CSCE 463/612 Networks and Distributed Processing Fall 2020

Transport Layer IV

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Chapter 3: Roadmap

3.1 Transport-layer services

3.2 Multiplexing and demultiplexing

3.3 Connectionless transport: UDP

3.4 Principles of reliable data transfer

3.5 Connection-oriented transport: TCP

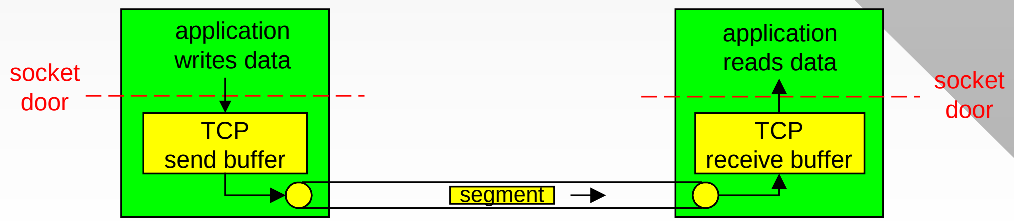
- Segment structure
- Reliable data transfer
- Flow control
- Connection management

3.6 Principles of congestion control

3.7 TCP congestion control

TCP: Overview [RFCs: 793, 1122, 1323, 2001, 2018, 2581, 3390, 5681]

- Point-to-point (unicast):
  - One sender, one receiver
- Reliable, in-order *byte stream:*
  - Packet boundaries are not visible to the application
- Pipelined:
  - TCP congestion and flow control set window size
- Send & receive buffers



- Full duplex data:
  - Bi-directional data flow in same connection
- MSS: maximum segment size (excluding headers)
- Connection-oriented:
  - Handshaking (exchange of control msgs) initializes sender/receiver state before sending data
- Flow controlled:
  - Sender will not overwhelm receiver

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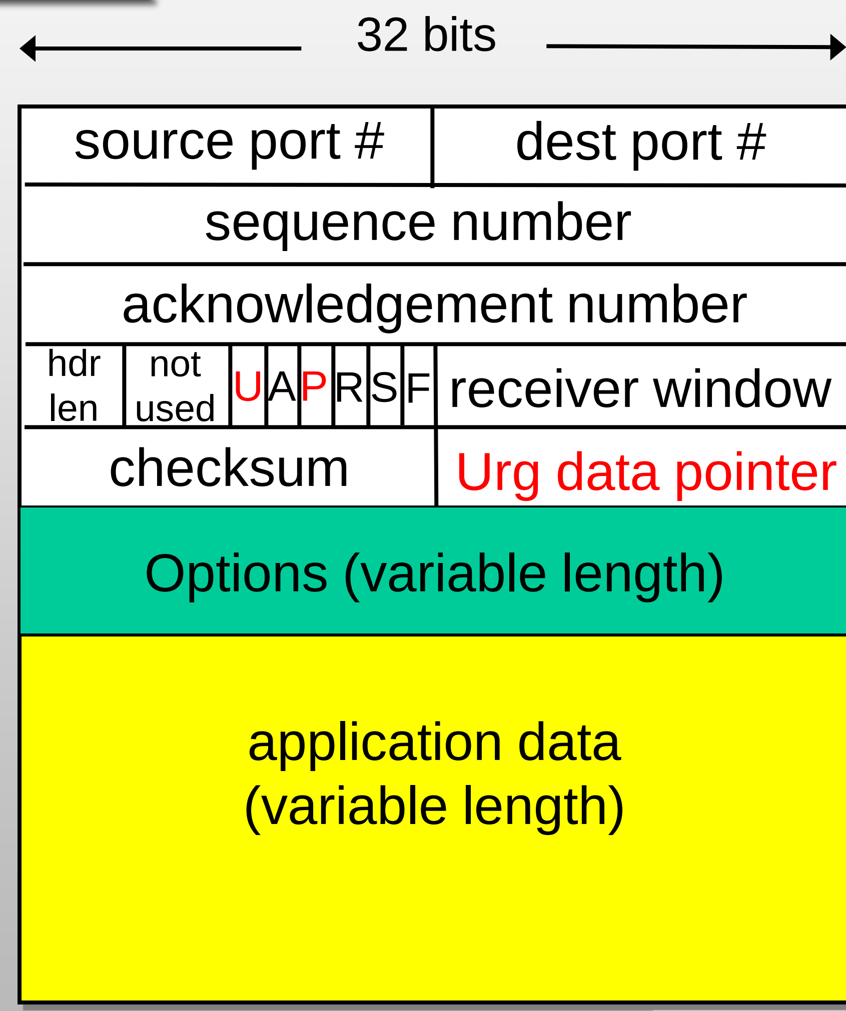
TCP Segment Structure

