

Transport Layer: Outline

- ❑ Transport-layer services
- ❑ Multiplexing and demultiplexing
- ❑ Connectionless transport: UDP
- ❑ Principles of reliable data transfer
- ❑ **Connection-oriented transport: TCP**
 - Segment structure
 - Reliable data transfer
 - Flow control
 - Connection management
- ❑ Principles of congestion control
- ❑ TCP congestion control

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TCP: Overview

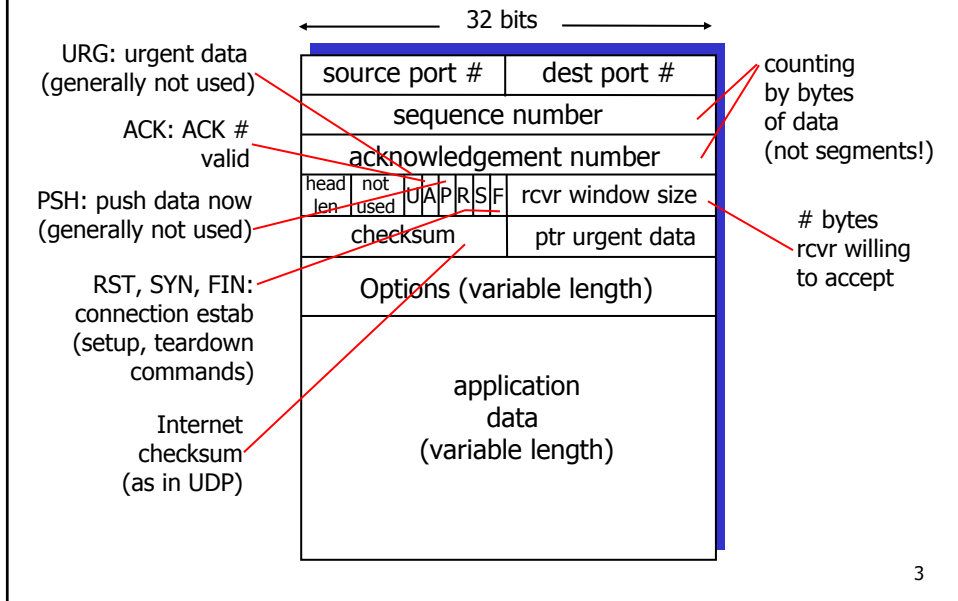
RFCs: 793, 1122, 1323, 2018, 2581

- ❑ **Point-to-point:**
 - One sender, one receiver
- ❑ **Reliable, in-order *byte stream*:**
 - No "message boundaries"
- ❑ **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
- ❑ **Connection-oriented:**
 - Handshaking (exchange of control msgs) init's sender, receiver state before data exchange
- ❑ **Flow controlled:**
 - Sender will not overwhelm receiver
- ❑ **Congestion controlled:**
 - Sender will not overwhelm receiver



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TCP segment structure



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TCP reliable data transfer

- TCP creates rdt service on top of IP's unreliable service
- Pipelined segments
- Cumulative acks
- TCP uses single retransmission timer
- Retransmissions are triggered by:
 - Timeout events
 - Duplicate acks
- Initially consider simplified TCP sender:
 - Ignore duplicate acks
 - Ignore flow control, congestion control
 - One way dataflow

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TCP seq. #'s and ACKs

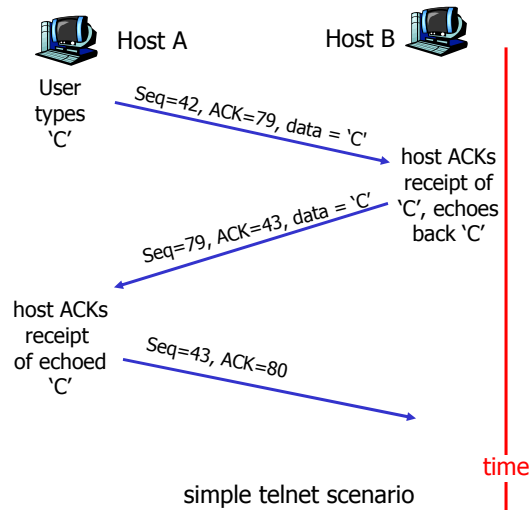
Seq. #'s:

- Byte stream "number" of first byte in segment's data

ACKs:

- Seq # of next byte expected from other side
- cumulative ACK

- Q:** How receiver handles out-of-order segments
- A: TCP spec doesn't say, - up to implementor



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TCP sender events

Data rcvd from app:

- ❑ Create segment with seq #
- ❑ Seq # is byte-stream number of first data byte in segment
- ❑ Start timer if not already running (think of timer as for oldest unacked segment)
- ❑ Expiration interval: `TimeoutInterval`

Timeout:

- ❑ Retransmit the **one** segment that caused timeout
- ❑ Restart timer

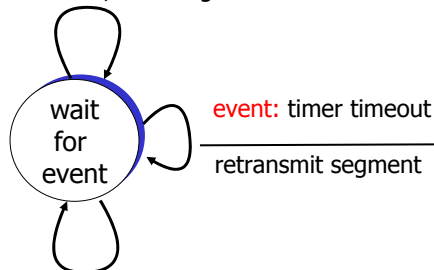
Ack rcvd:

- ❑ If acknowledges previously unacked segments
 - Update what is known to be acked
 - Restart timer if there are outstanding segments

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TCP: reliable data transfer

event: data received from application above
create, send segment



event: ACK received, with ACK # y
ACK processing

Simplified sender, assuming

- One way data transfer
- No flow, congestion control

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TCP sender (simplified)

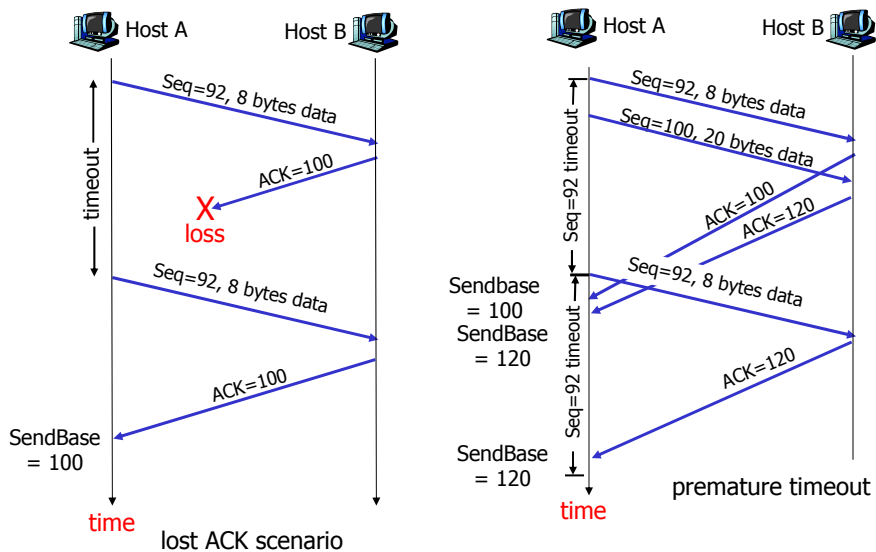
```

00 sendbase = initial_sequence number
01 nextseqnum = initial_sequence number
02
03 loop (forever) {
04   switch(event)
05     event: data received from application above
06       create TCP segment with sequence number nextseqnum
07       if (timer currently not running) start timer
08       pass segment to IP
09       nextseqnum = nextseqnum + length(data)
10     event: timer timeout
11       retransmit not-yet-acknowledged segment with
12         smallest sequence number
13       restart timer
14     event: ACK received, with ACK field value of y
15       if (y > sendbase) { /* cumulative ACK of all data up to y */
16         sendbase = y
17         if (currently not-yet-acknowledged segments) {
18           restart timer
19         }
20       }
21     }
22   } /* end of loop forever */

