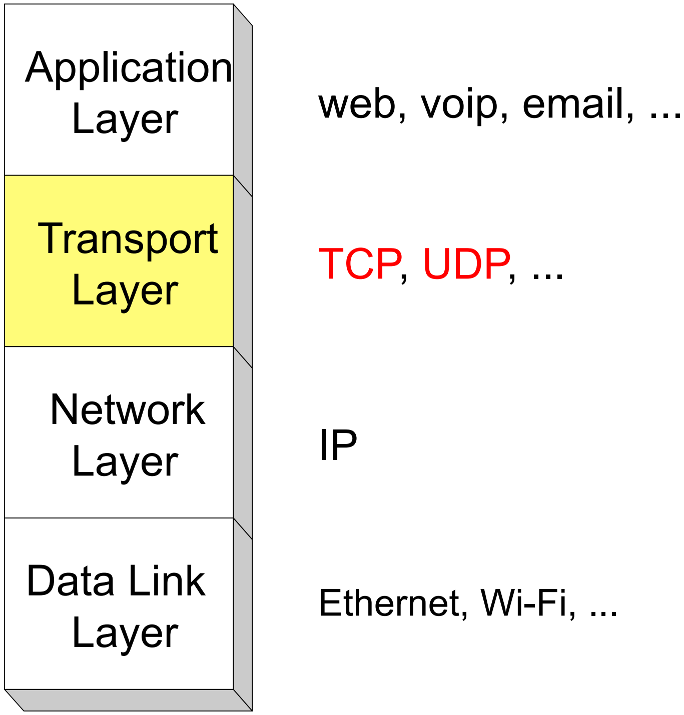
COMP/ELEC 429/556 Introduction to Computer Networks

The TCP Protocol

Some slides used with permissions from Edward W. Knightly, T. S. Eugene Ng, Ion Stoica, Hui Zhang

Transport Layer

- Purpose 1: Demultiplexing of data streams to different application processes
- Purpose 2: Provide value-added services that many applications want
  - Recall network layer in Internet provides a "Best-effort" service only, transport layer can add value to that
    - Application may want reliability, etc
  - No need to reinvent the wheel each time you write a new application



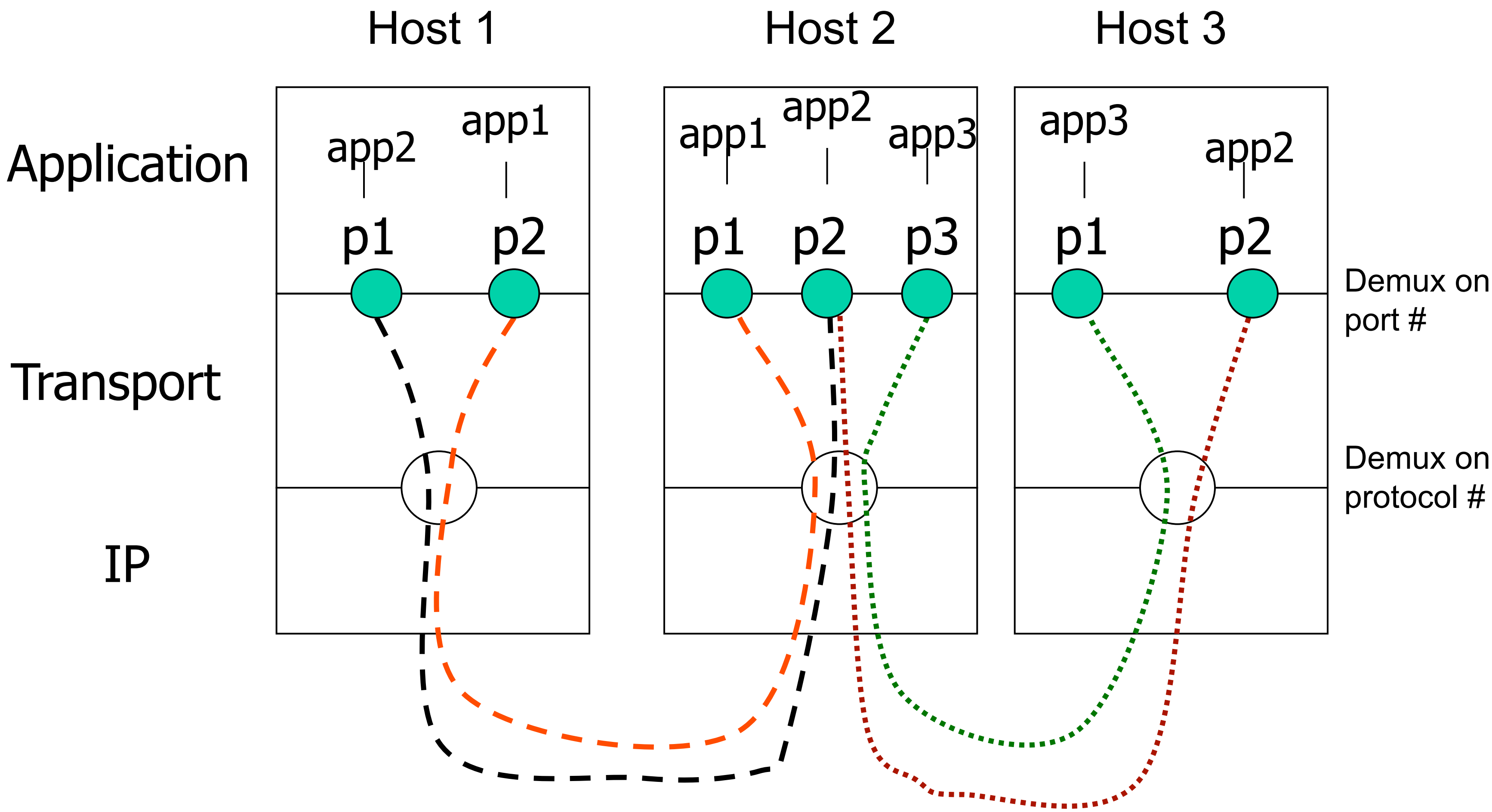
A very simple transport protocol: User Datagram Protocol (UDP)

- Connectionless datagram
  - Socket: SOCK_DGRAM
- Port number used for demultiplexing
  - port numbers = connection/application endpoint
- Adds end-to-end error checking through optional checksum
  - some protection against data corruption errors between source and destination (links, switches/routers, bus)
  - does not protect against packet loss, duplication or reordering

