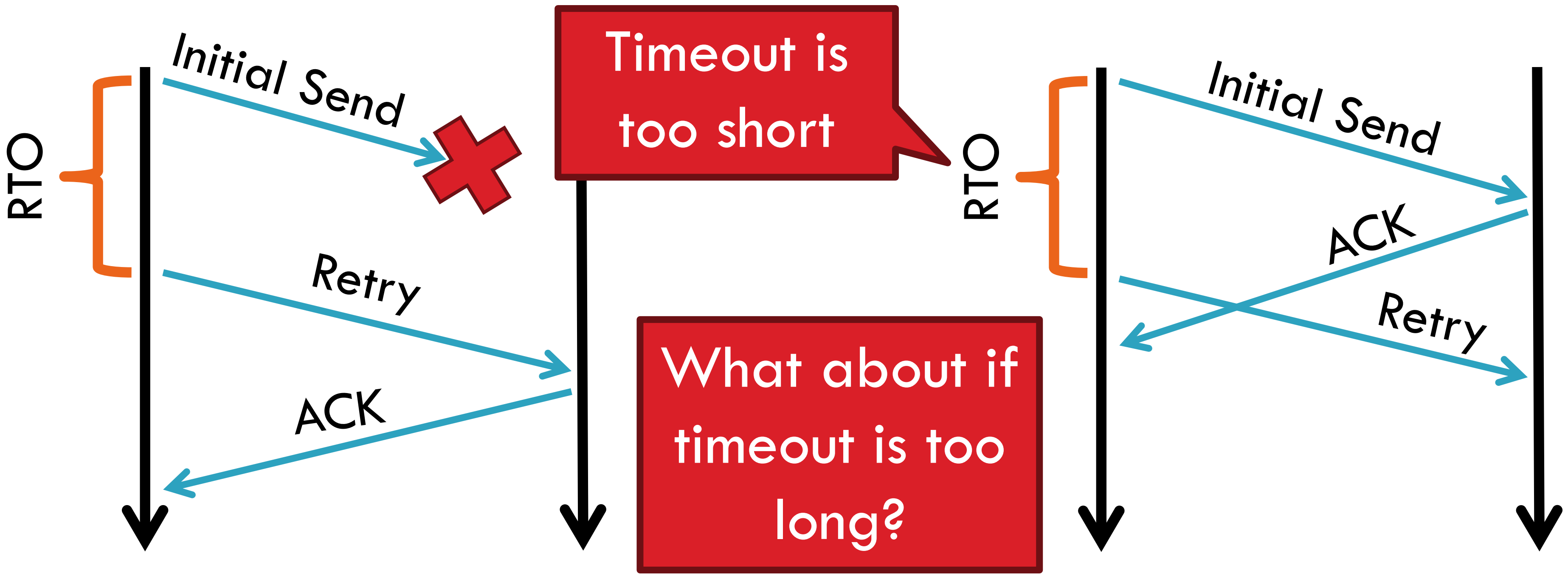
- Problem: Nagle's Algorithm delays transmissions
  - What if you need to send a packet immediately?
  1. int flag = 1;
  2. setsockopt(sock, IPPROTO_TCP, TCP_NODELAY, (char *) &flag, sizeof(int));

Error Detection

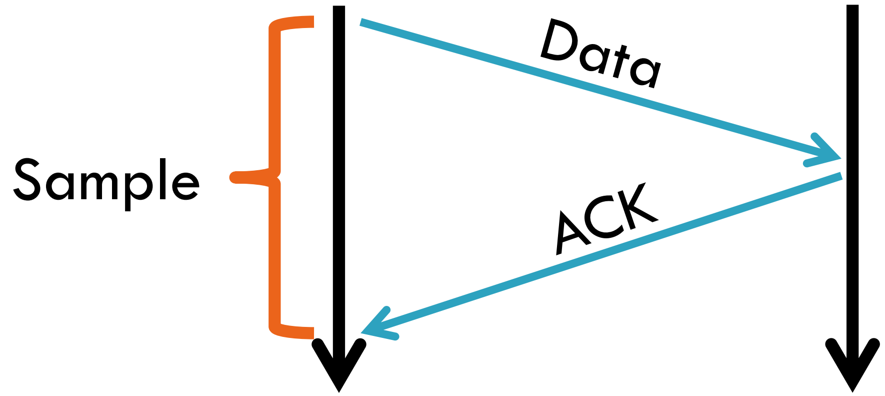
- Checksum detects (some) packet corruption
  - Computed over IP header, TCP header, and data
- Sequence numbers catch sequence problems
  - Duplicates are ignored
  - Out-of-order packets are reordered or dropped
  - Missing sequence numbers indicate lost packets
- Lost segments detected by sender
  - Use timeout to detect missing ACKs
  - Need to estimate RTT to calibrate the timeout
  - Sender must keep copies of all data until ACK

Retransmission Time Outs (RTO)

