

CSCE 463/612

Networks and Distributed Processing
Fall 2025

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):**
 - Client socket connected to server socket
- **Reliable, in-order *byte stream*:**
 - Packet boundaries are not visible to the application
- **Pipelined:**
 - TCP congestion/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:**
 - Handshake (exchange of control msgs) initializes sender / receiver state before transfer begins
- **Flow controlled:**
 - Sender will not overwhelm receiver



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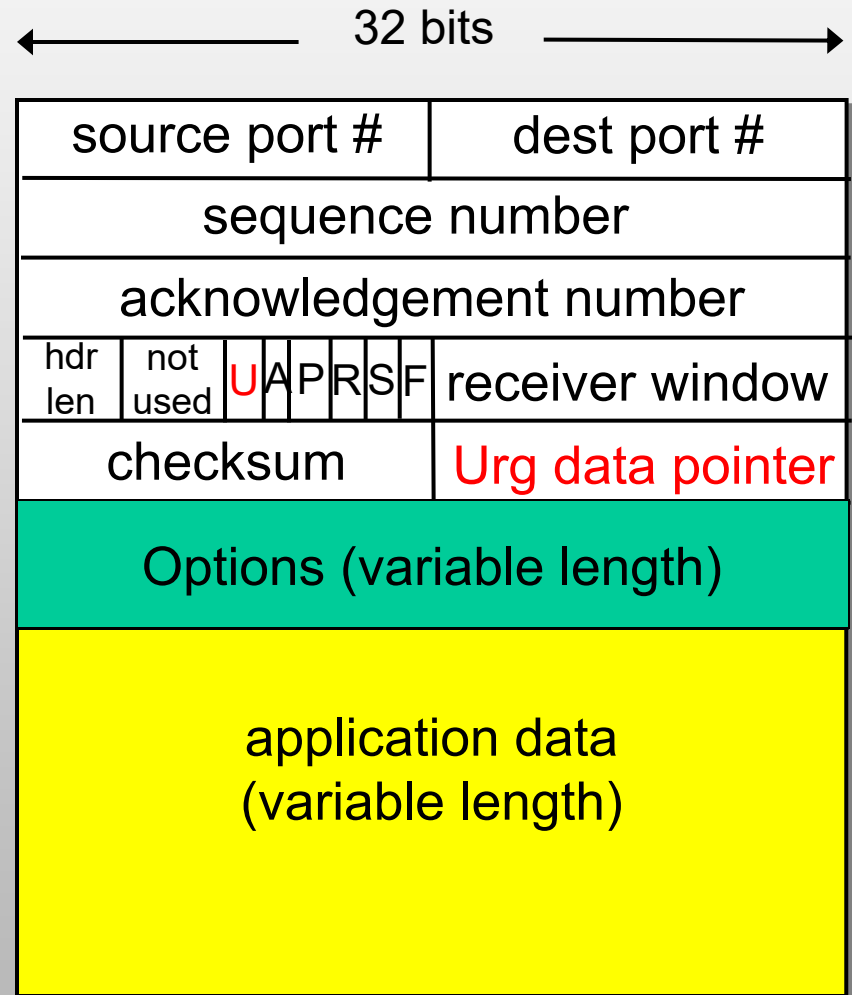
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TCP Segment Structure

- 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 (reduce latency)
 - 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:

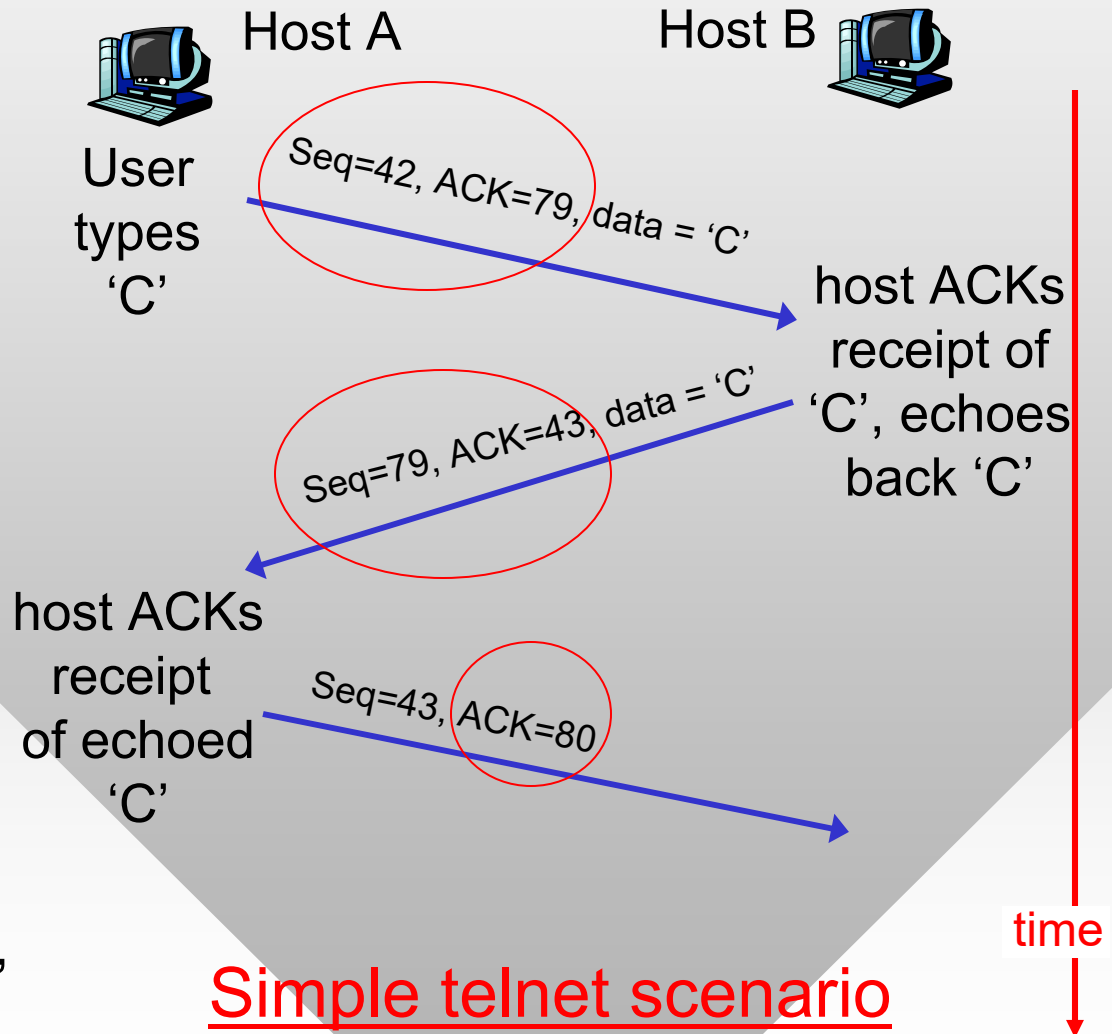
- Sequence number of the **first byte** in segment's data