0 16 32

Source Port	**Dest. Port**
Length	**Checksum**

Using Transport Layer Port Number to Demultiplex traffic

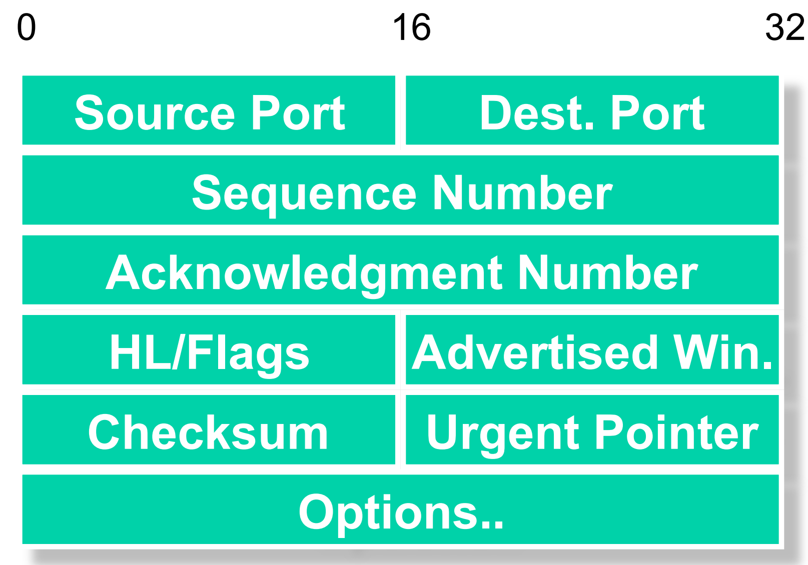


Usages of UDP

- Custom protocols/applications can be implemented on top of UDP
  - use the port addressing provided by UDP
  - implement specialized reliability, flow control, ordering, congestion control as the app sees fit
- Examples:
  - remote procedure call
  - multimedia streaming (real time protocol)
  - cluster computing communication libraries

Transmission Control Protocol (TCP)

- Reliable bidirectional in-order byte stream
  - Socket: SOCK_STREAM
- Connections established & torn down
- Multiplexing/ demultiplexing
  - Ports at both ends
- Error control
  - Users see correct, ordered byte sequences
- End-to-end flow control
  - Avoid overwhelming receiver at each end
- Congestion control
  - Avoid creating traffic jams within network



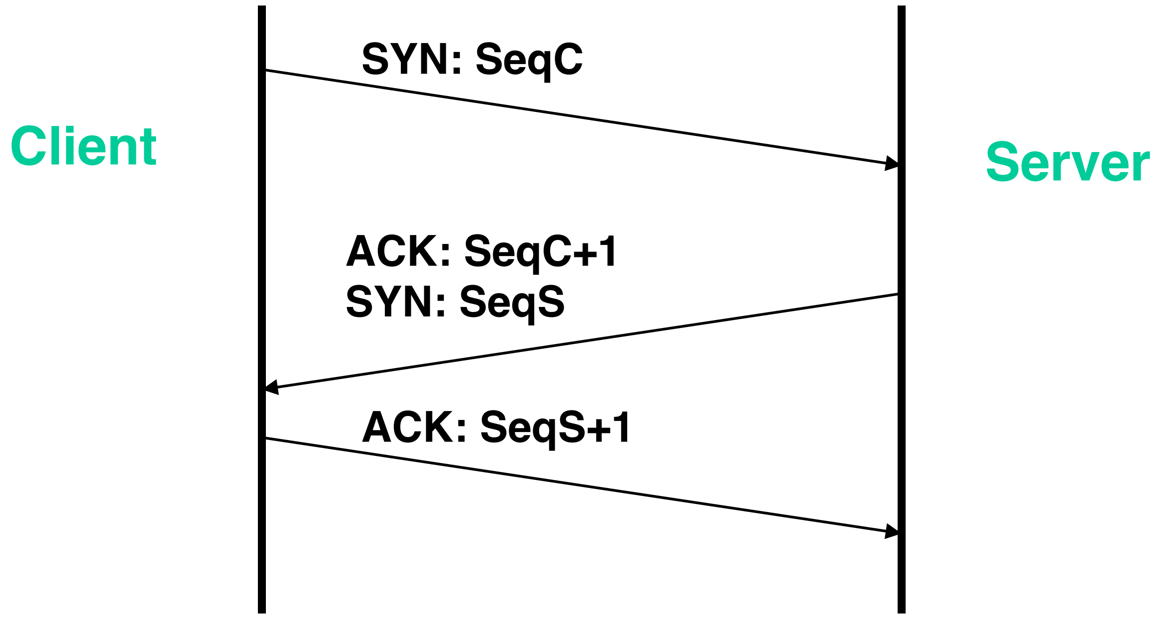
Connection Setup

- Why need connection setup?
- Mainly to agree on starting sequence numbers
  - Starting sequence number is randomly chosen
  - Reason: to reduce the chance that sequence numbers of old and new connections from overlapping

Important TCP Flags

- SYN: Synchronize
  - Used when setting up connection
- FIN: Finish
  - Used when tearing down connection
- ACK
  - Acknowledging received data

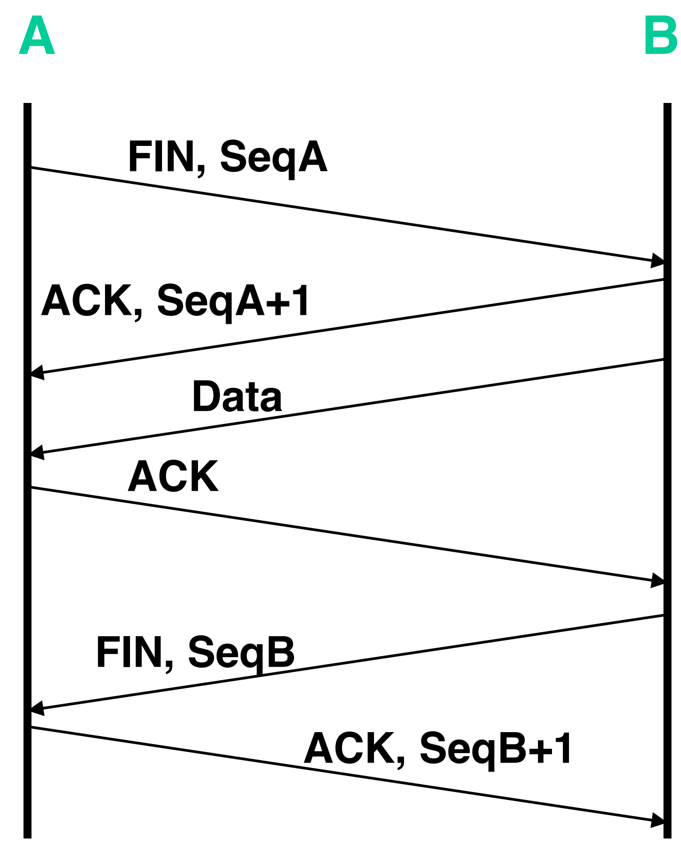
Establishing Connection



- Three-Way Handshake
  - Each side notifies other of starting sequence number it will use for sending
  - Each side acknowledges other's sequence number
    - SYN-ACK: Acknowledge sequence number + 1
  - Can combine second SYN with first ACK