- Problem: time-out is linked to round trip time

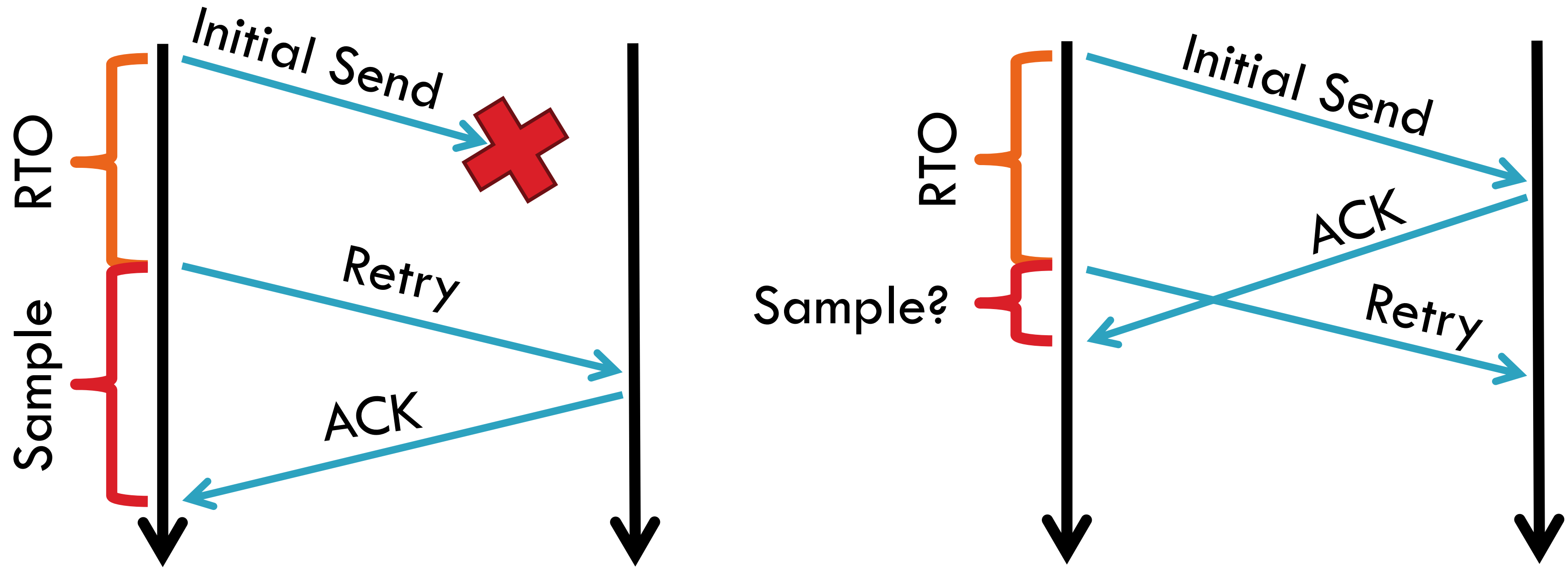


Round Trip Time Estimation



- Original TCP round-trip estimator
  - RTT estimated as a moving average
  - new_rtt = α (old_rtt) + (1 – α)(new_sample)
  - Recommended α: 0.8-0.9 (0.875 for most TCPs)
- RTO = 2 * new_rtt (i.e. TCP is conservative)

RTT Sample Ambiguity



- Karn's algorithm: ignore samples for retransmitted segments

TCP Congestion Control

- **The network is congested if the load in the network is higher than its capacity.**
- Each TCP connection has a window
  - Controls the number of unACKed packets
- Sending rate is ~ window/RTT
- Idea: vary the window size to control the send rate
- Introduce a congestion window at the sender
  - Congestion control is sender-side problem

Two Basic Components

1. Detect congestion
   - Packet dropping is most reliably signal
     - Delay-based methods are hard and risky
   - How do you detect packet drops? ACKs
     - Timeout after not receiving an ACK
     - Several duplicate ACKs in a row (ignore for now)
2. Rate adjustment algorithm
   - Modify *cwnd*
   - Probe for bandwidth
   - Responding to congestion

Rate Adjustment

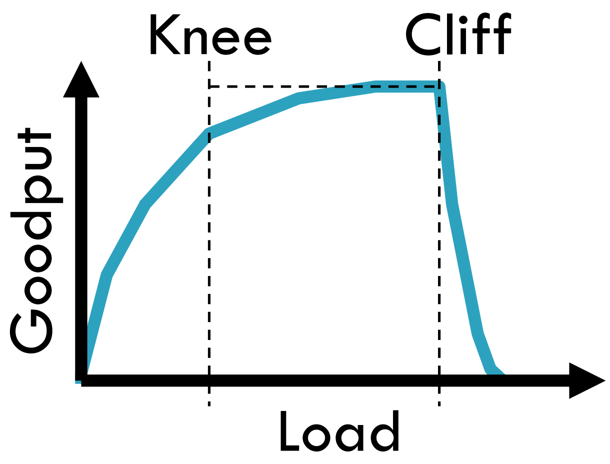
- Recall: TCP is ACK clocked
  - Congestion = delay = long wait between ACKs
  - No congestion = low delay = ACKs arrive quickly
- Basic algorithm
  - Upon receipt of ACK: increase cwnd
    - Data was delivered, perhaps we can send faster
    - *cwnd* growth is proportional to RTT
  - On loss: decrease cwnd
    - Data is being lost, there must be congestion
- Question: increase/decrease functions to use? !!!!

Implementing Congestion Control

- Maintains three variables:
  - *cwnd*: congestion window
  - *adv_wnd*: receiver advertised window
  - *ssthresh*: threshold size (used to update *cwnd*)
- For sending, use: *wnd = min(cwnd, adv_wnd)*
- Two phases of congestion control
  1. Slow start (*cwnd* < *ssthresh*)
     - Probe for bottleneck bandwidth
  2. Congestion avoidance (*cwnd* >= *ssthresh*)
     - AIMD

