Connection-oriented Transport: TCP

Outline:
- segment structure
- reliable data transfer
- flow control
- connection management

Principles on Reliable Data Transfer
- End-to-end solution to achieve reliable data transfer
  - guaranteed arrival
  - no error
  - in order delivery
- A correct protocol: stop-and-wait
  - one packet at a time
- An efficient protocol: sliding window
  - multiple packets simultaneously

TCP: Overview
- point-to-point:
  - one sender, one receiver
- connection-oriented:
  - handshaking (exchange of control msgs) to initialize sender, receiver state before data exchange
- pipelined:
  - multiple in-flight segments
- full duplex data:
  - bi-directional data flow in same connection
- reliable data transfer:
  - guaranteed arrival, no error, in order
- flow controlled:
  - sender does not overwhelm receiver
- congestion controlled:
  - sender does not overwhelm the network
- no delay or bandwidth guarantee.

TCP Segment Structure
- source port #
- dest port #
- sequence number
- acknowledgement number
- URG: urgent data (generally not used)
- ACK: ACK # (valid)
- PSH: push data now (generally not used)
- RST, SYN, FIN: connection estab (setup, teardown commands)
- Internet checksum (as in UDP)
- options (variable length)
- application data (variable length)
- receive window
- Urg data ptr
- Urg data pnter
- TCP header size?
- counting by bytes of data (not segments!)
- # bytes rcvr willing to accept
- # bytes of data of segments!
TCP Reliable Data Transfer

- TCP provides reliable data transfer service on top of IP’s unreliable service
- Pipelined transmissions
- Cumulative ACKs
- Single retransmission timer
- When the receiver receives out-of-order segments, it buffers them and re-ACK the last in-order data
- The sender retransmits at timeout or receiving duplicate ACKs
- Somewhere between Go-back-N and Selective Repeat.

TCP Timeout

**Q**: how to set transmission timeout value?
- longer than normal RTT (round trip time)
  - but RTT varies
- too short: premature timeout and unnecessary retransmissions
- too long: slow reaction to segment loss

Estimating Round Trip Time

**Q**: how to estimate RTT?
- $\text{SampleRTT}$: measured time from segment transmission until ACK receipt
- $\text{SampleRTT}$ fluctuates, we want estimated RTT "smoother" to avoid short-term spikes
  - average several recent measurements, not just current $\text{SampleRTT}$
  - we also want to give more recent measurements higher weight in case things do change

**EWMA – Exponentially Weighted Moving Average**

\[
\text{EstimatedRTT} = \frac{\text{SampleRTT}_1 + \alpha \text{SampleRTT}_2 + \alpha^2 \text{SampleRTT}_3 + \ldots}{1 + \alpha + \alpha^2 + \ldots}
\]

- SampleRTT$_1$ is RTT for the most recent data segment,
- SampleRTT$_2$ is RTT for the next recent data segment, etc.
- $\Rightarrow$ EstimatedRTT = (1-$\alpha$) * EstimatedRTT$_{last}$ + $\alpha$ * SampleRTT

- influence of past sample decreases exponentially fast
- typical value: $\alpha = 0.875$
Example RTT Estimation

TCP Timeout

Setting the timeout:
- EstimatedRTT plus "safety margin"
- large variation in EstimatedRTT → larger safety margin
- we need to estimate of how much SampleRTT deviates from EstimatedRTT (EWMA):

\[ \text{DevRTT} = (1-\beta) \times \text{DevRTT}_{last} + \beta \times |\text{SampleRTT} - \text{EstimatedRTT}| \]

(typically, \( \beta = 0.25 \))

Then set timeout interval:
\[ \text{TimeoutInterval} = \text{EstimatedRTT} + 4 \times \text{DevRTT} \]

TCP Sender Events and Processing

Data ready to send:
- 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 un-ACKed segment)
- timeout value: we just decided it!

Timeout:
- retransmit segment that caused timeout
- restart timer

ACK rcvd:
- slide sender window if acknowledges previously unacked segments
- retransmit if 3 duplicate ACKs

TCP byte-oriented 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

ACKs piggybacked in data segments in the other direction
TCP in Action: Cumulative ACK

Host A
Seq=92, 8 bytes data
Seq=100, 20 bytes data
ACK=100

Host B

X
loss

Cumulative ACK scenario

Fast Retransmission

- Time-out period often relatively long:
  - long delay before resending lost packet
- When receiver receives out-of-order segments, it re-ACK the last in-order byte
- If sender receives 3 ACKs for the same data, it supposes that segment after ACKed data was lost:
  - fast retransmission: resend segment before timer expires, restart timer

TCP in Action: Duplicate ACKs and Fast Retransmission

Host A
ACK=92
Seq=92, 8 bytes data
Seq=100, 20 bytes data
Seq=108, 8 bytes data
Seq=116, 8 bytes data

Host B

ACK=92
ACK=92
ACK=92

3 duplicate ACKs
 resend 92

Cumulative ACK scenario

Outline

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- connection management
**TCP Flow Control**

- Receive side of TCP connection has a receive buffer:
  - Flow control: sender does not overflow receiver's buffer by transmitting too much, too fast.
  - Speed-matching service: matching the send rate to the receiving app's drain rate.

- App process may be slow at reading from buffer.

**TCP Flow Control: how it works?**

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

**Where is RcvWindow in each segment?**

<table>
<thead>
<tr>
<th>Source port #</th>
<th>Dest port #</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sequence number</td>
<td></td>
</tr>
<tr>
<td>Acknowledgement number</td>
<td></td>
</tr>
<tr>
<td>Receive window</td>
<td></td>
</tr>
<tr>
<td>Checksum</td>
<td></td>
</tr>
<tr>
<td>Urg data pointer</td>
<td></td>
</tr>
<tr>
<td>Options (variable length)</td>
<td></td>
</tr>
<tr>
<td>Application data (variable length)</td>
<td></td>
</tr>
</tbody>
</table>

**TCP Connection Management**

- **Establishment:**
  - TCP sender, receiver establish "connection" before exchanging data segments.
  - Initialize TCP variables: starting seq. #s, MSS, buffers, flow control info (e.g., RcvWindow).
  - MSS is the maximum TCP segment size each side is willing to accept.
    - Typically the largest segment size fit into a link-layer frame.
    - What is MSS for an IPv4 host connected with Ethernet?
- **Teardown:**
  - Freeing up resources after mutually close.
TCP Connection Establishment

**Three way handshake:**

**Step 1:** client (active open) sends TCP SYN segment to server
- specifies initial seq #
- no data

**Step 2:** server (passive open) host receives SYN, replies with SYNACK segment
- server allocates buffers
- specifies server initial seq #

**Step 3:** client receives SYNACK, replies with ACK segment, which may contain data

TCP Connection Teardown

**Closing a connection:**
- close socket: `close(sockfd);`

**Step 1:** A (active closing host) sends TCP FIN control segment to server

**Step 2:** B (passive closing host) receives FIN, replies with ACK. Closes connection, sends FIN.

**Step 3:** A receives FIN, replies with ACK.
- Enters "timed wait" - resend ACK in case it is lost

**Step 4:** B receives ACK. Connection closed.

TCP State Transition Diagram

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