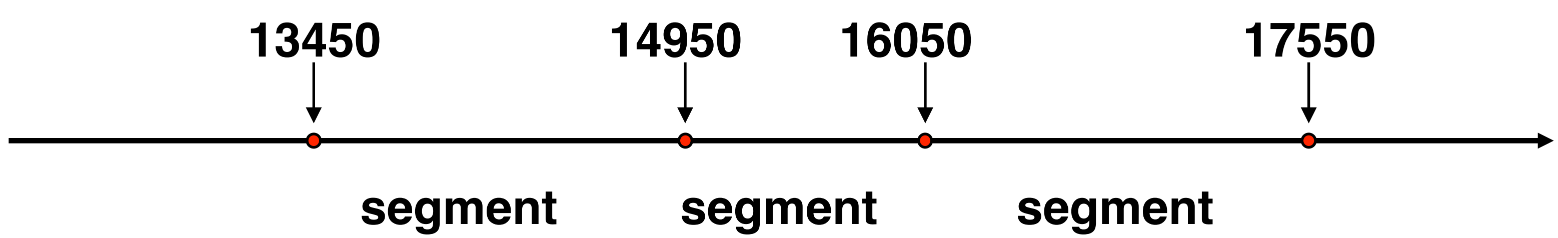
Tearing Down Connection

- Either Side Can Initiate Tear Down
  - Send FIN signal
  - "I'm not going to send any more data"
- Other Side Can Continue Sending Data
  - Half open connection
  - Must continue to acknowledge
- Acknowledging FIN
  - Acknowledge last sequence number + 1

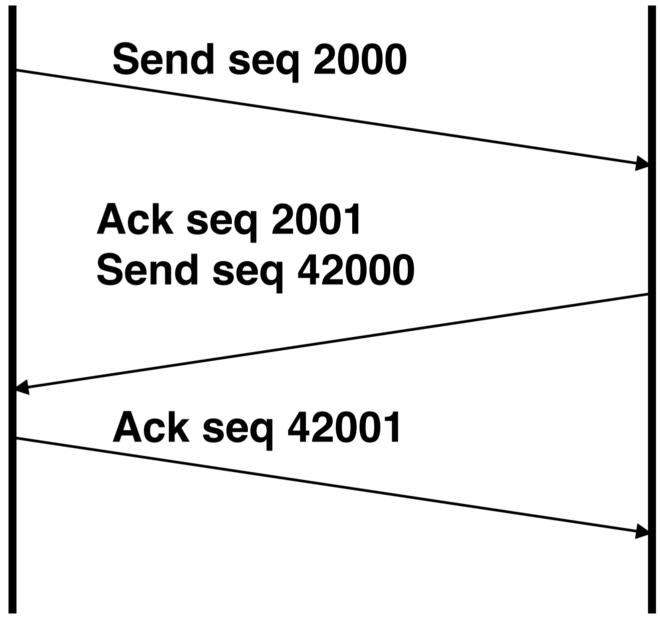


Sequence Number Space

- Each byte in byte stream is numbered
  - 32 bit value
  - Wraps around
  - Initial values selected at start up time
- TCP breaks up the byte stream into segments
  - Each segment transmitted by a packet
  - Limited by the Maximum Segment Size
  - Set to prevent packet fragmentation
- Each segment has a sequence number
  - Indicates where it fits in the byte stream



Bidirectional Communication



- Each Side of Connection can Send *and* Receive
- What this Means
  - Maintain different sequence numbers for each direction
  - A single packet can contain new data for one direction, plus acknowledgement for other, may also contain only data or only acknowledgement
- When there is a loss, e.g. seq # 2000, 2002, 2003, 2004 are received, TCP receiver acks 2001, 2001, 2001, 2001
  - Called duplicate ACKs
  - There are TCP variants that don't do this – beyond our scope

Sequence Numbers

- 32 Bits, Unsigned

- Why So Big?
  - For sliding window, must have
    |Sequence Space| > 2* |Sending Window|
    - Sending window size of basic TCP is at most 2^16 bytes
    - 2^32 > 2 * 2^16; no problem
  - Also, want to guard against stray packets
    - With IP, assume packets have maximum segment lifetime (MSL) of 120s
      - i.e. can linger in network for upto 120s
    - Sequence number would wrap around in this time at 286Mbps

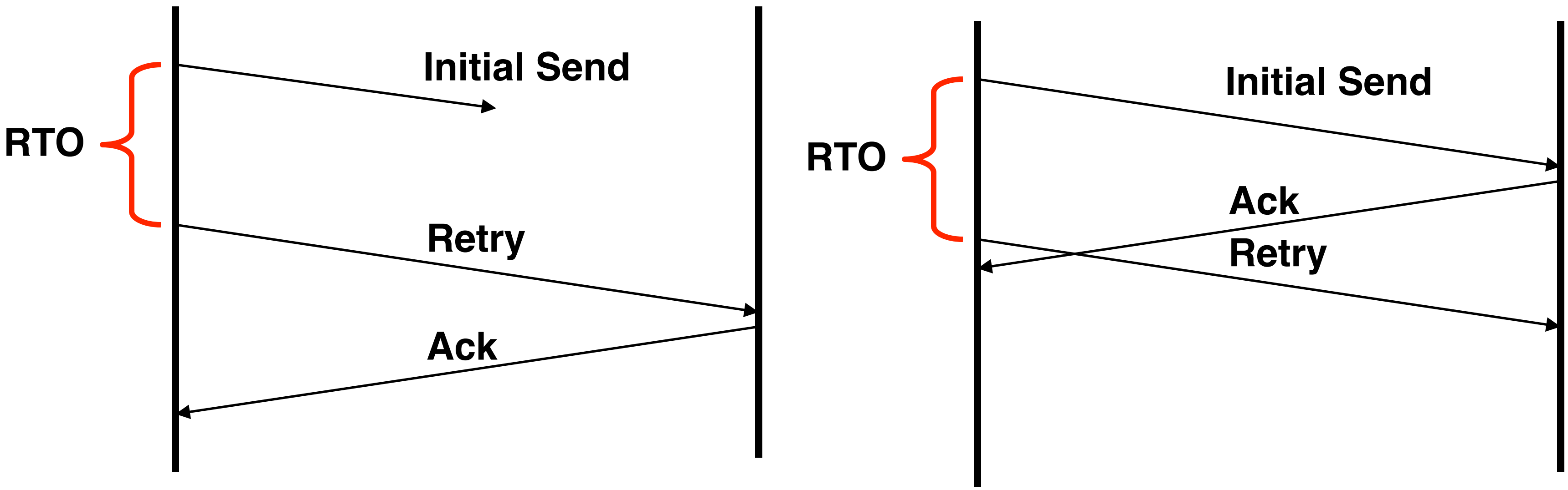
Error Control

- Checksum provides some end-to-end error protection
- Sequence numbers detect packet sequencing problems:
  - Duplicate: ignore
  - Reordered: reorder or drop
  - Lost: retransmit
- Lost segments retransmitted by sender
  - Use time out to detect lack of acknowledgment
  - Need estimate of the roundtrip time to set timeout
- Retransmission requires that sender keep copy of the data
  - Copy is discarded when ack is received

TCP Must Operate Over Any Internet Path

- Retransmission time-out should be set based on round-trip delay
- But round-trip delay different for each path!
- Must estimate RTT dynamically

Setting Retransmission Timeout (RTO)