- Sequence/ACK numbers
  - Count bytes, not segments
  - ACKs piggybacked on data packets
- Flags (U-A-P-R-S-F)
  - Urgent data (not used)
  - ACK field is valid
  - PUSH (not used)
  - RST (reset connection)
  - SYN (connection request)
  - FIN (connection close)
- Hdr length in DWORDs (4-bit field)
  - Normally 20 bytes, but longer if options are present



TCP Seq. #'S and ACKs

Seq. #'s:

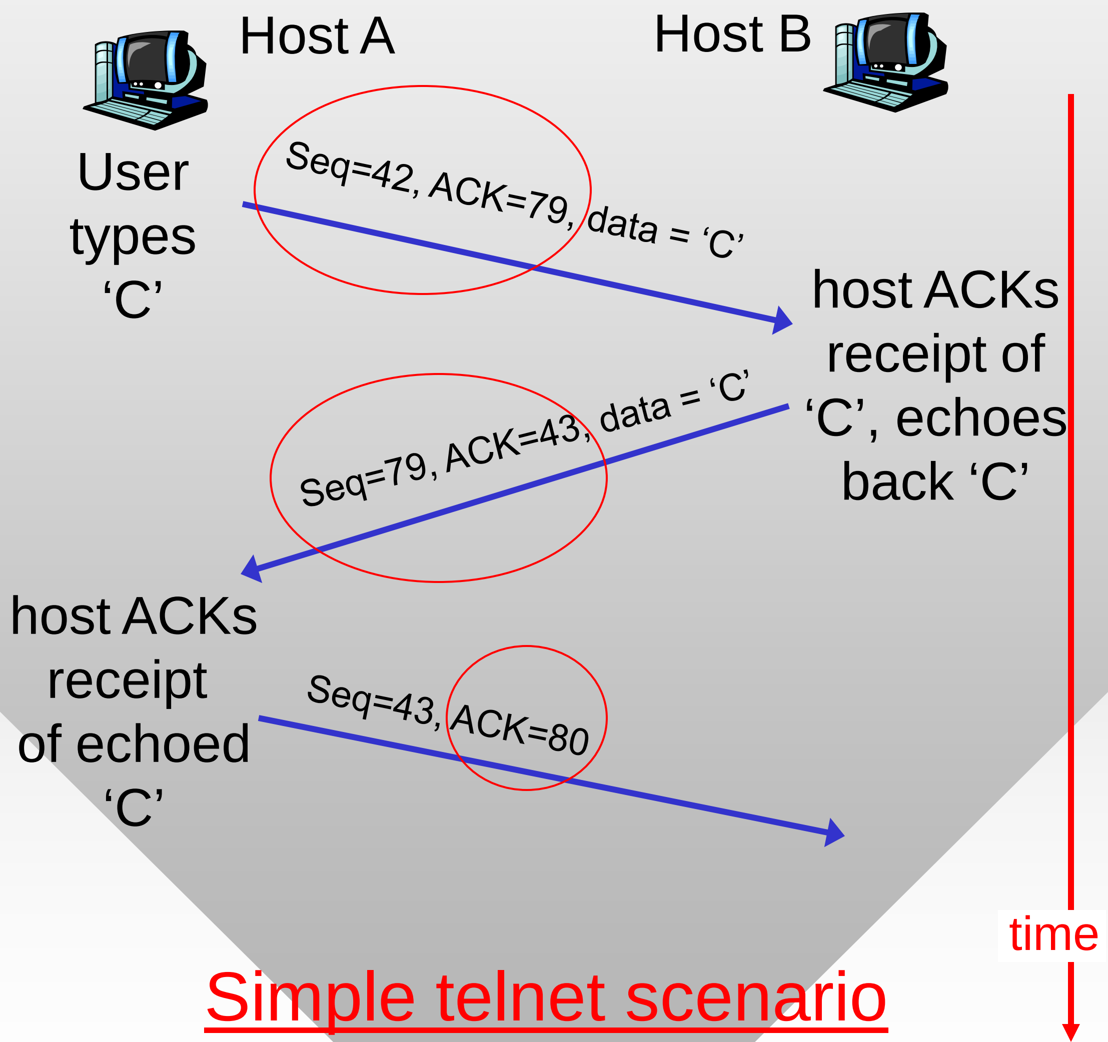
- Sequence number of the first byte in segment's data

ACKs:

- Seq # of next byte expected from sender
- Cumulative ACK

Q: how receiver handles out-of-order segments?

A: TCP spec doesn't say, up to implementor



Simple telnet scenario

TCP Round Trip Time and Timeout

Q: how to set TCP timeout value (RTO)?