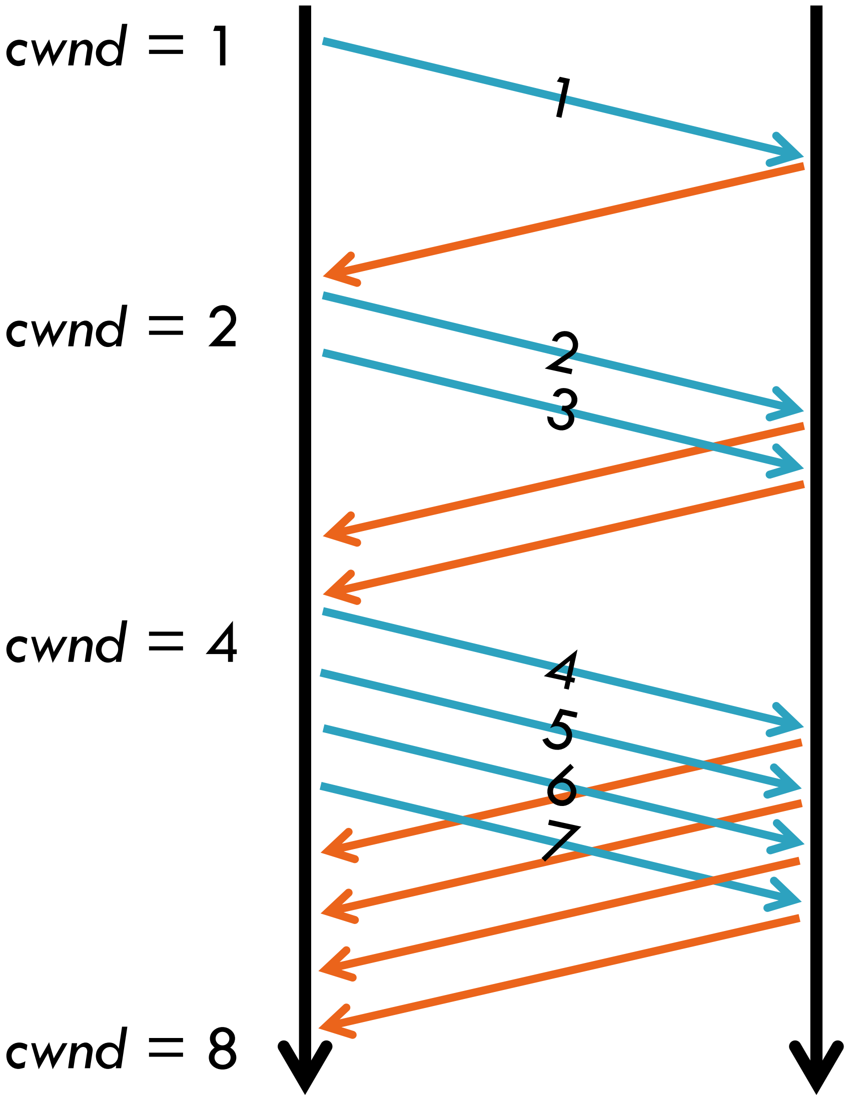
Slow Start

- Goal: reach knee quickly
- Upon starting (or restarting) a connection
  - *cwnd* =1
  - *ssthresh* = *adv_wnd*
  - Each time a segment is ACKed, *cwnd*++
- Continues until…
  - *ssthresh* is reached
  - Or a packet is lost
- Slow Start is not actually slow
  - *cwnd* increases exponentially



Slow Start Example

- *cwnd* grows rapidly
- Slows down when…
  - *cwnd* >= *ssthresh*
  - Or a packet drops