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



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**

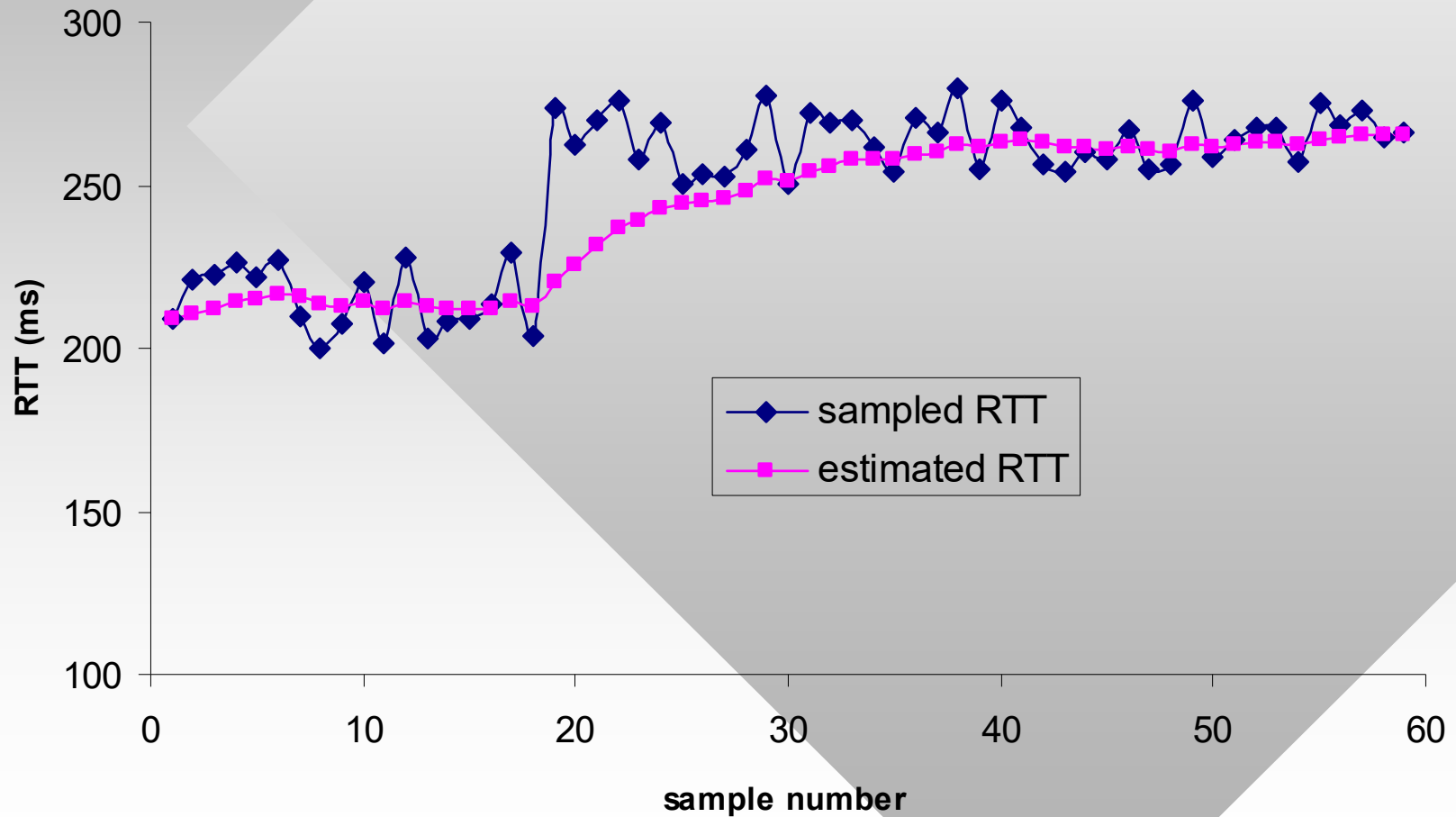
TCP Round Trip Time and Timeout

$$\text{EstimatedRTT}(n) = (1-\alpha) * \text{EstimatedRTT}(n-1) + \alpha * \text{SampleRTT}(n)$$

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

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

Example RTT Estimation:



TCP Round Trip Time and Timeout

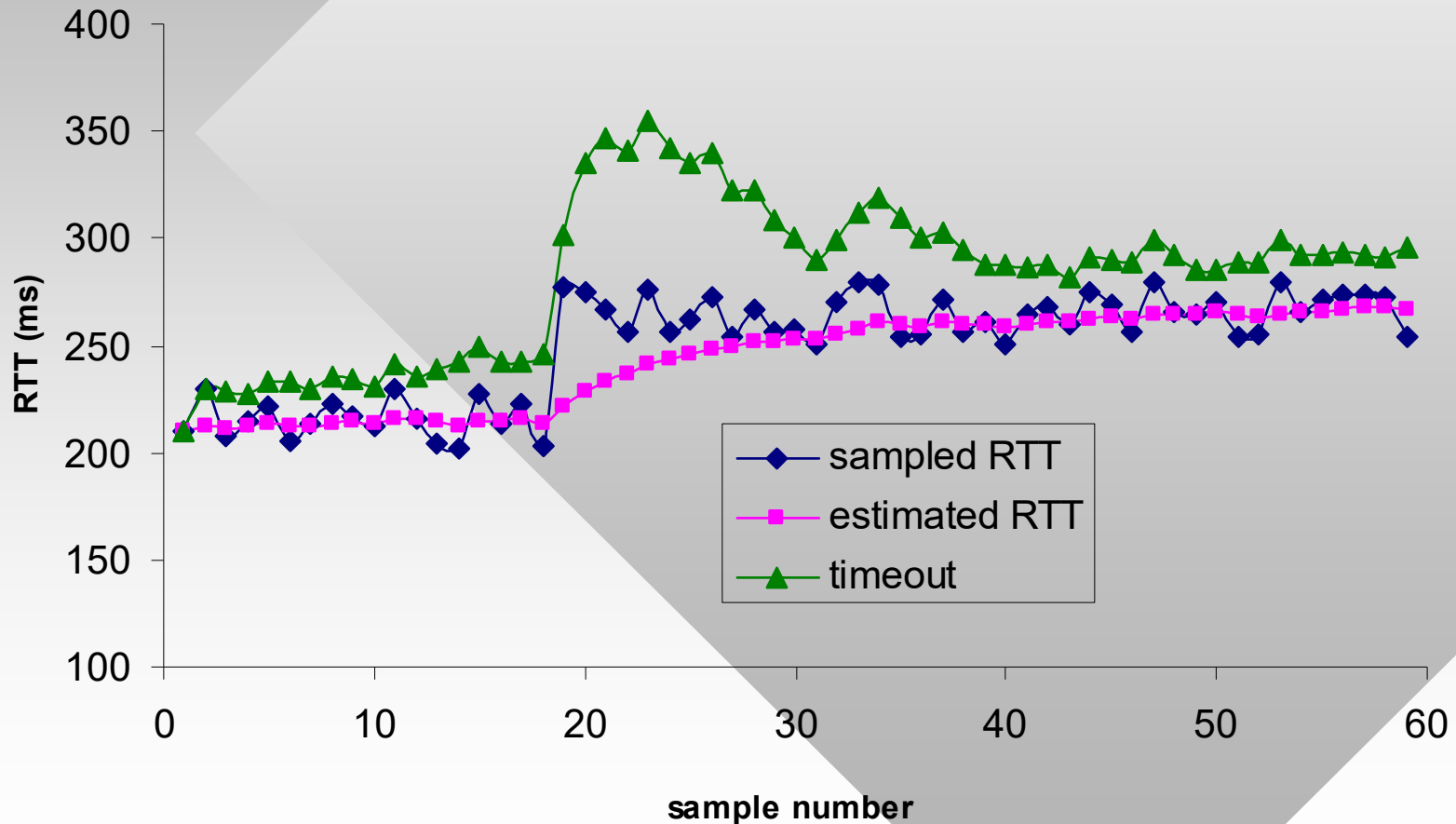
- Setting the timeout:
- EstimatedRTT plus a “safety margin”
 - Larger variation in EstimatedRTT → larger safety margin
- First estimate how much SampleRTT deviates from EstimatedRTT (typically, $\beta = 1/4$):

$$\text{DevRTT}(n) = (1-\beta) * \text{DevRTT}(n-1) + \beta * |\text{SampleRTT}(n) - \text{EstimatedRTT}(n)|$$

Then set retransmission timeout (RTO):

$$\text{RTO}(n) = \text{EstimatedRTT}(n) + 4 * \text{DevRTT}(n)$$

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
 - Retx only the base
- Retransmissions are triggered by:
 - Timeout events
 - Duplicate acks
- Initially consider simplified TCP sender:
 - Ignore duplicate acks
 - Ignore flow control, congestion control

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 */

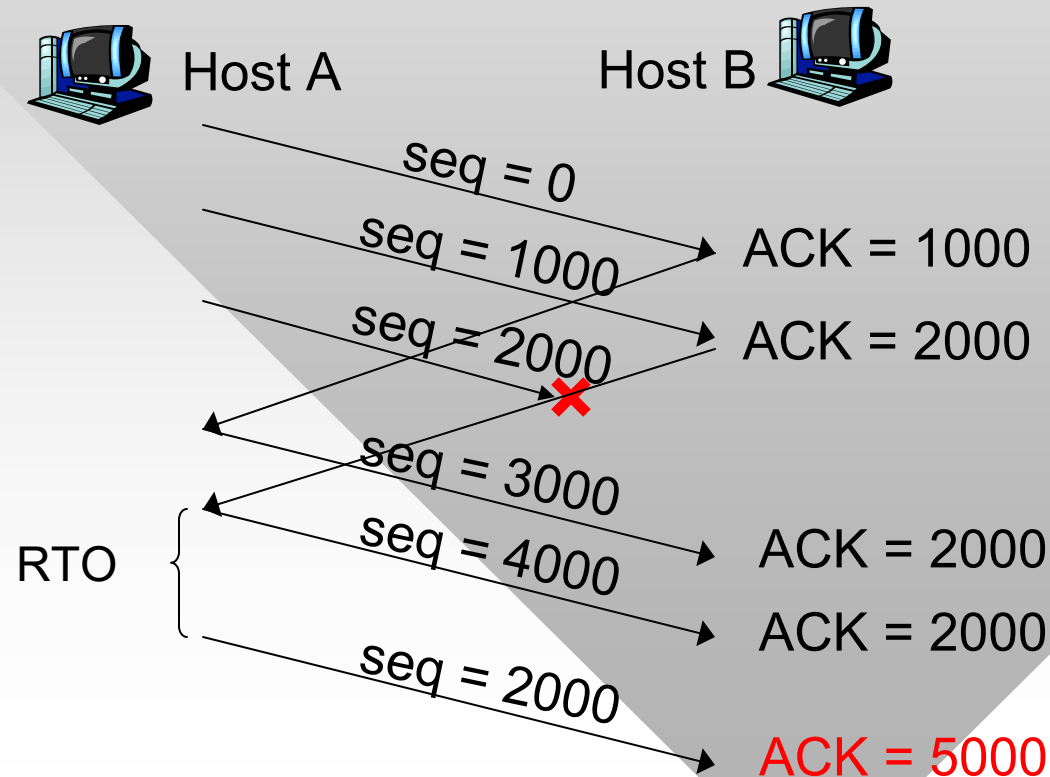
TCP Sender
(Simplified)

