TCP Reliability

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How does TCP implement reliable transfer?
TCP Segmentation

Process writes data

Socket

TCP send buffer

Segment

Segment

TCP receive buffer

Process reads data

Socket
Sequence and ACK Numbers

- sequence number: byte number of this segment within the byte stream
- ACK number: sequence number of next byte expected from sender
TCP Retransmission Scenarios

Host A

Timeout

Seq=92, 8 bytes data

ACK=100

X (loss)

Seq=92, 8 bytes data

ACK=100

Time

Host B

Seq=92, 8 bytes data

ACK=100

Time

Host A

seq=92 timeout interval

Seq=100, 20 bytes data

ACK=100

Seq=92, 8 bytes data

ACK=120

Time

Host B

Seq=92, 8 bytes data

ACK=120

Time
TCP Retransmission Scenarios (continued)

Diagram showing the sequence of events in TCP retransmission scenarios. The diagram includes steps such as sending a segment (e.g., Seq=100, 20 bytes data), receiving an acknowledgment (ACK=100), and detecting a loss (Seq=92 timeout interval).
# Generating TCP ACKs

<table>
<thead>
<tr>
<th>Receiver Event</th>
<th>TCP Action</th>
</tr>
</thead>
<tbody>
<tr>
<td>Arrival of in-order segment with expected sequence number. All previous data ACKed.</td>
<td>Delayed ACK. Wait up to 500 ms for next segment. If no next segment, send ACK.</td>
</tr>
<tr>
<td>Arrival of in-order segment with expected sequence number. One other segment has ACK pending.</td>
<td>Immediately send single cumulative ACK that covers both in-order segments.</td>
</tr>
<tr>
<td>Arrival of out-of-order segment, sequence number larger than expected. Gap detected.</td>
<td>Immediately send duplicate ACK, indicating sequence number of next expected byte.</td>
</tr>
<tr>
<td>Arrival of segment that partially or completely fills gap.</td>
<td>Immediately send ACK, provided that segment starts at lower end of gap.</td>
</tr>
</tbody>
</table>
RTT Estimation
Setting the TCP Timer

- how long should the TCP timer be?
  - needs to be longer than RTT
  - too short: premature timeout - duplicate segments
  - too long: slow reaction to segment loss

- problem: RTT can vary dramatically, depending on queueing delay
Estimating RTT

- **SampleRTT**: measured time from segment transmission until ACK is received
  - ignore retransmissions due to loss
- average measurements of SampleRTT over time
- \(\text{EstimatedRTT} = (1 - \alpha) \times \text{EstimatedRTT} + \alpha \times \text{SampleRTT}\)
  - EWMA: exponential weighted moving average
  - influence of past samples decreases exponentially fast
  - typical value \(\alpha = 0.125\)
  - produces a smooth estimate of RTT
Example

- sample delay from a machine at UMASS to a machine at Eurecom in France
Fine-Tuning the RTT Estimate

- calculate the deviation of the `SampleRTT` from the `EstimatedRTT`

  \[ \text{DevRTT} = (1 - \beta) \times \text{DevRTT} + \beta \times |\text{SampleRTT} - \text{EstimatedRTT}| \]
  - typical value \( \beta = 0.25 \)

- `TimeoutInterval = EstimatedRTT + 4 \times \text{DevRTT}`
  - large variation in estimated RTT provides a larger safety margin
Fast Retransmit
Fast Retransmit

- problem: timeout period is generally very long
  - long delay when recovering lost packet
  - big performance hit
- solution: detect lost segments with duplicate ACKs
  - duplicate acks when one packet in a window is lost
  - if sender gets 3 duplicate ACKs, assume packet loss
  - fast retransmit before timer expires
- why not switch to selective ACKs?
  - requires only a small change to TCP sender
Fast Retransmit Example

Host A

- seq=92, 8 bytes of data
- seq=100, 20 bytes of data
- seq=120, 15 bytes of data
- seq=135, 6 bytes of data
- seq=141, 16 bytes of data

Timeout

- seq=100, 20 bytes of data

Host B

- ack=100
- ack=100
- ack=100
- ack=100
Flow Control
Flow Control

- receiver stores incoming packets in a receive buffer
- application pulls data from the buffer
- TCP can’t control how fast data is removed from buffer, so it may fill up
- flow control: ensure sender doesn’t overflow the receiver’s buffer
Advertised Window

- TCP receiver advertises a **RcvWindow** to sender, which is equivalent to the spare room in its buffer
- \[ \text{RcvWindow} = \text{RcvBuffer} - (\text{LastByteRecvd} - \text{LastByteRead}) \]
- sender limits un-ACKed data to **RcvWindow**
Connection Management
Opening a Connection: Three-Way Handshake

1. client sends TCP SYN segment
   - client specifies initial sequence number, MSS, RcvWindow
   - no data
2. server responds with SYN/ACK segment
   - server allocates buffers
   - server specifies initial sequence number, MSS
3. client responds with ACK segment
   - may contain data
Closing a Connection

1. Client sends TCP FIN segment to server
2. Server responds with ACK
3. Server sends FIN segment
4. Client responds with ACK
   - Enters timed wait
   - Responds to FINs with ACK

- Lots of variations, e.g. combining FIN/ACK
- Various scenarios lead to different timers being set
TCP Client Connection States

- CLOSED
  - Client application initiates a TCP connection
  - Send SYN

- SYN_SENT
  - Receive SYN & ACK, send ACK

- ESTABLISHED
  - Send FIN
  - Client application initiates close connection

- FIN_WAIT_1
  - Receive ACK, send nothing

- FIN_WAIT_2
  - Receive FIN, send ACK

- TIME_WAIT
  - Wait 30 seconds

- CLOSED
  - grossly simplified
TCP Server Connection States

- CLOSED
  - Server application creates a listen socket
- LISTEN
  - Receive SYN, send SYN & ACK
- SYN_RCVD
  - Receive ACK, send nothing
- ESTABLISHED
  - Receive FIN, send ACK
- CLOSE_WAIT
  - Send FIN
- LAST_ACK
  - Receive ACK, send nothing

- grossly simplified
TCP State Transitions
TCP Header

- **sequence and ACK number**: count in terms of bytes
- **flags**
  - A: ACK number is valid
  - R: RST: reset connection
  - S: SYN: establish connection
  - F: FIN: close connection
  - U: URG: urgent data, typically not used
  - P: PSH: push (send) data immediately, used for TELNET
- **receive window**: number of bytes receiver can accept