- Want it slightly larger than the next RTT
  - But the RTT varies
- Too short: premature timeout
  - Unnecessary retransmissions
- Too long: slow reaction to segment loss
  - Protocol may stall, exhibit low performance

- Idea: dynamically measure RTT, average these samples, then add safety margin
- `SampleRTT`: measured time from segment transmission until ACK receipt
  - Ignore retransmissions, why?
- `SampleRTT` will vary, want estimated RTT “smoother”
  - Average several recent measurements, not just current **`SampleRTT`**

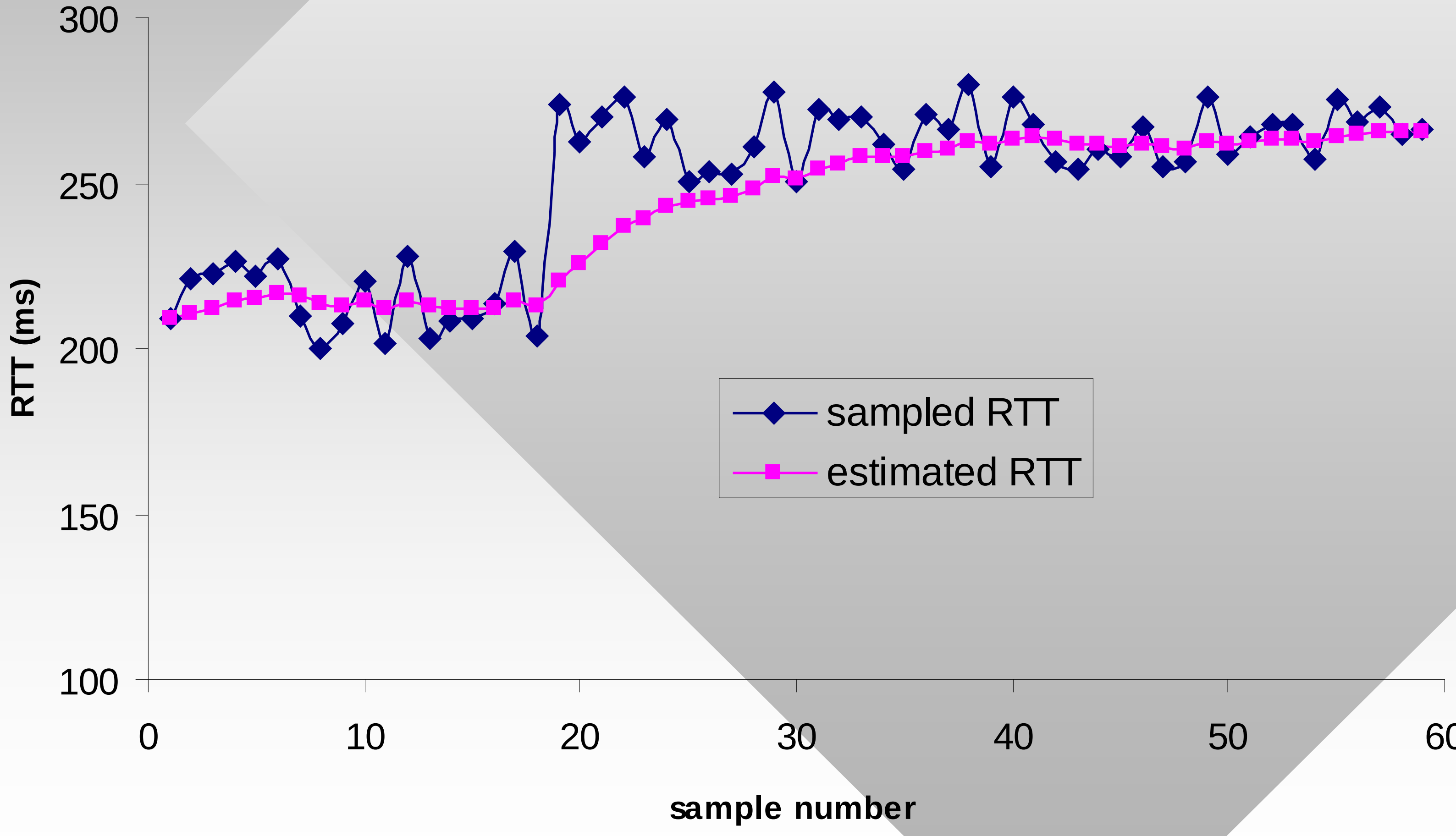
TCP Round Trip Time and Timeout

EstimatedRTT(n) = $(1$-$\alpha)$*EstimatedRTT$(n$-1$)$ + α*SampleRTT(n)