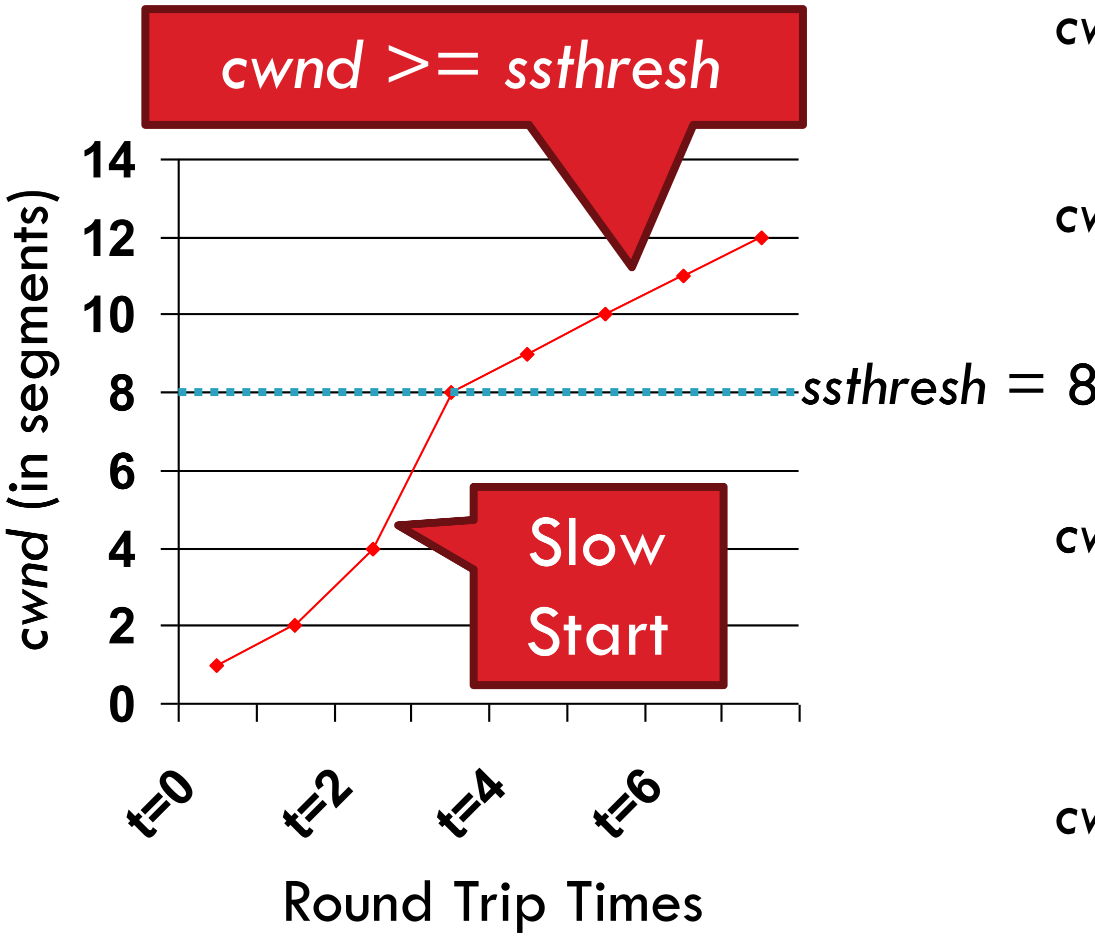
Congestion Avoidance

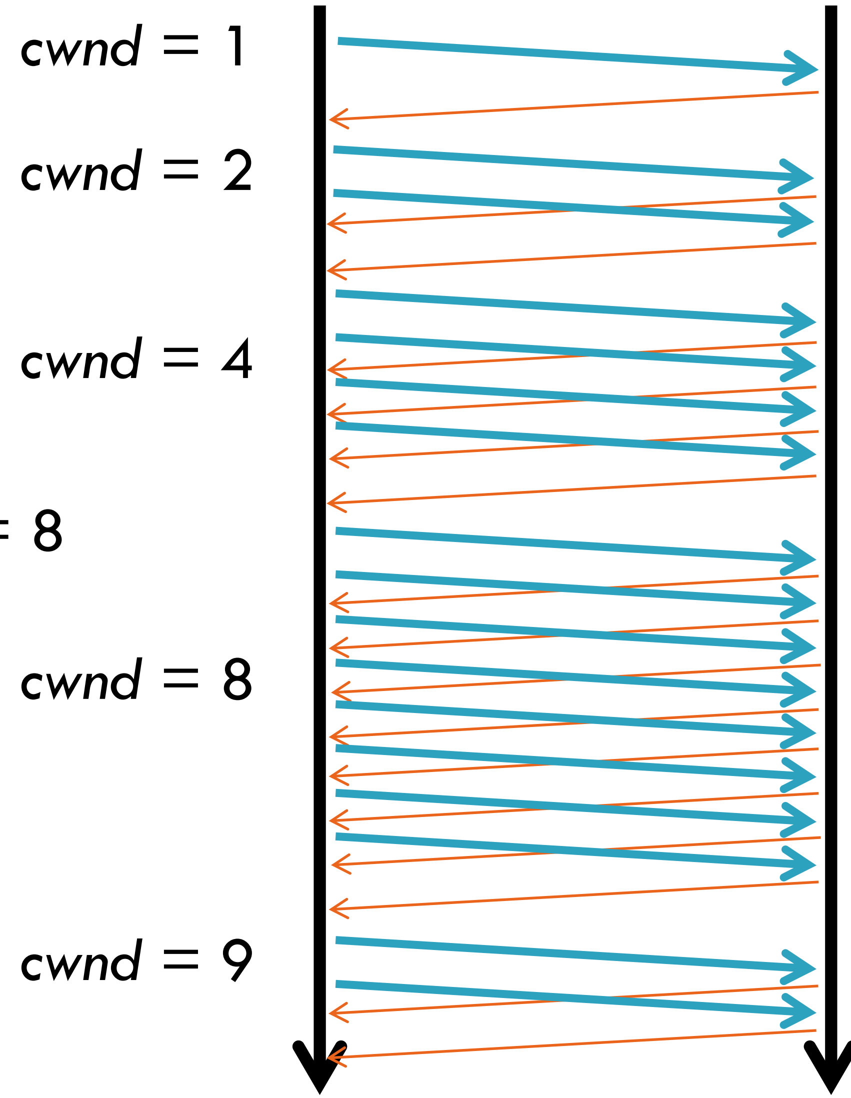
- Additive Increase Multiplicative Decrease (AIMD) mode
- *ssthresh* is lower-bound guess about location of the knee
- **If** *cwnd* >= *ssthresh* **then**

  each time a segment is ACKed

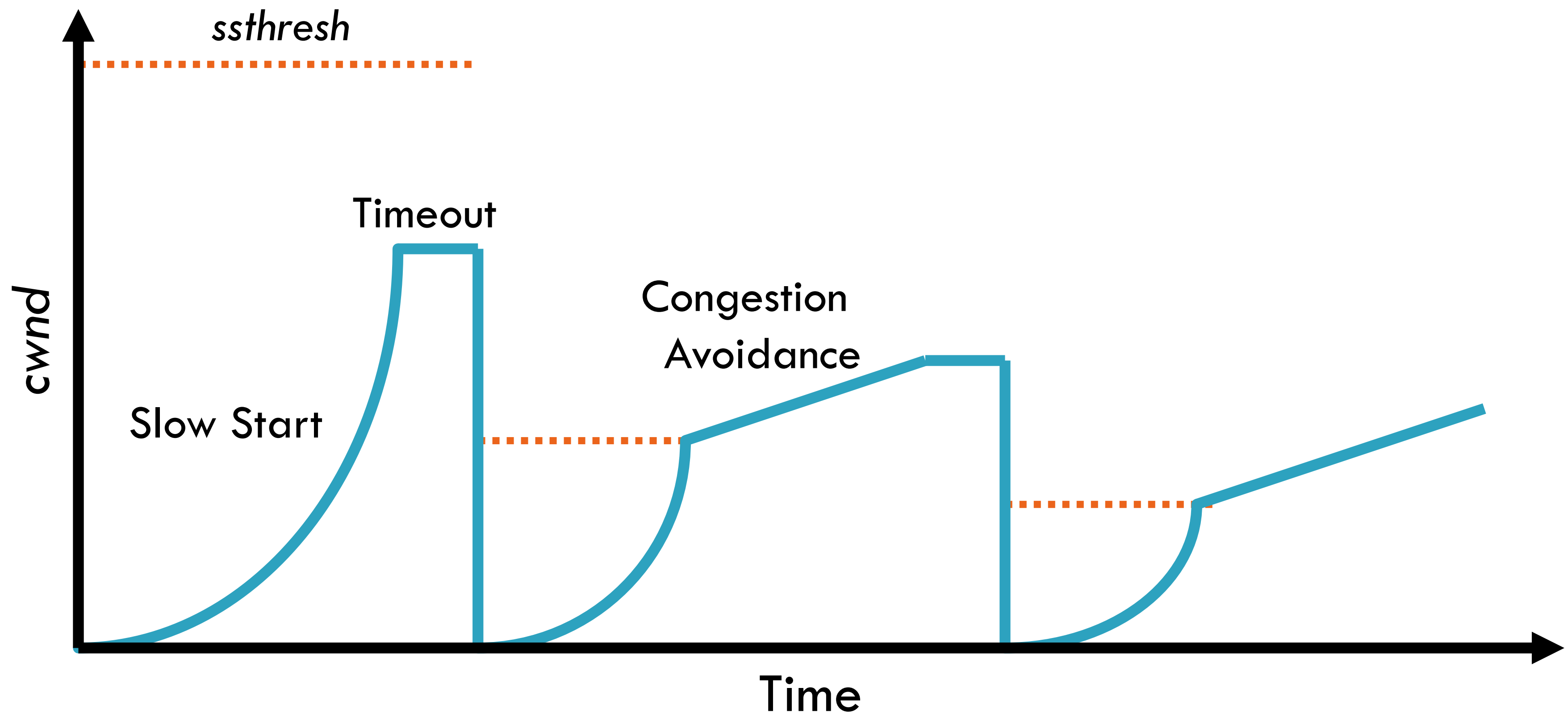
  increment *cwnd by 1/cwnd* (*cwnd* += *1/cwnd*).
- So *cwnd* is increased by one only if all segments have been acknowledged