```

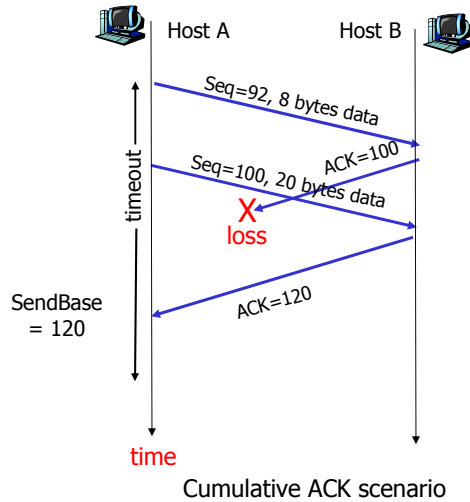
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TCP retransmission scenarios



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TCP retransmission scenarios (2.)



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TCP round trip time and timeout

Q: How to set TCP timeout value?

- Longer than RTT
 - Note: RTT will vary
- Too short: premature timeout
 - Unnecessary retransmissions
- Too long: slow reaction to segment loss

Q: How to estimate RTT?

- **SampleRTT**: measured time from segment transmission until ACK receipt
 - Ignore retransmissions, cumulatively ACKed segments
- **SampleRTT** will vary, want estimated RTT "smoother"
 - Use several recent measurements, not just current **SampleRTT**

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TCP round trip time and timeout

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

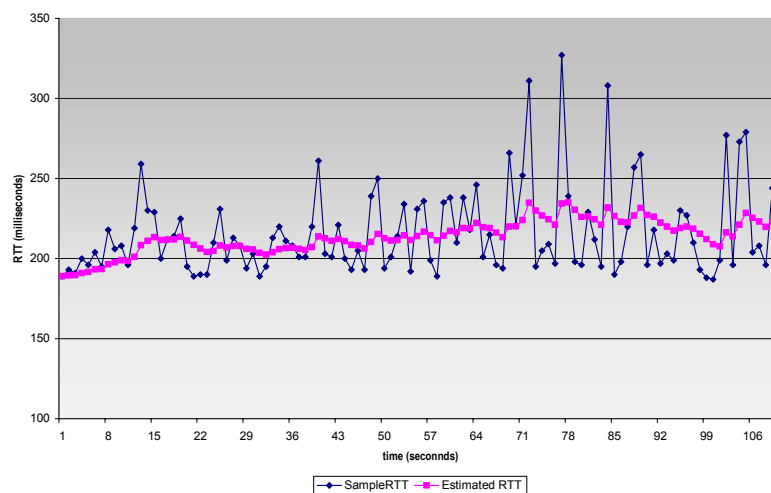
- Exponential weighted moving average
- Influence of given sample decreases exponentially fast
- Typical value of α : 0.125

- Key observation:
 - At high loads round trip variance is high

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Example RTT estimation

RTT: gaia.cs.umass.edu to fantasia.eurecom.fr



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TCP round trip time and timeout

Setting the timeout

- **EstimatedRTT** plus "safety margin"
 - Large variation in **EstimatedRTT** -> larger safety margin
- First estimate of how much **SampleRTT** deviates from **EstimatedRTT**:

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

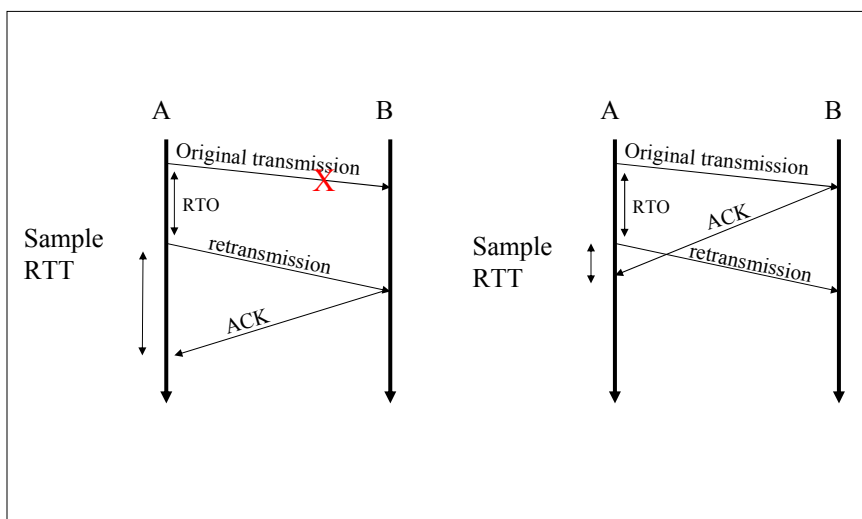
(typically, $\beta = 0.25$)