Detect dropped packet

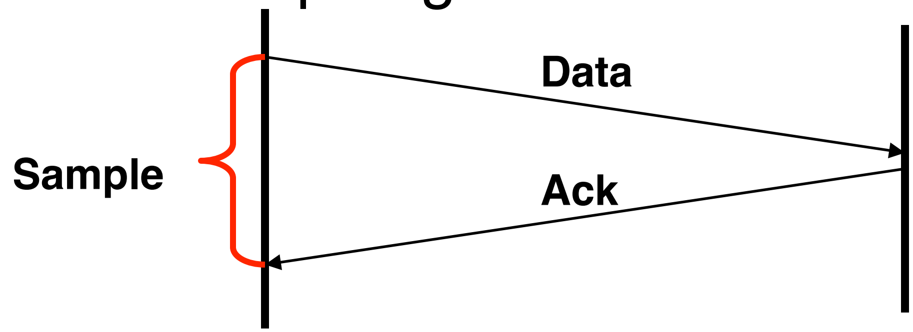
RTO too short

- Time between sending & resending segment

- Challenge
  - Too long: Add latency to communication when packets dropped
  - Too short: Send too many duplicate packets
  - General principle: Must be > 1 Round Trip Time (RTT)

Round-trip Time Estimation

- Every Data/Ack pair gives new RTT estimate

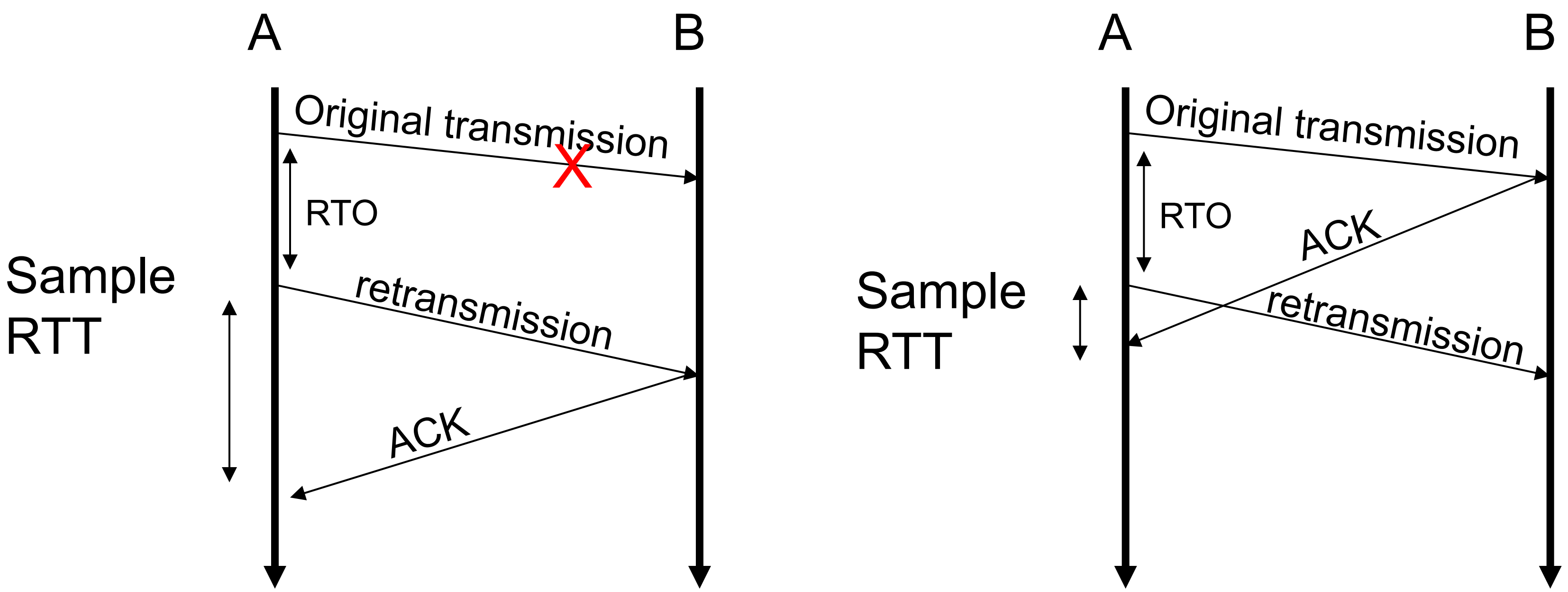


- Can Get Lots of Short-Term Fluctuations
  - How to address this problem?

Exponential Smoothing Technique

- Round trip times estimated as a moving average:
  - Smoothed RTT = α (Smoothed RTT) + (1 - α) (new RTT sample)
  - Recommended value for α: 0.8 - 0.9
- Retransmission timeout (RTO) is a function of
  - Smoothed RTT (SRTT)
  - RTT variation (RTTVAR)
- RTO = SRTT + 4 * RTTVAR
  - Details in RFC 6298