Congestion Avoidance Example





The Big Picture – TCP Tahoe (the original TCP)



Outline

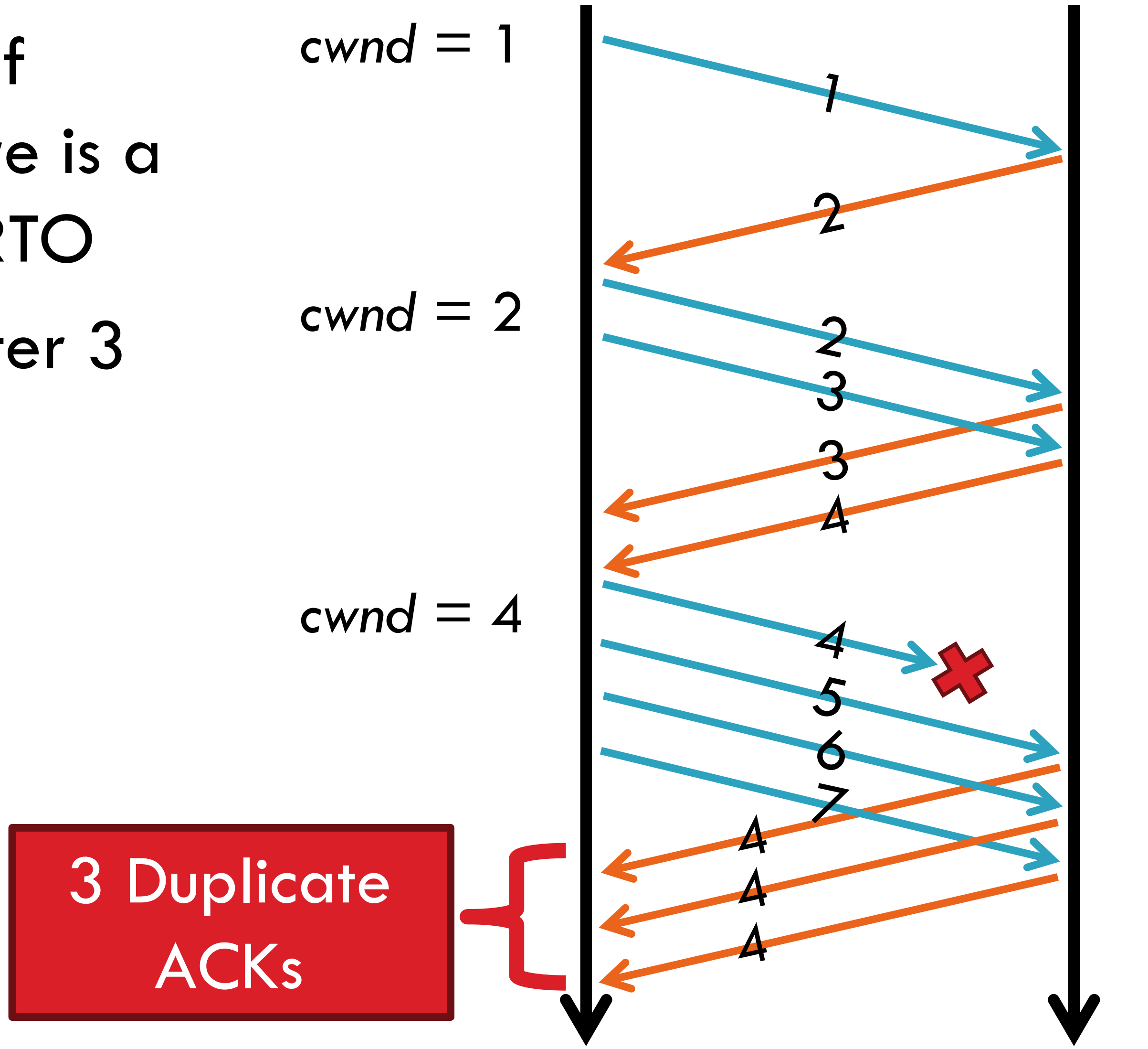
- UDP
- TCP
- Congestion Control
- **Evolution of TCP**
- Problems with TCP

The Evolution of TCP

- Thus far, we have discussed TCP Tahoe
  - Original version of TCP
- However, TCP was invented in 1974!
  - Today, there are many variants of TCP
- Early, popular variant: TCP Reno
  - Tahoe features, plus…
  - Fast retransmit
    - 3 duplicate ACKs? -> retransmit (don't wait for RTO)
  - Fast recovery
    - On loss: set cwnd = cwnd/2 (ssthresh = new cwnd value)