- Exponentially weighted moving average (EWMA)
  - Influence of past sample decreases exponentially fast
  - Typical value: $\alpha = 0.125 = 1/8$
- Task: derive a non-recursive formula for EstimatedRTT(n)
  - Assume EstimatedRTT(0) = SampleRTT(0)
  - Let $Y(n) =$ EstimatedRTT(n) and $y(n) =$ SampleRTT(n)

$$Y(n) = (1-\alpha)^n y(0) + \alpha \sum_{i=0}^{n-1} (1-\alpha)^i y(n-i)$$

Example RTT Estimation:



TCP Round Trip Time and Timeout

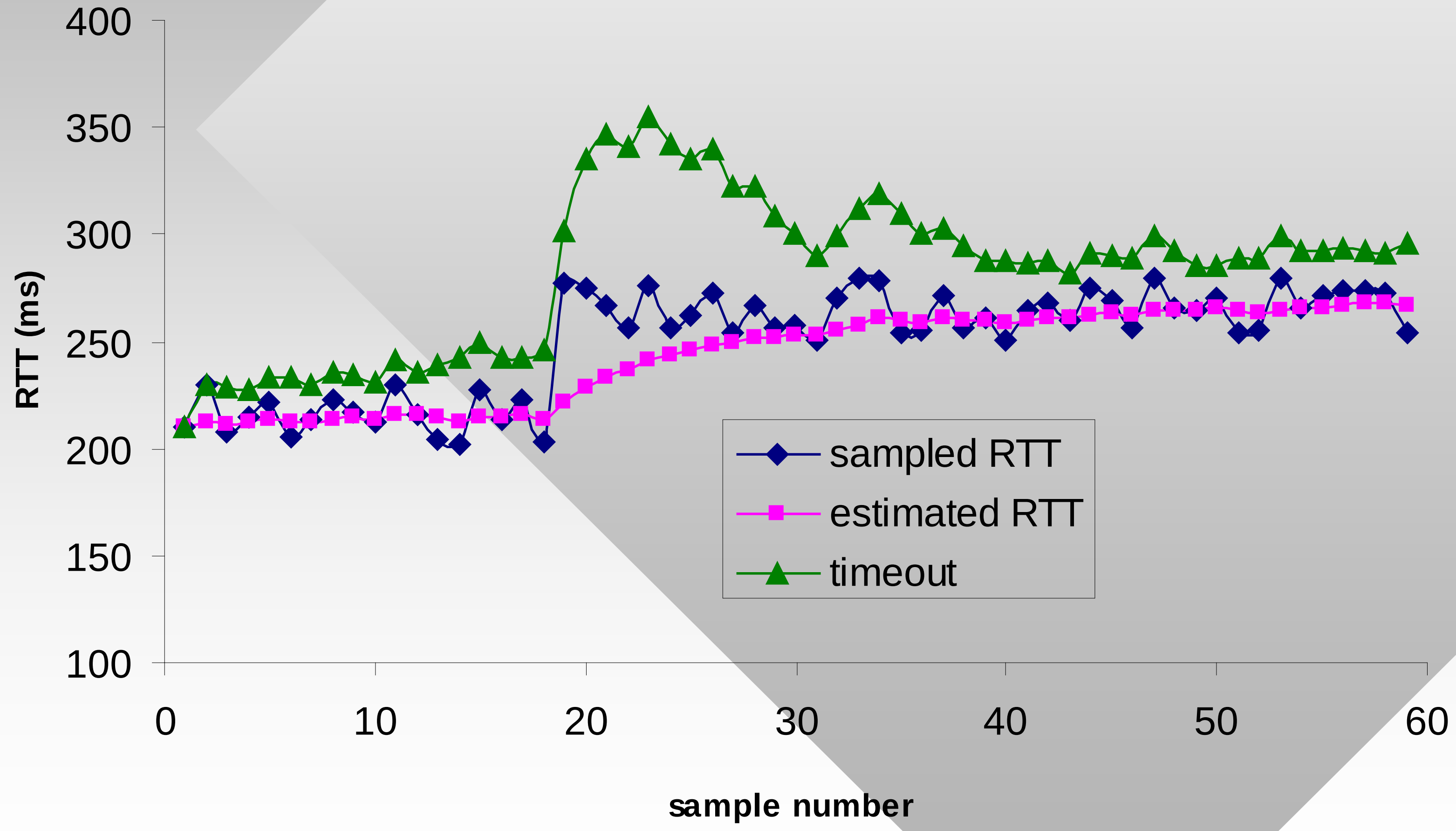
- Setting the timeout:
- `EstimatedRTT` plus a "safety margin"
  - Larger variation in `EstimatedRTT` → larger safety margin
- First estimate how much SampleRTT deviates from EstimatedRTT (typically, $\beta = 0.25$):

```
DevRTT(n) = (1-β)*DevRTT(n-1)
                + β *|SampleRTT(n)-EstimatedRTT(n)|
```

Then set retransmission timeout (RTO):

```
RTO(n) = EstimatedRTT(n) + 4*DevRTT(n)
```