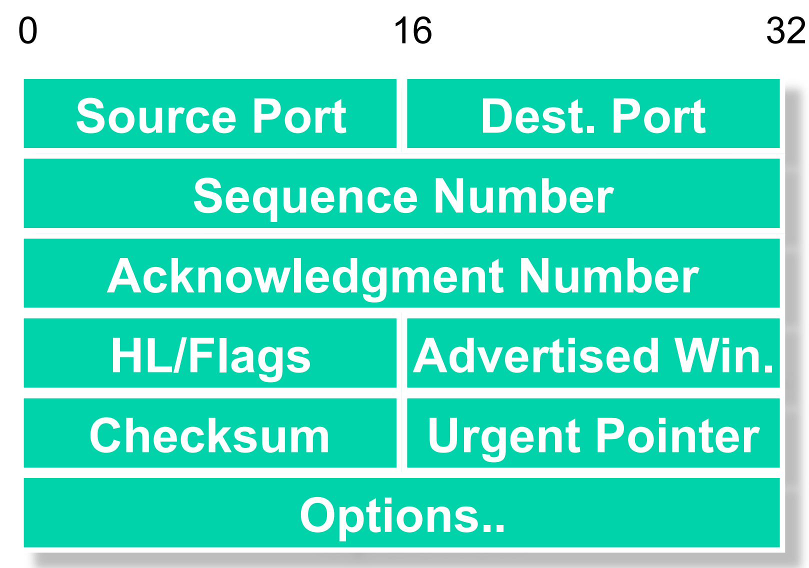
RTT Sample Ambiguity



- Ignore sample for segment that has been retransmitted

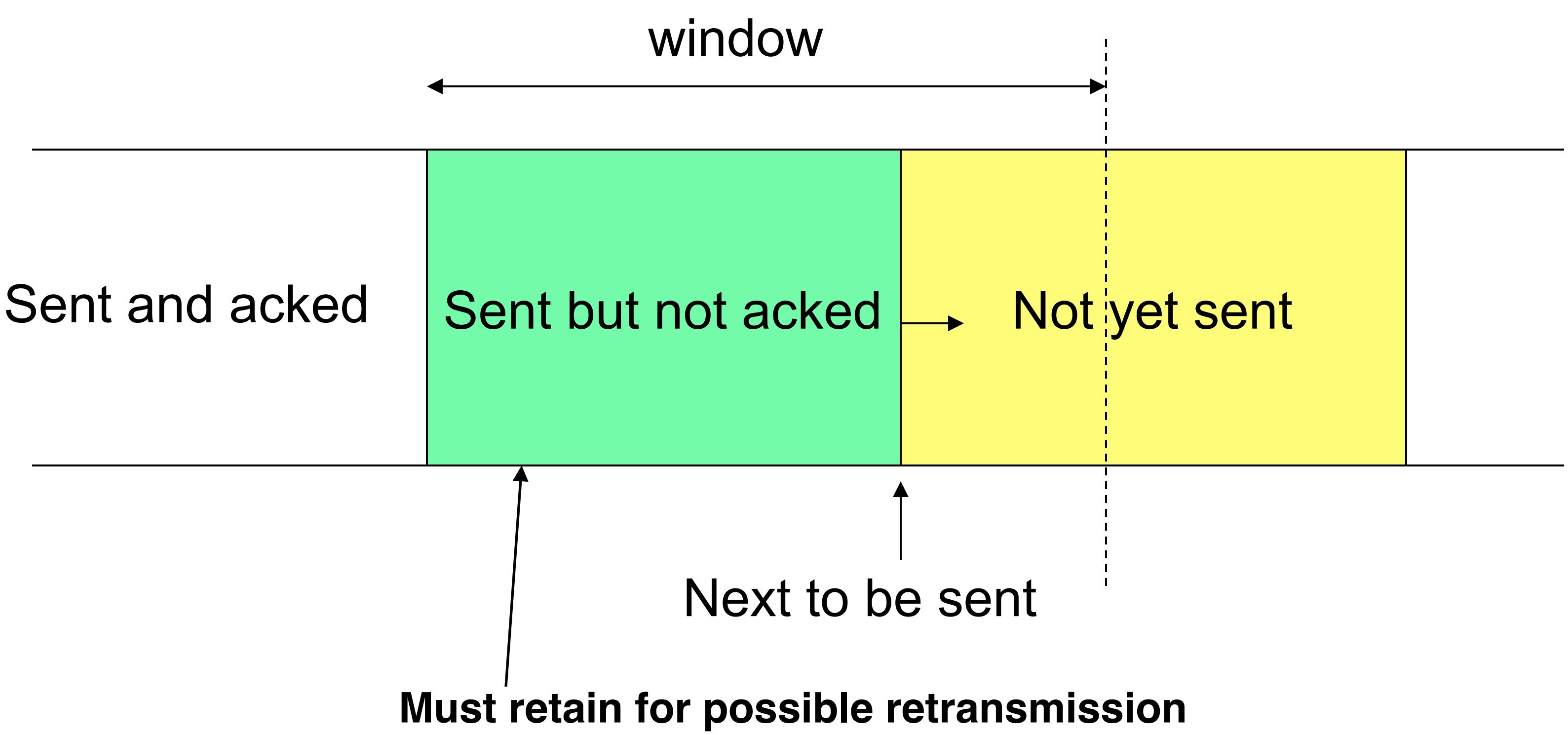
TCP Speed Control

- Sliding window protocol
  - For window size *n*, can send up to *n* bytes without receiving new acknowledgement
  - Speed proportional to n/RTT
  - When the data are acknowledged then the window slides forward

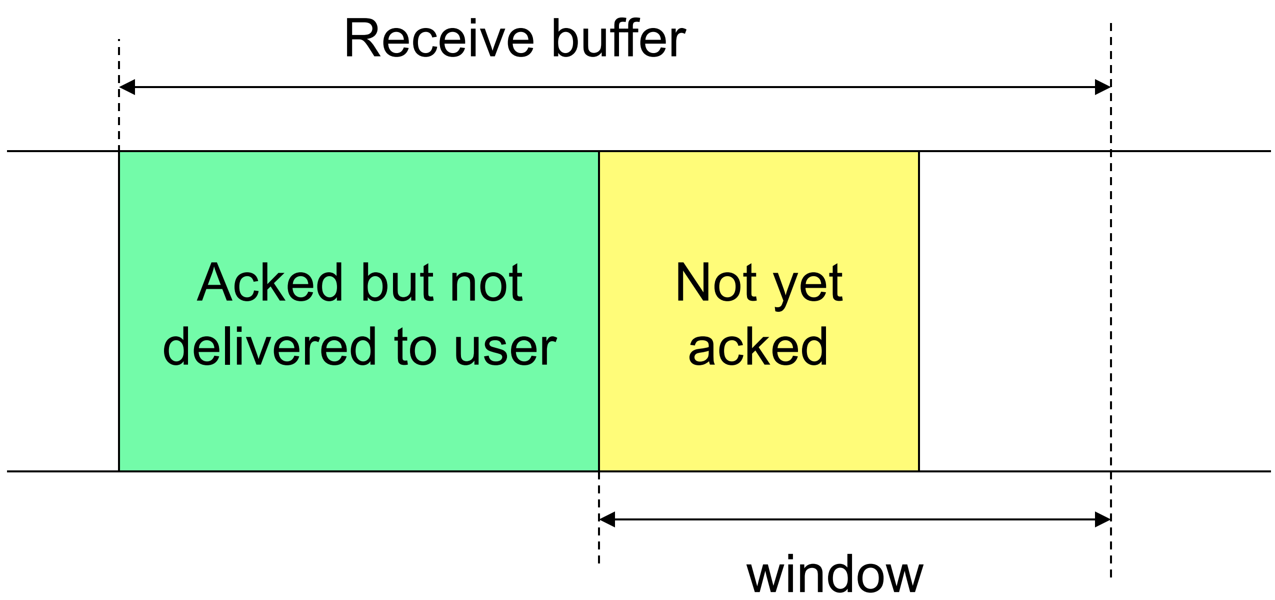


- Send window size set to **minimum (advertised window, congestion window)**

Window Flow Control: Send Side



Window Flow Control: Receive Side



- TCP receiver can delete acknowledged data only after the data has been delivered to the application
- So, depending on how fast the application is reading the data, the receiver's window size may change!!!

Solution

- Receiver tells sender the current advertised window size in every packet it transmits to the sender
- Sender uses this current advertised window size as an upper bound
  - send window size = minimum (advertised window, congestion window)
- Advertised window size is continuously changing
- Can go to zero!
  - Sender not allowed to send anything!