Then set timeout interval:

$$\text{TimeoutInterval} = \text{EstimatedRTT} + 4 * \text{DevRTT}$$

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Retransmission ambiguity



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Karn's RTT estimator

- ❑ Accounts for retransmission ambiguity
 - If a segment has been retransmitted: Don't count RTT sample on ACKs for this segment
- ❑ If retransmission timer expires
 - Double retransmission TimeoutInterval
 - Do not use RTT estimate to calculate TimeoutInterval until successful retransmission
- ❑ Timer restarted (not due to timeout)
 - Reuse RTT estimate

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Timestamp extension

- ❑ Used to improve timeout mechanism by more accurate measurement of RTT
- ❑ When sending a packet, insert current timestamp into option
 - 4 bytes for seconds, 4 bytes for microseconds
- ❑ Receiver echoes timestamp in ACK
 - Actually will echo whatever is in timestamp
- ❑ Removes retransmission ambiguity
 - Can get RTT sample on any packet

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Timer granularity

- Many TCP implementations set RTO in multiples of 200, 500, 1000ms
- Why?
 - Avoid spurious timeouts – RTTs can vary quickly due to cross traffic
 - Make timers interrupts efficient

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Fast retransmit

- Time-out period often relatively long:
 - Long delay before resending lost packet
- Detect lost segments via duplicate ACKs.
 - Sender often sends many segments back-to-back
 - If segment is lost, there will likely be many duplicate ACKs.
- If sender receives 3 ACKs for the same data, it supposes that segment after ACKed data was lost:
 - **Fast retransmit:** resend segment before timer expires

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Fast retransmit algorithm:

```

event: ACK received, with ACK field value of y
  if (y > SendBase) {
    SendBase = y
    if (there are currently not-yet-acknowledged segments)
      restart timer
  }
  else {
    increment count of dup ACKs received for y
    if (count of dup ACKs received for y = 3) {
      resend segment with sequence number y
    }
  }
  
```

Duplicate ACK for
already ACKed segment

Fast retransmit

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TCP ACK generation [RFC 1122, RFC 2581]

<u>Event at Receiver</u>	<u>TCP Receiver action</u>
In-order segment with expected seq #. All data up to expected seq # already ACKed	Delayed ACK. Wait up to 500ms for next segment. If no next segment, send ACK
In-order segment with expected seq #. One other segment has ACK pending	Immediately send single cumulative ACK, ACKing both in-order segments
Out-of-order segment higher-than-expected seq. # . Gap detected	Immediately send <i>duplicate ACK</i> , indicating seq. # of next expected byte
Segment that partially or completely fills gap	Immediate send ACK, provided that segment starts at lower end of gap

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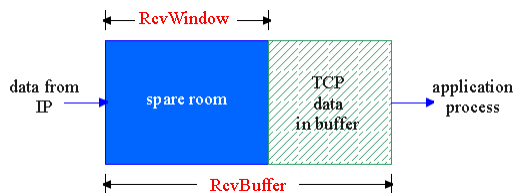
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TCP flow control

- ❑ Receive side of TCP connection has a receive buffer:



- ❑ App process may be slow at reading from buffer

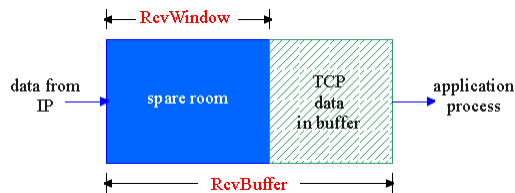
flow control

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

- ❑ Speed-matching service: match the send rate to the receiving app's drain rate