TCP Seq. #'S and ACKs

FTP Example:

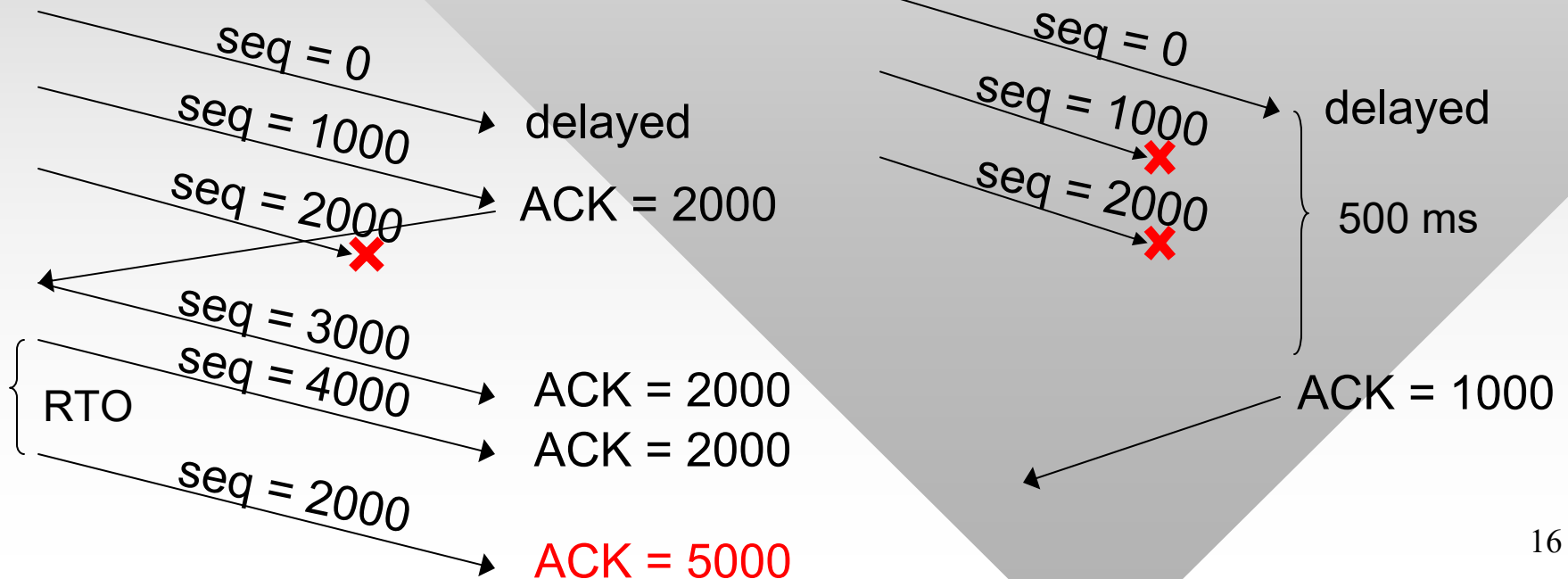
- Suppose MSS = 1,000 bytes and the sender has a large file to transmit (we ignore seq field in ACKs and ACK field in data pkts)