Setting Congestion Window

Send window size = minimum (advertised window, congestion window)

Phases of TCP congestion control

1. Slow start (getting to equilibrium)
   - Want to find this very very fast and not waste time
2. Congestion Avoidance
   - Additive increase - gradually probing for additional bandwidth
   - Multiplicative decrease – decreasing congestion window upon loss/timeout

Variables Used in Implementation

- **Congestion Window (cwnd)**
  Initial value is 1 MSS (=maximum segment size) counted as bytes

- **Actual sender window size used by TCP = minimum (advertised win, cwnd)**

- **Slow-start threshold Value (ss_thresh)**
  Initial value is the advertised window size

- **slow start** (cwnd < ssthresh)
- **congestion avoidance** (cwnd >= ssthresh)

TCP: Slow Start

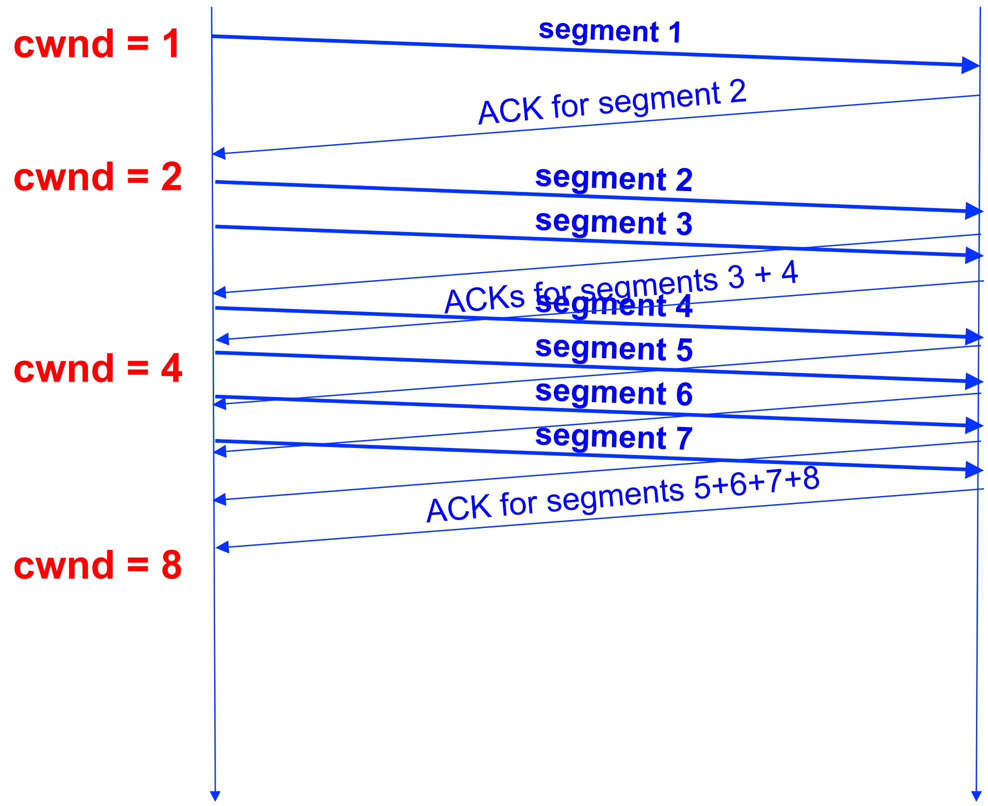
- Goal: discover roughly the proper sending rate quickly

- Whenever starting traffic on a new connection, or whenever increasing traffic after congestion was experienced:
    - Intialize *cwnd* = 1 MSS (max segment size)
    - Each time a segment is acknowledged, increment *cwnd* by one MSS (*cwnd*++).

- Continue until
    – Reach ss_thresh or
    – Packet loss

Slow Start Illustration

- The congestion window size grows very rapidly
- Observe:
  - Each ACK generates two packets
  - slow start increases rate exponentially fast (doubled every RTT)!



Congestion Avoidance

- Slow Start figures out roughly the rate at which the network starts getting congested
- Congestion Avoidance continues to react to network condition
  – Probes for more bandwidth, increase cwnd if more bandwidth available
  – If congestion detected, aggressive cut back cwnd

Congestion Avoidance: Additive Increase

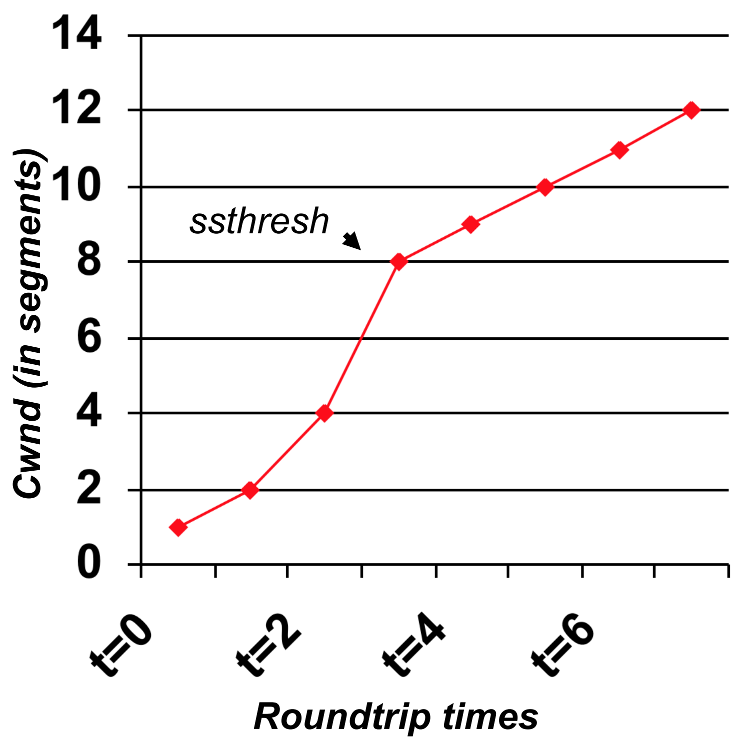
- Slowly increase cwnd to probe for additional available bandwidth

- **If** *cwnd >= ss_thresh* **then**
  each time a segment is newly acknowledged
  cwnd += 1/cwnd

- *cwnd* is increased by one MSS only if all segments in the window have been acknowledged
  - Increases by 1 per RTT

Example of Slow Start + Congestion Avoidance

Assume that *ss_thresh = 8*



Detecting Congestion via Timeout

- If there is a packet loss, the ACK for that packet will not be received
- The packet will eventually timeout
  - No ack is seen as a sign of congestion