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TCP flow control: How it works



(Suppose TCP receiver discards out-of-order segments)

- Spare room in buffer

= $RcvWindow$

= $RcvBuffer - [LastByteRcvd - LastByteRead]$

- Rcvr advertises spare room by including value of $RcvWindow$ in segments
- Sender limits unACKed data to $RcvWindow$
 - Guarantees receive buffer doesn't overflow

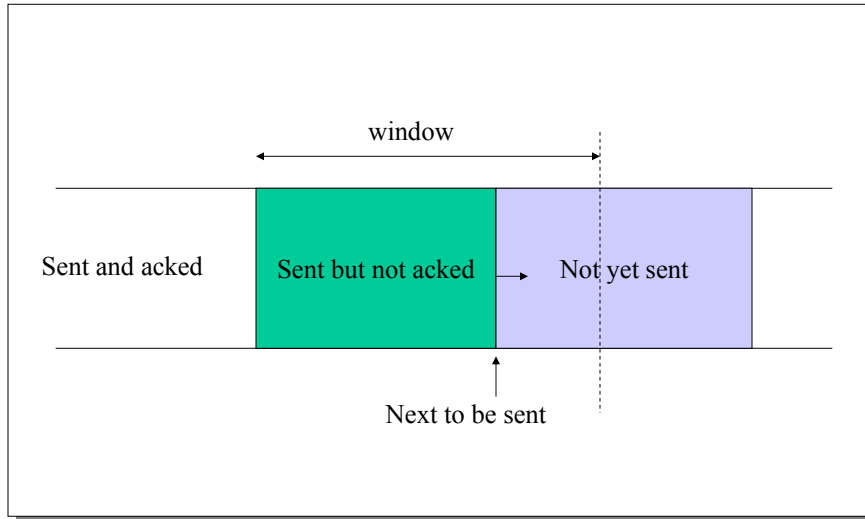
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TCP flow control: How it works (2.)

- TCP is a sliding window protocol
 - For window size n , can send up to n bytes without receiving an acknowledgement
 - When the data is acknowledged then the window slides forward
- Each packet advertises a window size
 - Indicates number of bytes the receiver has space for
- Original TCP always sent entire window
 - Congestion control now limits this

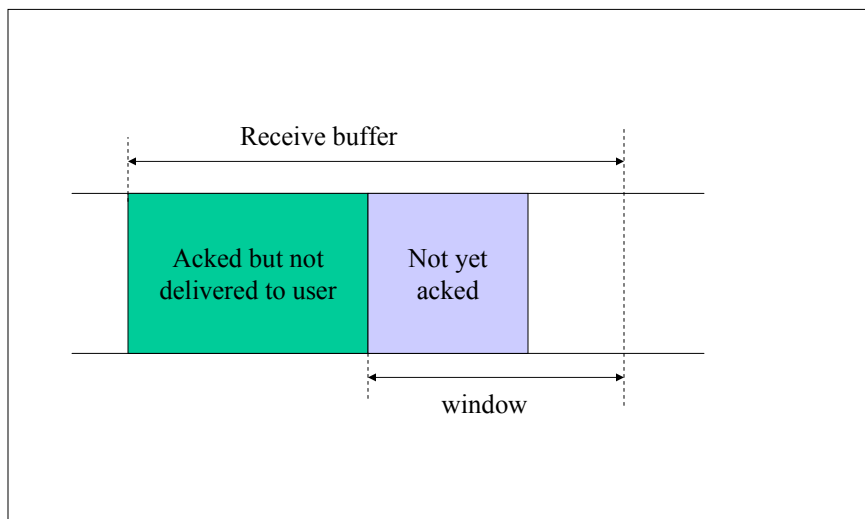
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Window flow control: Sender side



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Window flow control: Receiver side



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TCP persist

- ❑ What happens if window is 0?
 - Receiver updates window (i.e., sends ACK with new window size) when application reads data
 - What if this update is lost?
- ❑ TCP persist state
 - Sender periodically sends 1 byte packets
 - Receiver responds with ACK even if it can't store the packet