TCP Reno: Fast Retransmit

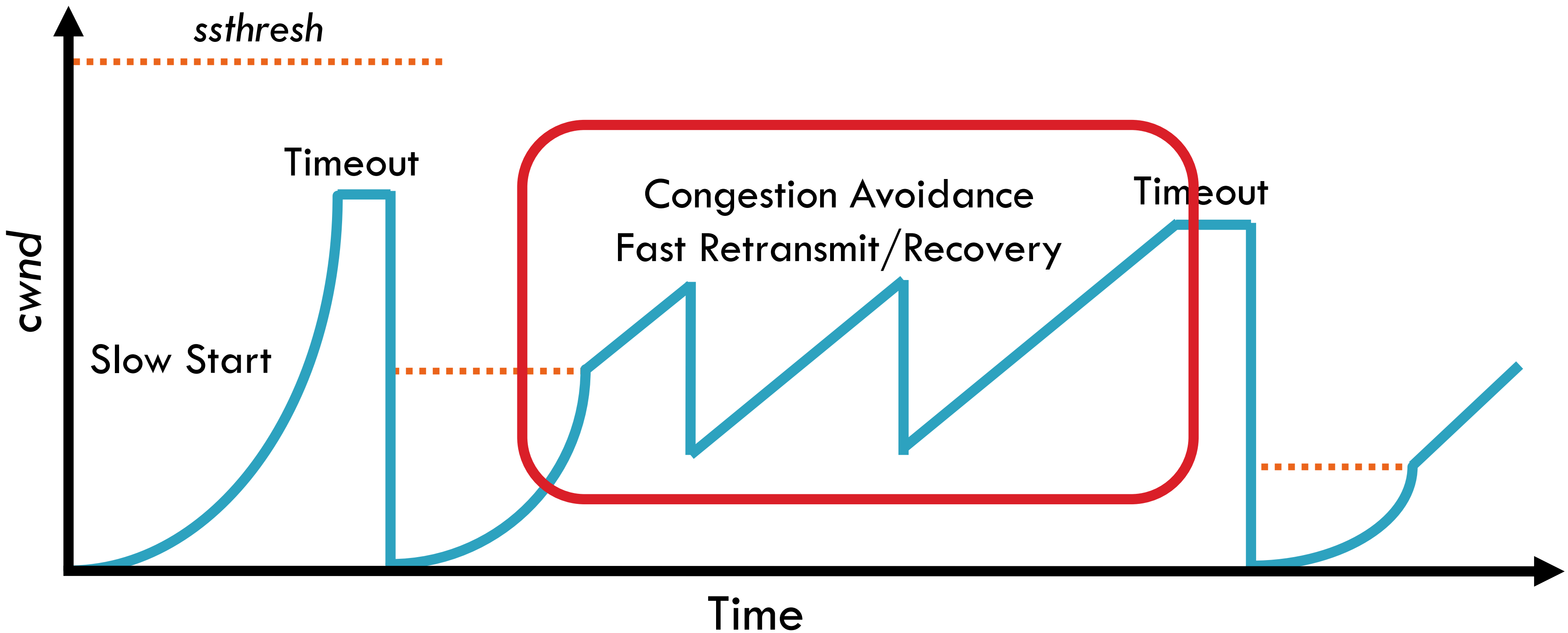
- Problem: in Tahoe, if segment is lost, there is a long wait until the RTO
- Reno: retransmit after 3 duplicate ACKs



TCP Reno: Fast Recovery

- After a fast-retransmit set *cwnd* to *cwnd*/2
  - Also reset ssthresh to the new halved cwnd value
  - i.e. don't reset *cwnd* to 1
  - Avoid unnecessary return to slow start
  - Prevents expensive timeouts
- But when RTO expires still do *cwnd* = 1
  - Return to slow start, same as Tahoe
  - Indicates packets aren't being delivered at all
  - i.e. congestion must be really bad

Fast Retransmit and Fast Recovery



- At steady state, *cwnd* oscillates around the optimal window size
- TCP always forces packet drops

Many TCP Variants…

- Tahoe: the original
  - Slow start with AIMD
  - Dynamic RTO based on RTT estimate
- Reno:
  - fast retransmit (3 dupACKs)
  - fast recovery (cwnd = cwnd/2 on loss)
- NewReno: improved fast retransmit
  - Each duplicate ACK triggers a retransmission
  - Problem: >3 out-of-order packets causes pathological retransmissions
- Vegas: delay-based congestion avoidance
- And many, many, many more…

TCP in the Real World

- What are the most popular variants today?
  - Key problem: TCP performs poorly on high bandwidth-delay product networks (like the modern Internet)
  - Compound TCP (Windows)
    - Based on Reno
    - Uses two congestion windows: delay based and loss based
    - Thus, it uses a *compound* congestion controller
  - TCP CUBIC (Linux)
    - Enhancement of BIC (Binary Increase Congestion Control)
    - Window size controlled by cubic function
    - Parameterized by the time *T* since the last dropped packet

High Bandwidth-Delay Product

- Key Problem: TCP performs poorly when
  - The capacity of the network (bandwidth) is large
  - The delay (RTT) of the network is large
  - Or, when bandwidth * delay is large
    - b * d = maximum amount of in-flight data in the network
    - a.k.a. the bandwidth-delay product
- Why does TCP perform poorly?
  - Slow start and additive increase are slow to converge
  - TCP is ACK clocked
    - i.e. TCP can only react as quickly as ACKs are received
    - Large RTT → ACKs are delayed → TCP is slow to react