Congestion Avoidance: Multiplicative Decrease

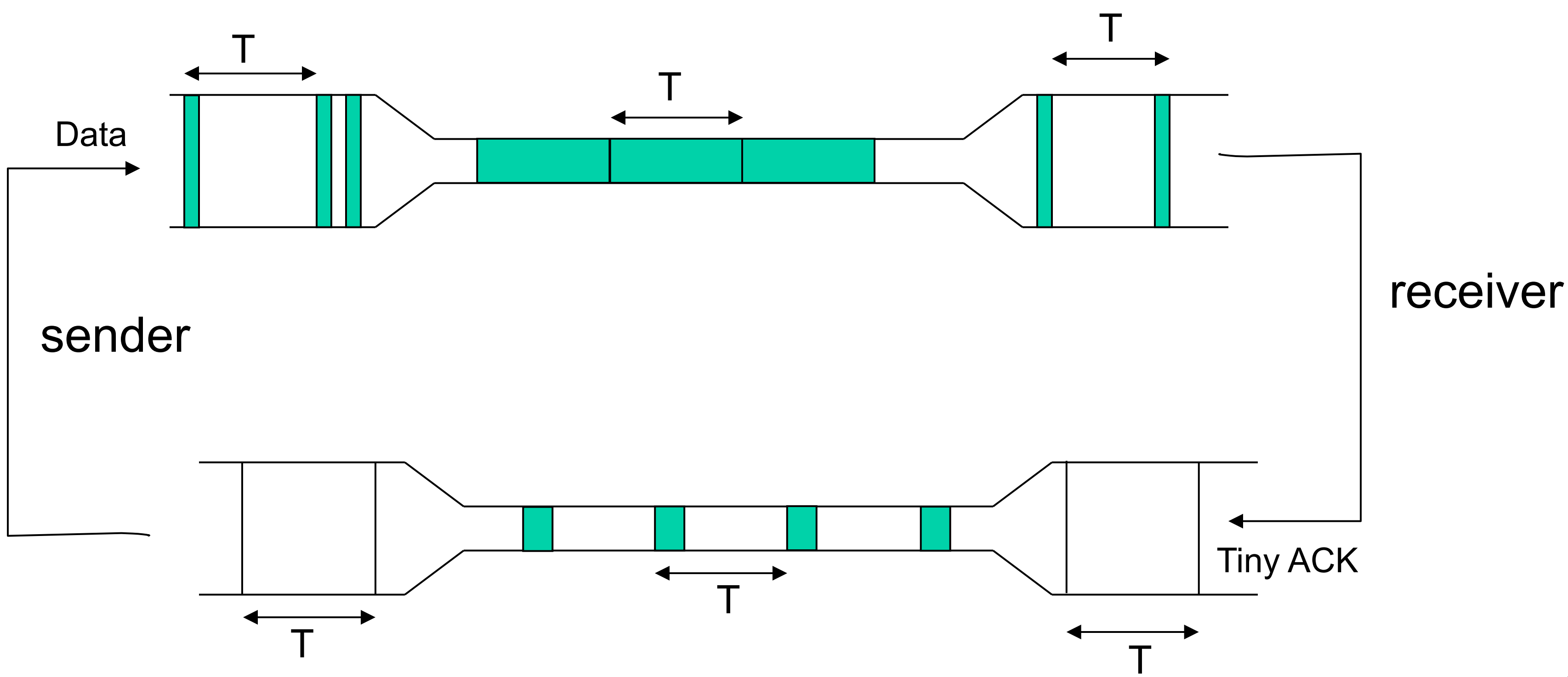
- Each time when timeout occurs
  - ss_thresh is set to half the current size of the congestion window:

    ss_thresh = cwnd / 2
  - cwnd is reset to one:

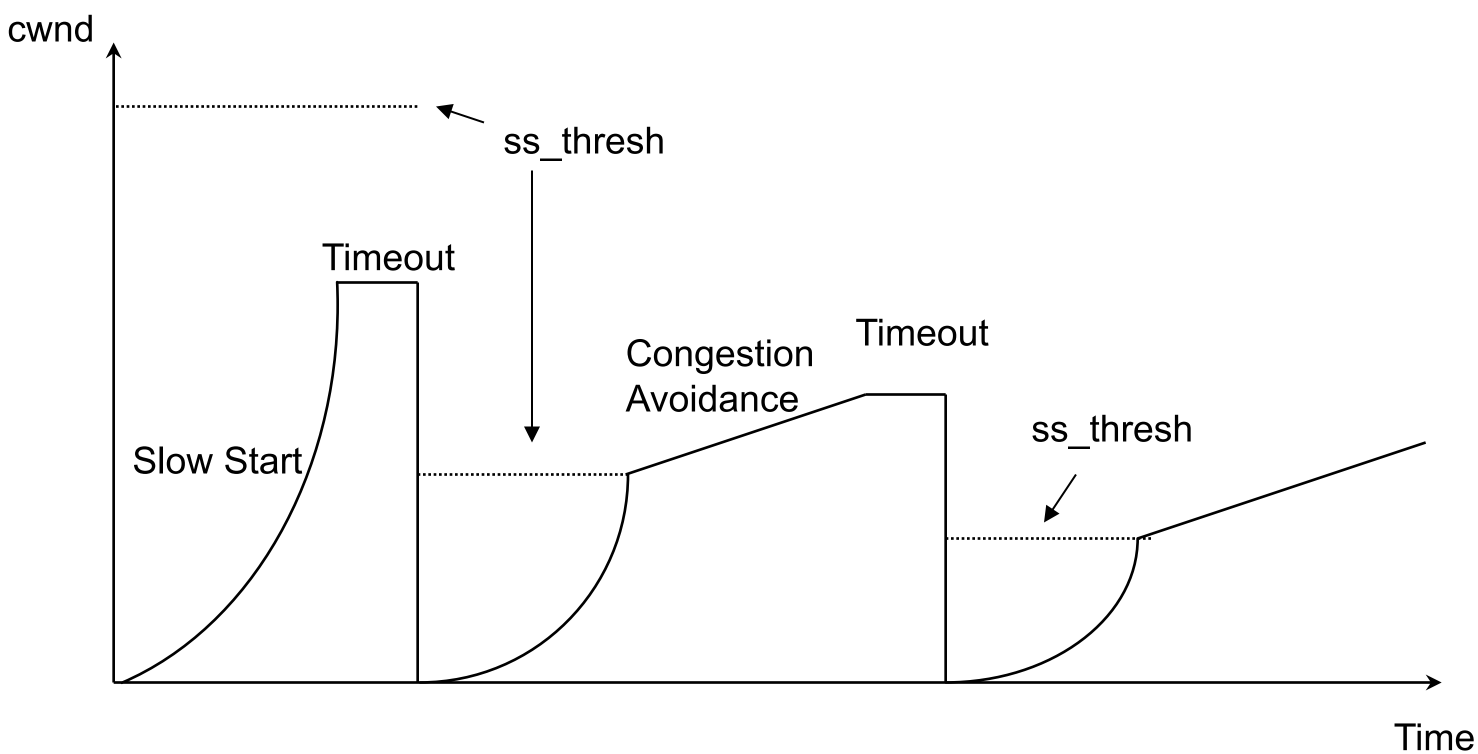
    cwnd = 1
  - and slow-start is entered

Packet Pacing

- ACKs “self-pace” the data to avoid a burst of packets to be sent
- Observe: received ACK spacing ≅ bottleneck bandwidth



TCP (Tahoe variant) Illustration



Many Variants of TCP

- Common variants of TCP

  - **TCP Tahoe** - the basic algorithm (discussed previously)
  - **TCP Reno** - Tahoe + fast retransmit & fast recovery
    - Many end hosts today implement TCP Reno

- and many more:
  - TCP Vegas (use timing of ACKs to avoid loss)
  - TCP SACK (selective ACK)