Example Timeout Estimation:



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TCP Reliable Data Transfer

- TCP creates rdt service on top of IP's unreliable service
  - Hybrid of Go-back-N and Selective Repeat
- Pipelined segments
- Cumulative acks
- TCP uses single retransmission timer
  - For the oldest unACK'ed packet
- Retransmissions are triggered by:
  - Timeout events
  - Duplicate acks
- Initially consider simplified TCP sender:
  - Ignore duplicate acks
  - Ignore flow control, congestion control

TCP Sender (Simplified)

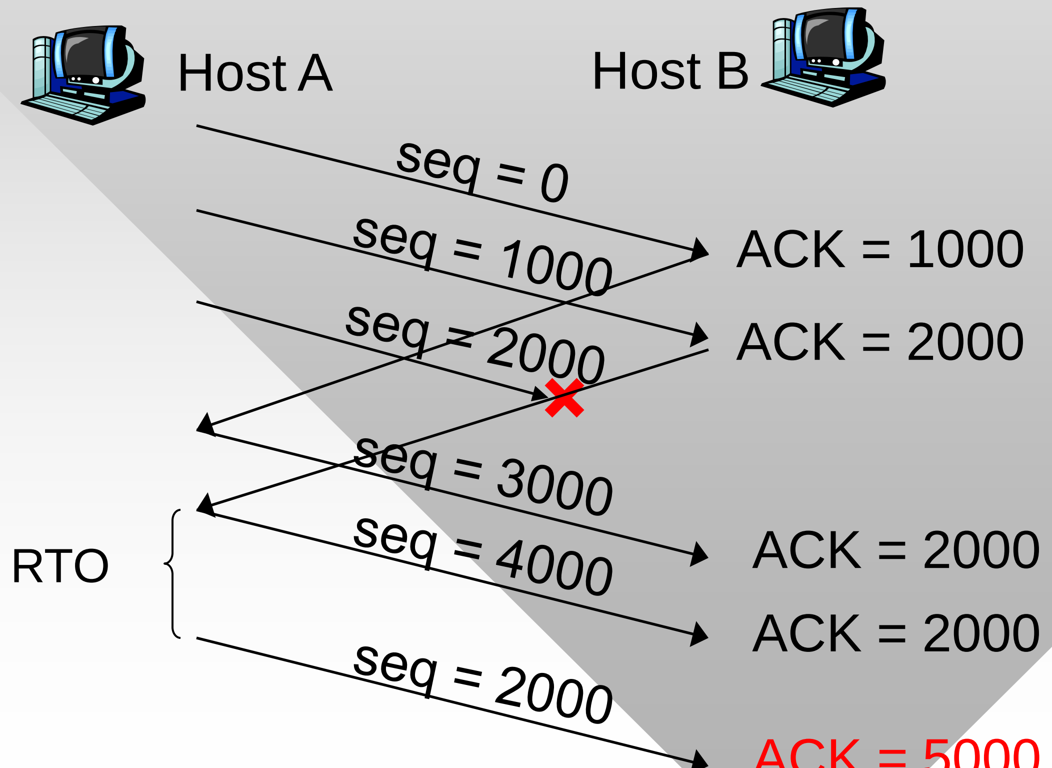
```
NextSeqNum = InitialSeqNum // random for each transfer
SendBase = InitialSeqNum
 loop (forever) {
   switch(event) {
   (a) data received from application above (assuming it fits into window):
         create TCP segment with sequence number NextSeqNum
         if (timer currently not running)
                  start timer
         pass segment to IP
         NextSeqNum = NextSeqNum + length(data)
   (b) timeout:
         retransmit pending segment with smallest sequence
         number (i.e., SendBase); restart timer
   (c) ACK received, with ACK field value of y
     if (y > SendBase) {
       SendBase = y
       if (there are currently not-yet-acknowledged segments)
          restart timer with latest RTO
       else cancel timer }
  }
} /* end of loop forever */
```

TCP Seq. #'S and ACKs

FTP Example:

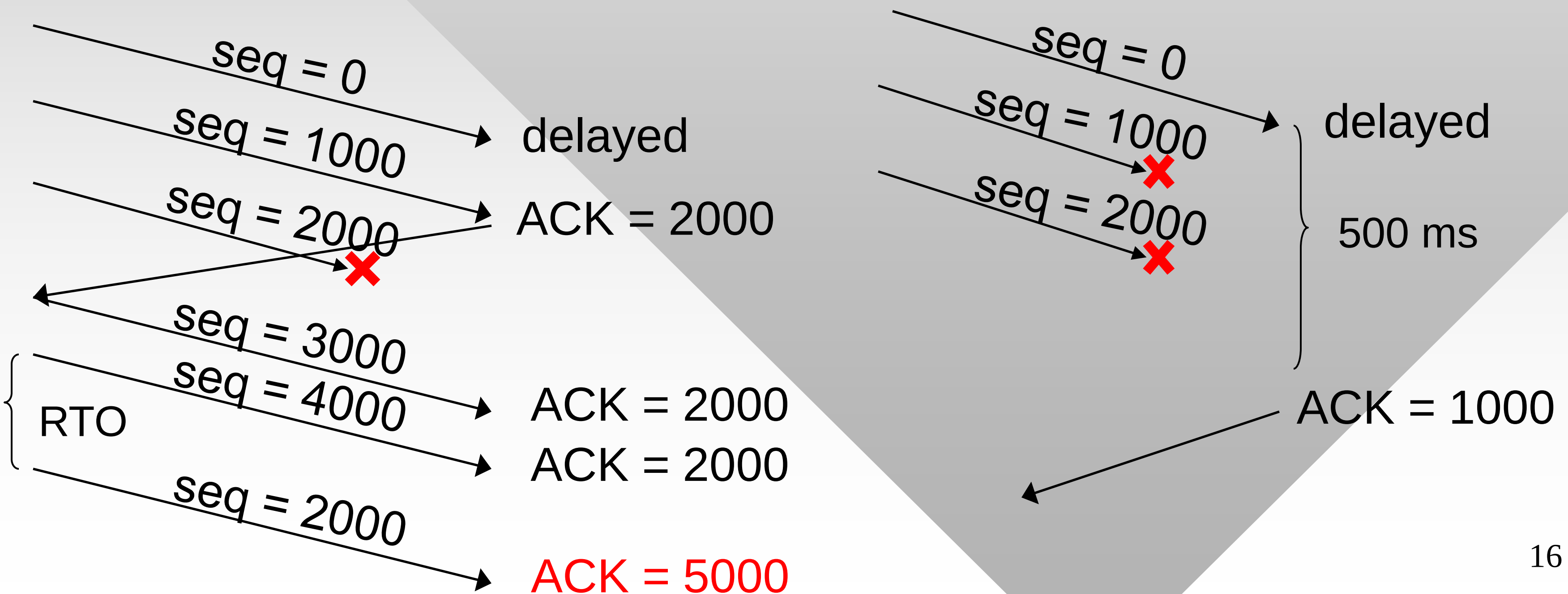
- Suppose MSS = 1,000 bytes and the sender has a large file to transmit (we ignore seq field in ACKs and ACK field in data pkts)

What is the sender window size?



TCP ACK Generation [RFC 1122, RFC 2581]

- Receiver immediately ACKs the base of its window in all cases except Nagle's algorithm:
  - For *in-order* arrival of packets, send ACKs for every *pair* of segments; if second segment of a pair not received in 500ms, ACK the first one alone



Fast Retransmit

- Time-out period often relatively long
  - Especially in the beginning of transfer (3 seconds in RFC 1122)
- Idea: infer loss via duplicate ACKs
  - Sender often sends many segments back-to-back
  - If a segment is lost, there will be many duplicate ACKs
- If sender receives 3 duplicate ACKs for its base, it assumes this packet was lost
  - Fast Retransmit: resend the base segment immediately (i.e., without waiting for RTO)
- Note that reordering may trigger unnecessary retx
  - To combat this problem, modern routers avoid load-balancing packets of same flow along multiple paths

Fast Retransmit Algorithm:

```
(c) event: ACK received, with ACK field value of y
        if (y > SendBase) {
            SendBase = y; dupACK = 0;
            if (SendBase != NextSeqNum)
                restart timer with latest RTO;
            else
                cancel timer;  // last pkt in window
        }
        else if (y == SendBase) {
            dupACK++;
            if (dupACK == 3)
                { resend segment with sequence y; restart timer}
        }
```

a duplicate ACK for already ACKed segment

fast retransmit

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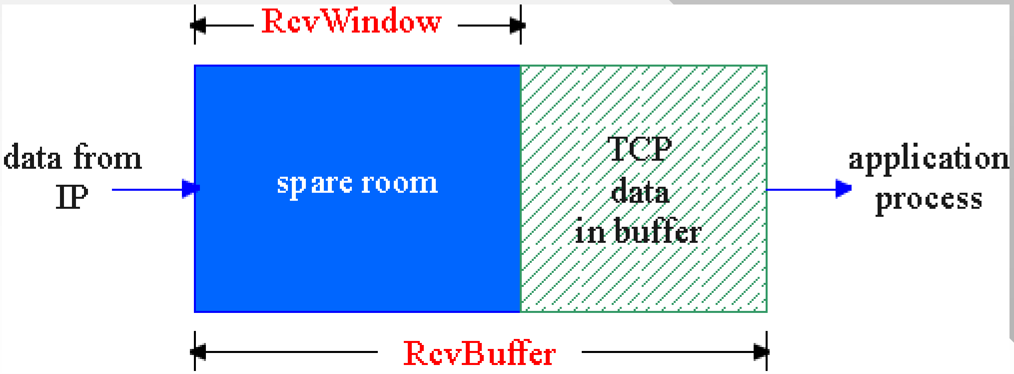
TCP Flow Control

- Assume packets received without loss, but the application does not call recv()
  - How to prevent sender from overflowing TCP buffer?