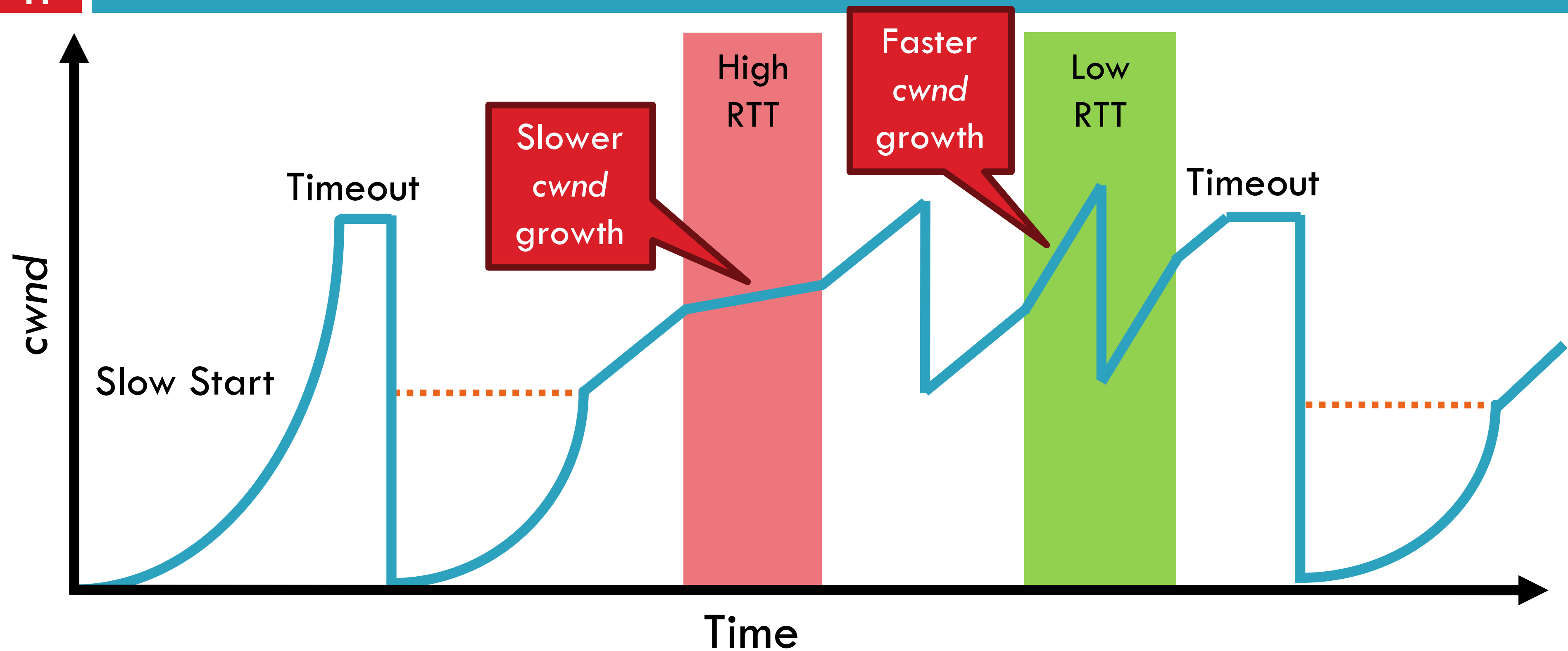
Goals

- Fast window growth
  - Slow start and additive increase are too slow when bandwidth is large
  - Want to converge more quickly
- Maintain fairness with other TCP varients
  - Window growth cannot be too aggressive
- Improve RTT fairness
  - TCP Tahoe/Reno flows are not fair when RTTs vary widely
- Simple implementation

Compound TCP Implementation

- Default TCP implementation in Windows
- Key idea: split *cwnd* into two separate windows
  - Traditional, loss-based window
  - New, delay-based window
- *wnd* = min(*cwnd* + *dwnd*, *adv_wnd*)
  - *cwnd* is controlled by AIMD
  - *dwnd* is the delay window
- Rules for adjusting *dwnd:*
  - If RTT is increasing, decrease *dwnd* (*dwnd* >= 0)
  - If RTT is decreasing, increase *dwnd*
  - Increase/decrease are proportional to the rate of change

Compound TCP Example



- Aggressiveness corresponds to changes in RTT
- Advantages: fast ramp up, more fair to flows with different RTTs
- Disadvantage: must estimate RTT, which is very challenging

TCP CUBIC Implementation

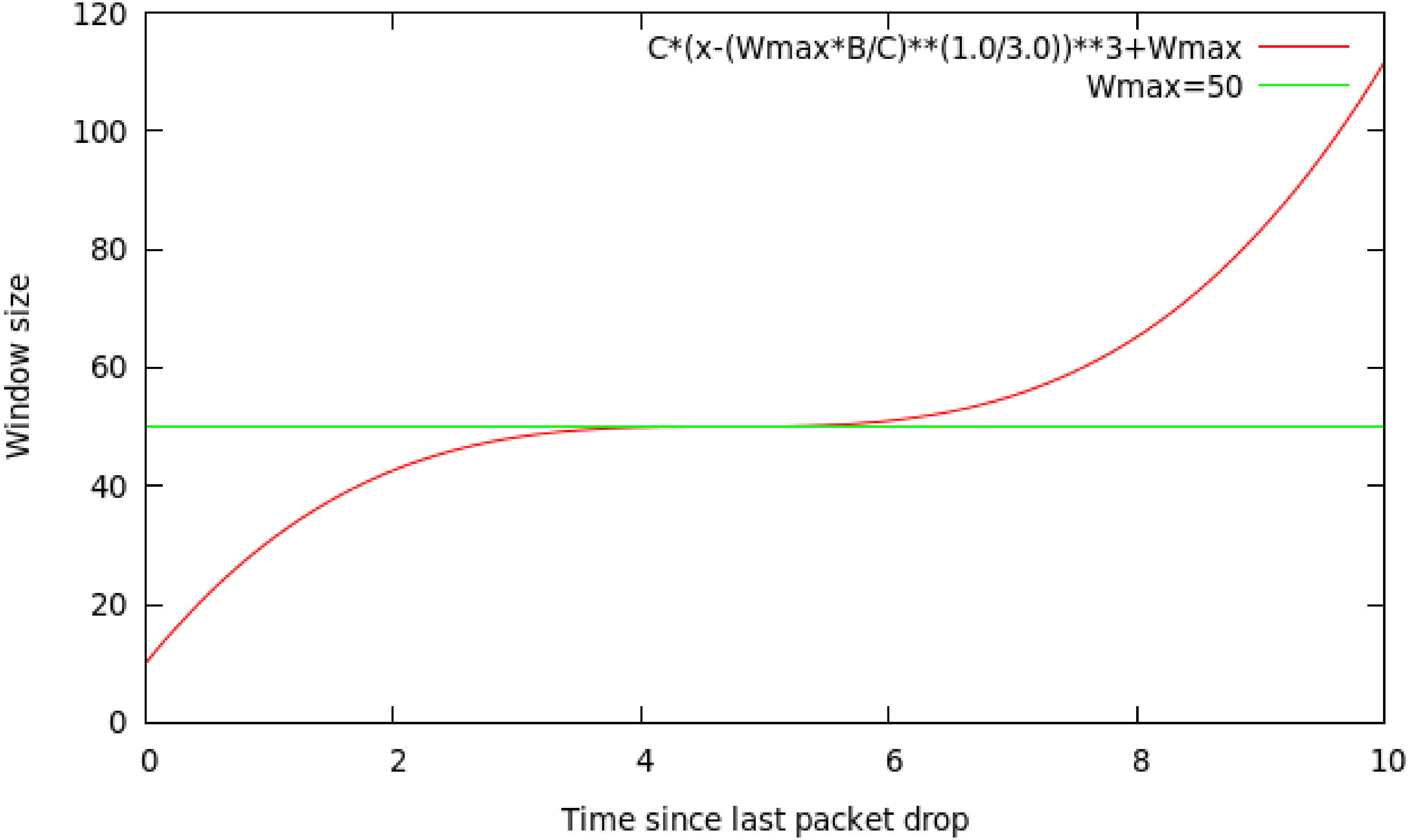
- Default TCP implementation in Linux
- Replace AIMD with cubic function