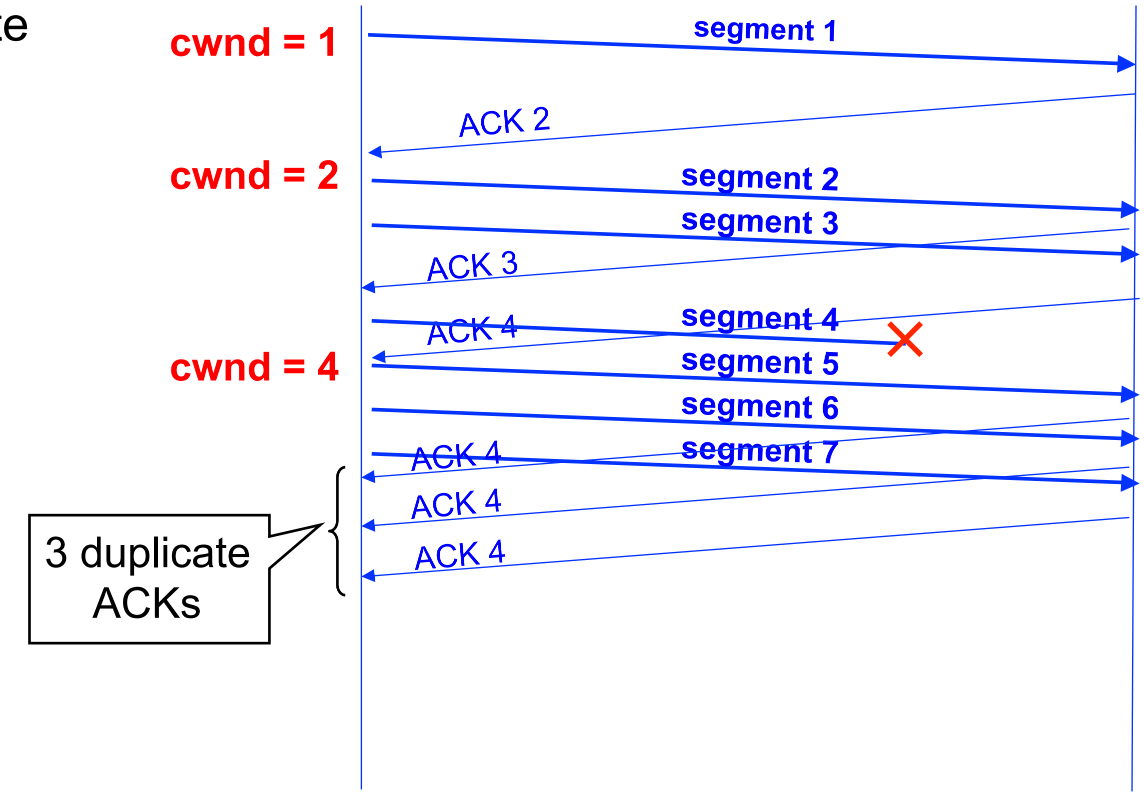
TCP Reno

- Problem with Tahoe: If a segment is lost, there is a long wait until timeout
- Reno adds a **fast retransmit** and **fast recovery mechanisms**

- Upon receiving 3 duplicate ACKs, retransmit the presumed lost segment ("fast retransmit")
- But do not enter slow-start. Instead enter congestion avoidance ("fast recovery")

Fast Retransmit

- Resend a segment after 3 duplicate ACKs
  - remember a duplicate ACK means that an out-of sequence segment was received
  - ACK-n means packets 1, …, n-1 all received

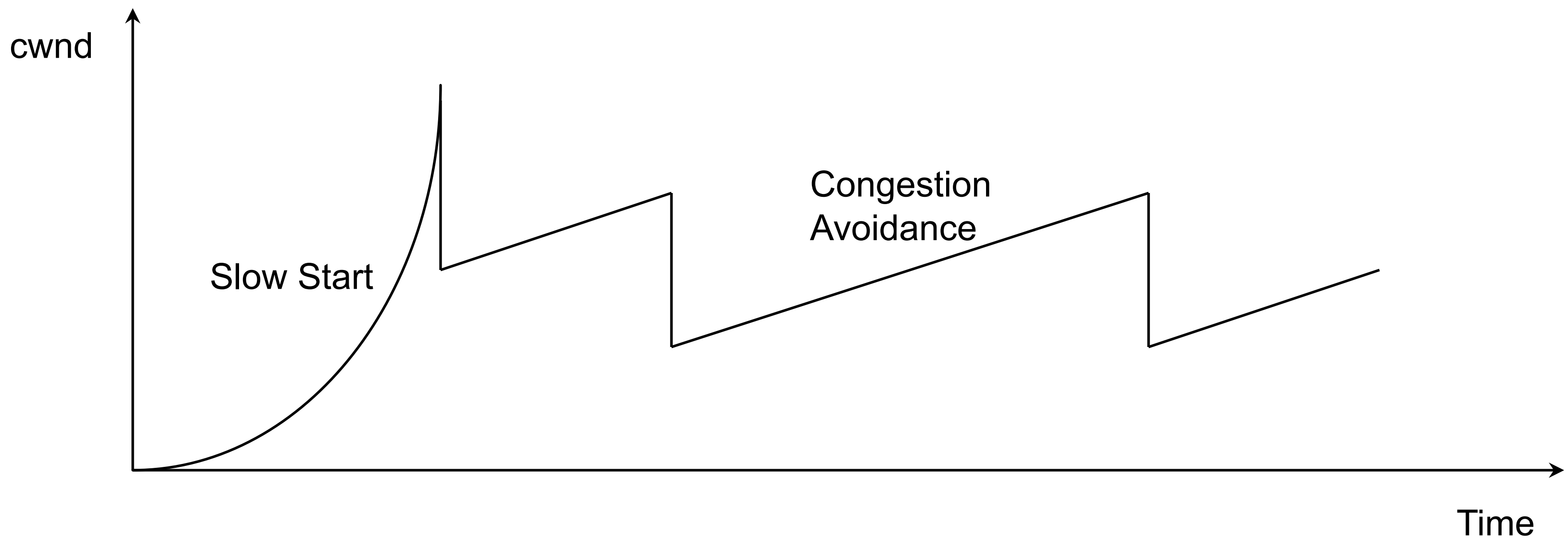


- Notes:
  - duplicate ACKs can be due to packet reordering!
  - if window is small, may not get 3 duplicate ACKs!

Fast Recovery

- After a fast-retransmit
  - cwnd = cwnd/2
  - ss_thresh = cwnd
  - i.e. starts congestion avoidance at new cwnd
- After a timeout
  - Same as TCP Tahoe
  - *ss*_thresh = *cwnd/2*
  - cwnd = 1
  - Do slow start

Fast Retransmit and Fast Recovery



- Retransmit after 3 duplicate ACKs
  - prevent expensive timeouts
- Slow start only once per session (if no timeouts)
- In steady state, *cwnd* oscillates around the ideal window size.

TCP Reno Summary

- Slow-Start if cwnd < ss_thresh
  - cwnd++ upon every new ACK (exponential growth)
  - Timeout: ss_thresh = cwnd/2 and cwnd = 1
- Congestion avoidance if cwnd >= ss_thresh
  - Additive Increase Multiplicative Decrease (AIMD)
  - ACK: cwnd = cwnd + 1/cwnd
  - Timeout: ss_thresh = cwnd/2 and cwnd = 1
- Fast Retransmit & Recovery
  - 3 duplicate ACKs (interpret as packet loss)
  - Retransmit lost packet
  - cwnd=cwnd/2, ss_thresh = cwnd

TCP Reno Saw Tooth Behavior