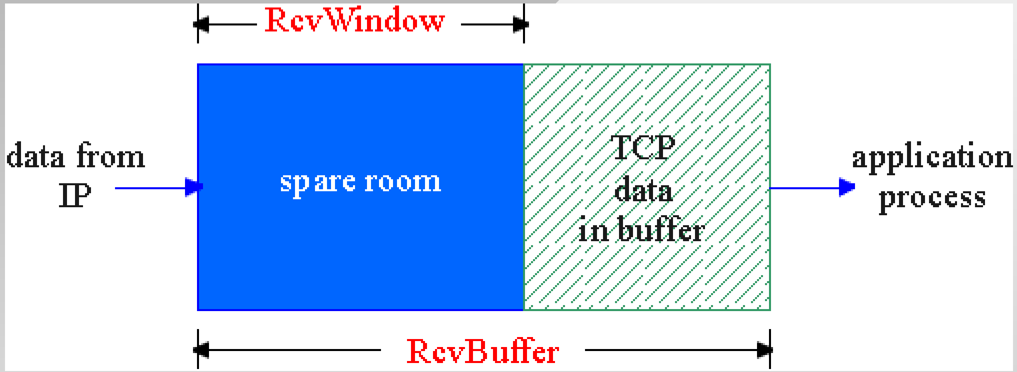
Flow control

Sender won't overflow receiver buffer by transmitting too much, too fast

- Speed-matching service: sender rate to suit the receiving app's ability to process incoming data



TCP Flow Control: How It Works



- Receiver advertises spare room by including value of `RcvWin` in segments
- Sender enforces `seq < ACK + RcvWin`
  - Guarantees receiver buffer doesn't overflow

- Spare room in buffer

```
RcvWin = RcvBuffer –
[LastByteReceivedInOrder - LastByteDelivered]
```

last ACK-1

went to application

- Combining both constraints (sender, receiver):

```
seq < min(sndBase+sndWin, ACK+RcvWin)
```

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TCP Connection Management

- Purpose of connection establishment:
  - Exchange initial seq #s
  - Exchange flow control info (i.e., `RcvWin`)
  - Negotiate options (SACK, large windows, etc.)

Three way handshake:

- Step 1: client sends TCP SYN to server
  - Specifies initial seq # X and buffer size `RcvWin`
  - No data, ACK = 0
- Step 2: server gets SYN, replies with SYN+ACK
  - Sends server initial seq # Y and buffer size `RcvWin`
  - No data, ACK = X+1
- Step 3: client receives SYN+ACK, replies with ACK segment
  - Seq = X+1, ACK = Y+1
  - May contain regular data, but many servers will break
- Step 4: regular packets
  - Seq = X+1, ACK = Y+1

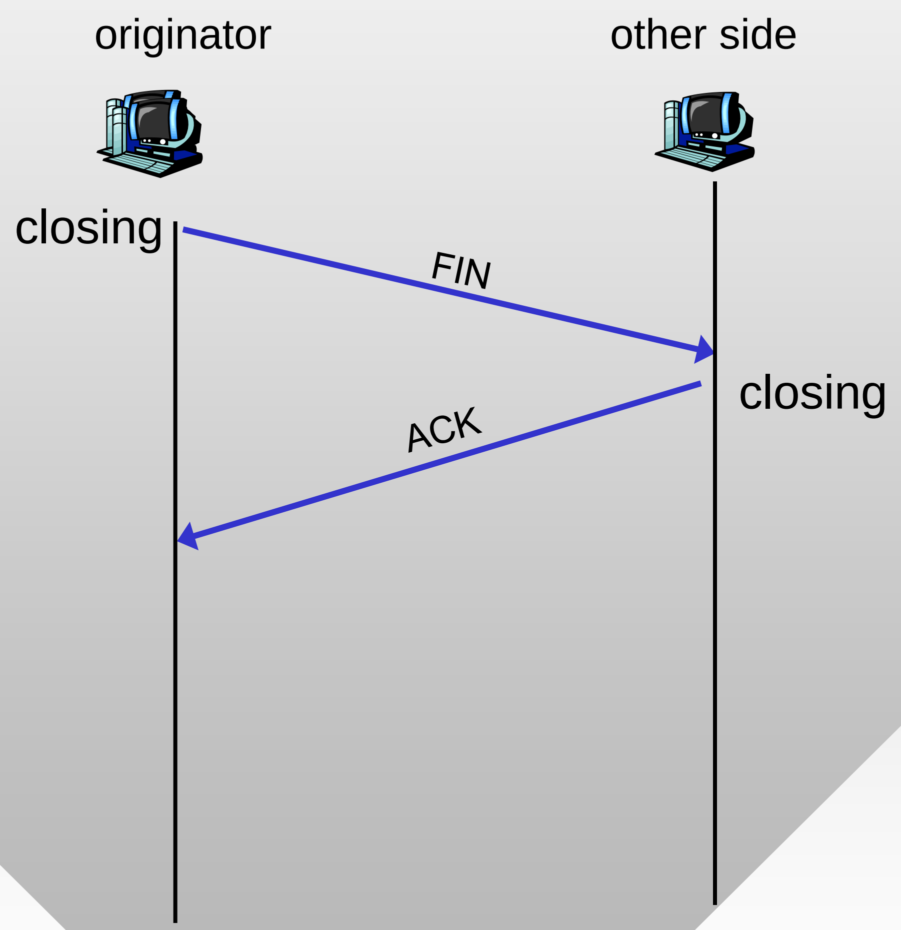
TCP Connection Management (Cont.)