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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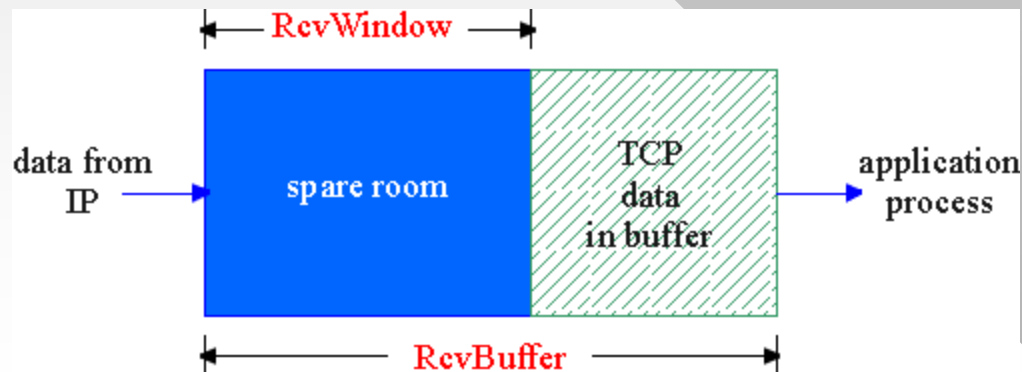
TCP Flow Control

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

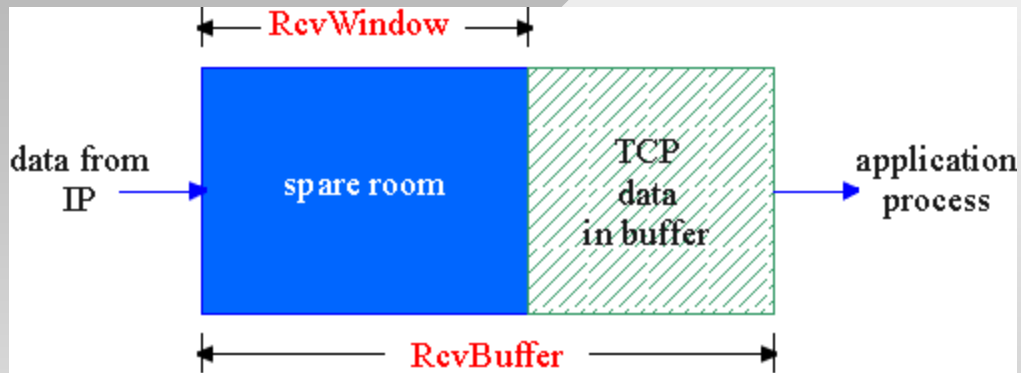
Flow control

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



- Spare room in buffer

$$\text{RcvWin} = \text{RcvBuffer} - [\text{LastByteReceivedInOrder} - \text{LastByteDelivered}]$$

↑
last ACK-1

↑
went to application

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

- Combining both constraints (sender, receiver):
 $\text{seq} < \min(\text{sndBase} + \text{sndWin}, \text{ACK} + \text{RcvWin})$

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

Three way handshake:

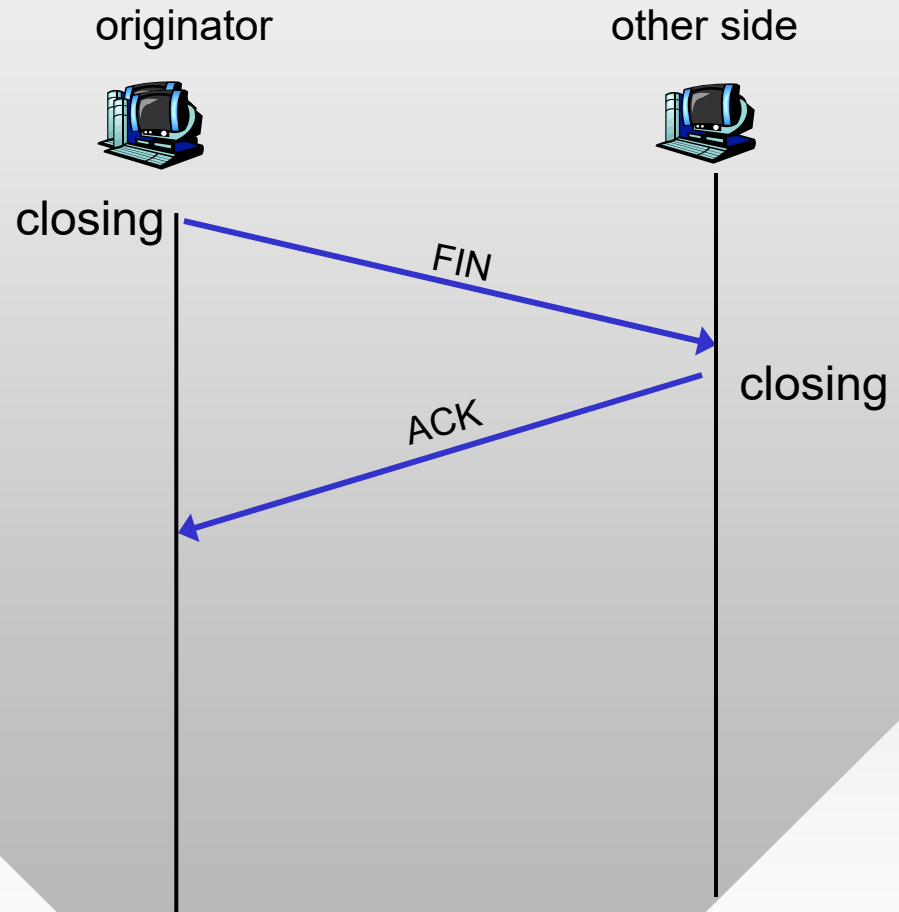
TCP Connection Management (Cont.)

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

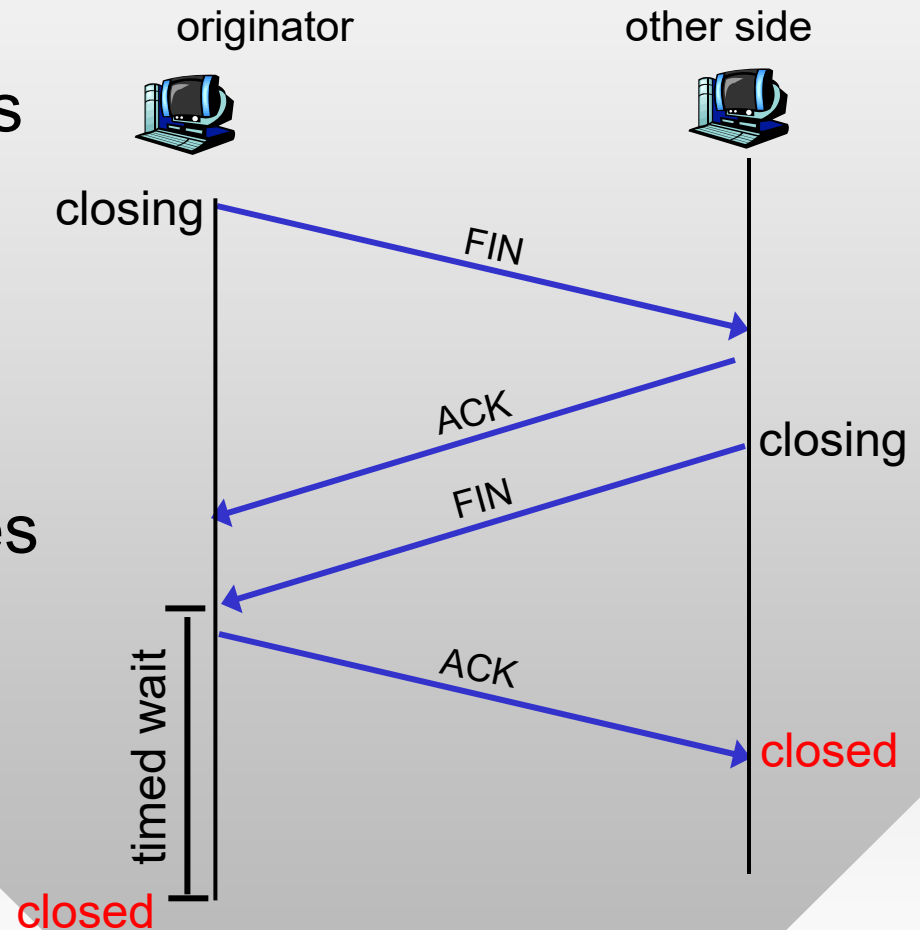
TCP Connection Management (Cont.)

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



biirectional transfer means both sides must agree to close