$$W_{cubic} = C(T - K)^3 + W_{max} \qquad (1)$$

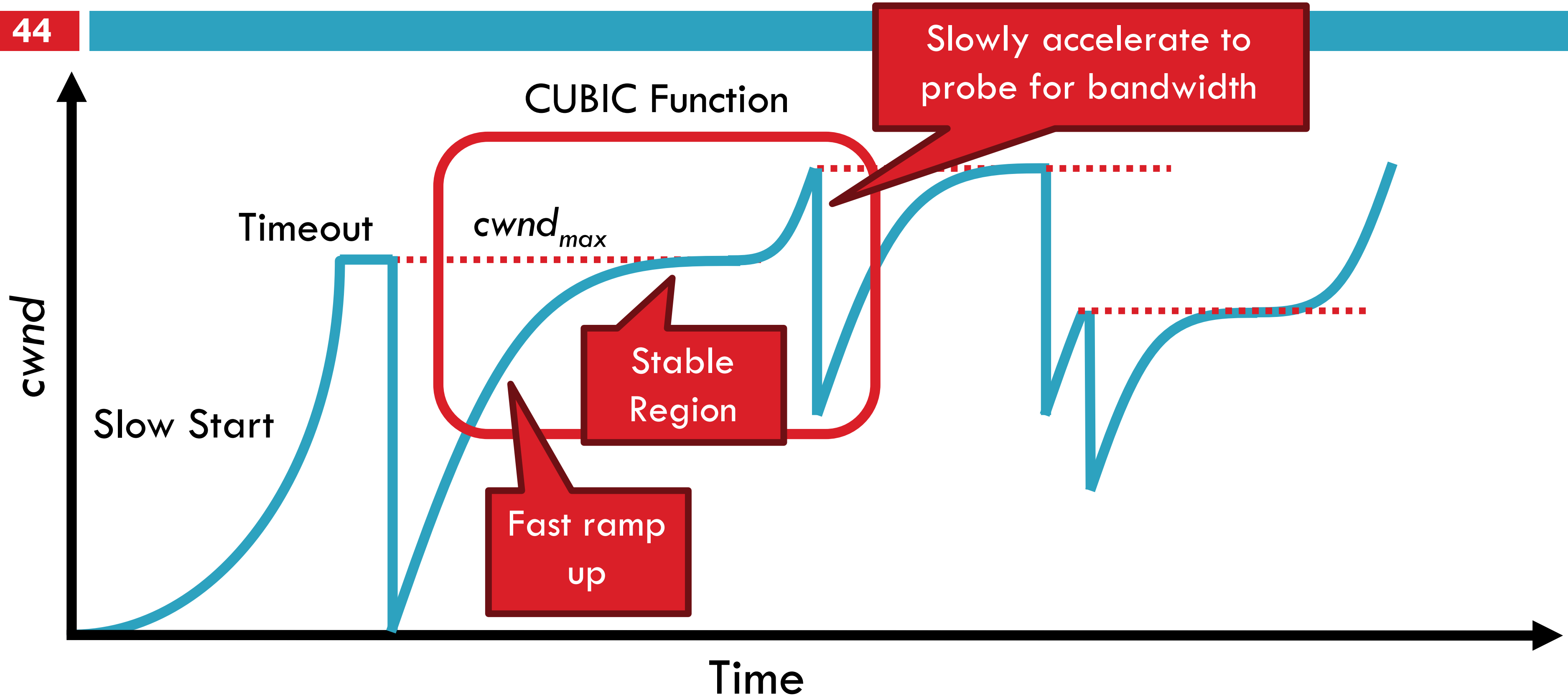
$$\text{C is a scaling constant, and K} = \sqrt[3]{\frac{W_{max}\beta}{C}}$$

  - B → a constant fraction for multiplicative increase
  - T → time since last packet drop
  - W_max → cwnd when last packet dropped

TCP CUBIC Implementation



TCP CUBIC Example



- Less wasted bandwidth due to fast ramp up
- Stable region and slow acceleration help maintain fairness
  - Fast ramp up is more aggressive than additive increase
  - To be fair to Tahoe/Reno, CUBIC needs to be less aggressive

Outline

- UDP
- TCP
- Congestion Control
- Evolution of TCP
- Problems with TCP

Issues with TCP

- The vast majority of Internet traffic is TCP
- However, many issues with the protocol
  - Poor performance with small flows
  - Really poor performance on wireless networks
  - Susceptibility to denial of service

Small Flows

- Problem: TCP is biased against short flows
  - 1 RTT wasted for connection setup (SYN, SYN/ACK)
  - *cwnd* always starts at 1
- Vast majority of Internet traffic is short flows
  - Mostly HTTP transfers, <100KB
  - Most TCP flows never leave slow start!
- Proposed solutions (driven by Google):
  - Increase initial *cwnd* to 10
  - TCP Fast Open: use cryptographic hashes to identify receivers, eliminate the need for three-way handshake

Wireless Networks

- Problem: Tahoe and Reno assume loss = congestion
  - True on the WAN, bit errors are very rare
  - False on wireless, interference is very common
- TCP throughput ~ 1/sqrt(drop rate)
  - Even a few interference drops can kill performance
- Possible solutions:
  - Break layering, push data link info up to TCP
  - Use delay-based congestion detection (TCP Vegas)
  - Explicit congestion notification (ECN)

Denial of Service

- Problem: TCP connections require state
  - Initial SYN allocates resources on the server
  - State must persist for several minutes (RTO)
- SYN flood: send enough SYNs to a server to allocate all memory/meltdown the kernel
- Solution: SYN cookies
  - Idea: don't store initial state on the server
  - Securely insert state into the SYN/ACK packet (sequence number field)
  - Client will reflect the state back to the server