Closing a connection:

- Closing a socket:
  `closesocket(sock);`

Step 1: originator end system sends TCP FIN control segment to server

Step 2: other side receives FIN, replies with ACK. Connection in “closing” state, sends FIN



TCP initiates a close when it has all ACKs for the transmitted data

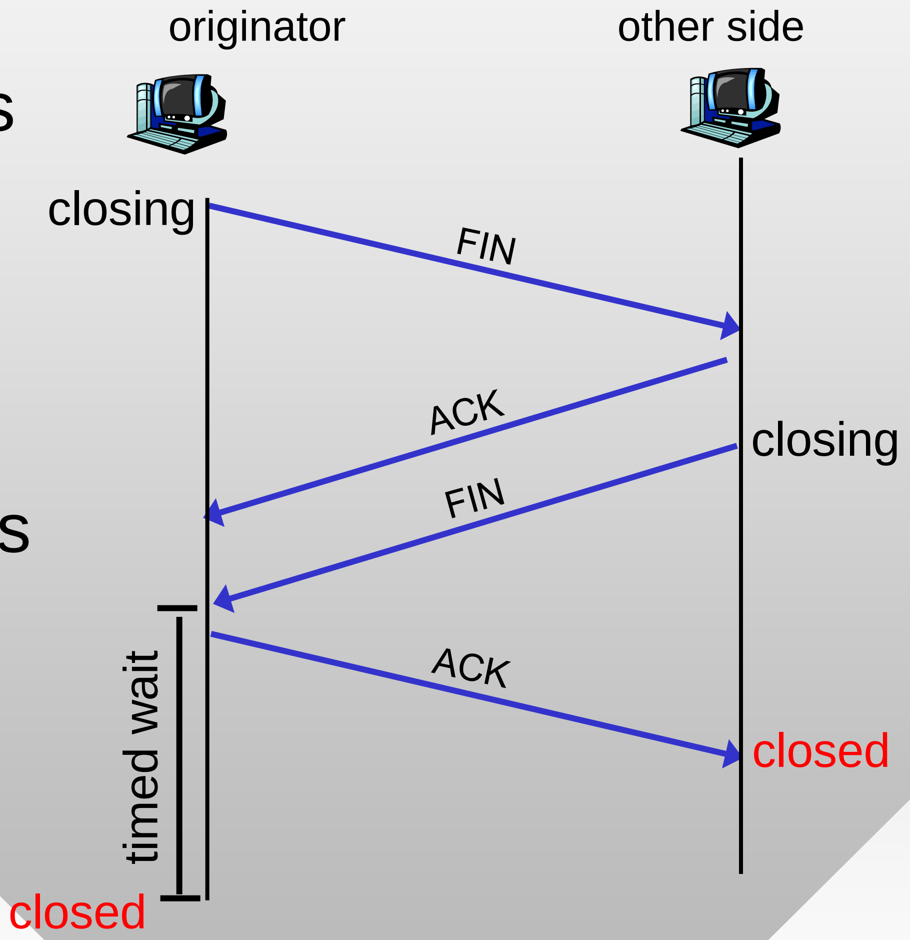
TCP Connection Management (Cont.)

Step 3: originator receives FIN, replies with ACK

- Enters "timed wait" - will respond with ACK to received FINs

Step 4: other side receives ACK; its connection considered closed

Step 5: after a timeout (TIME_WAIT state lasts 240 seconds), originator's connection is closed as well



birectional transfer means both sides must agree to close