TCP

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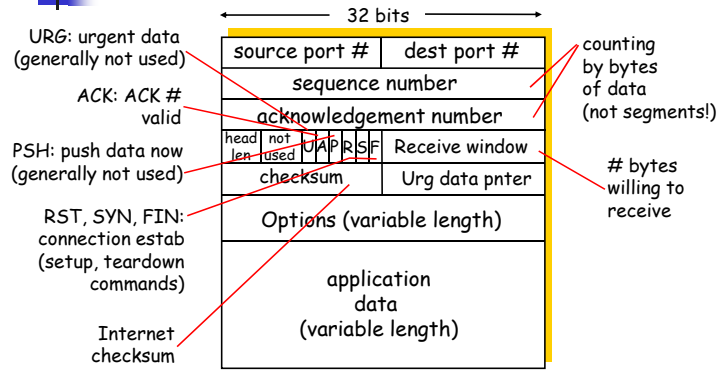
TCP: Overview

- 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 one 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



Maximum Segment Size (MSS)

- MSS is the maximum TCP segment that'd fit into the link layer frame
- Local MSS
- Path MSS
  - Try and error probing

TCP Reliable Data Transfer

- TCP provides reliable data transfer service on top of IP's unreliable service
  - Pipelined transmissions
  - Cumulative ACKs
  - When the receiver receives out-of-order segments, it buffers them and re-ACKs the last in-order data
  - Retransmit a single segment at each timeout
  - The sender retransmits at timeout or receiving duplicate ACKs
- Somewhere between Go-back-N and Selective Repeat, with some additional twists.

TCP Timeout

Q: principles for setting transmission timeout value?

- too short: premature timeout and unnecessary retransmissions
- too long: slow reaction to segment loss
- longer than normal RTT (round trip time)
  - one challenge is that RTT varies

Estimating Round Trip Time

Q: how to estimate RTT?

- Basic measurement: measured time from segment transmission until ACK receipt
- Stability: RTT fluctuates, we want to avoid instability (pre-mature reaction to short-term spikes)
  - average several recent measurements, not just current RTT
- Agility: in case things do change, we want to adjust quickly
  - give more recent measurements higher weight

EWMA – Exponentially Weighted Moving Average

- influence of past sample decreases exponentially fast

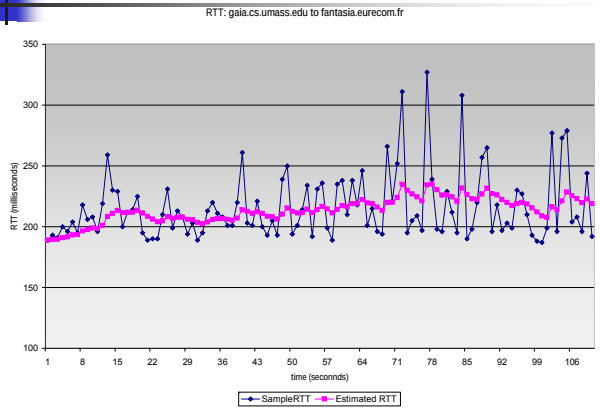
$$\text{EstimatedRTT} = \frac{\text{SampleRTT}_1 + \alpha*\text{SampleRTT}_2 + \alpha^2*\text{SampleRTT}_3 + \ldots\ldots}{1 + \alpha + \alpha^2 + \ldots\ldots}$$

SampleRTT_1 is RTT for the most recent data segment,
SampleRTT_2 is RTT for the next recent data segment, etc.

$\Rightarrow \text{EstimatedRTT} = \alpha*\text{EstimatedRTT}_{\text{last}} + (1-\alpha)*\text{SampleRTT}_1$

- typical value: $\alpha = 0.875$

Example RTT Estimation



TCP Timeout

Setting the timeout:

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

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

(typically, β = 0.75)

Then set timeout interval:

TimeoutInterval = EstimatedRTT + 4*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
- timeout value: we just decided it!!

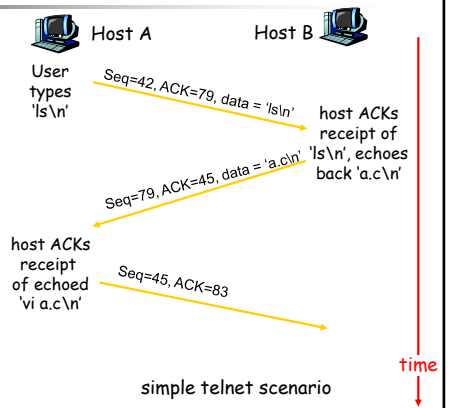
Timeout:
- retransmit segment that caused timeout
- restart timer

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

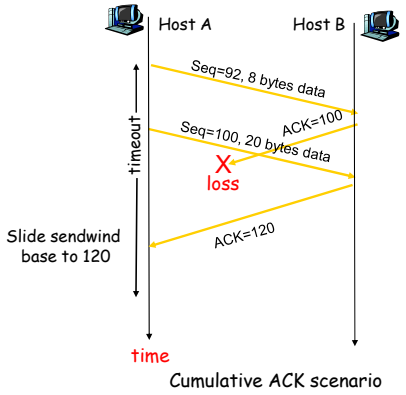
TCP byte-oriented seq. #'s and ACKs

Seq. #'s:
- byte stream "number" of first byte in segment's data

ACKs:
- seq # of next byte expected from other side
- cumulative ACK
- piggybacked in data segments in the reverse direction



TCP in Action: Cumulative ACK

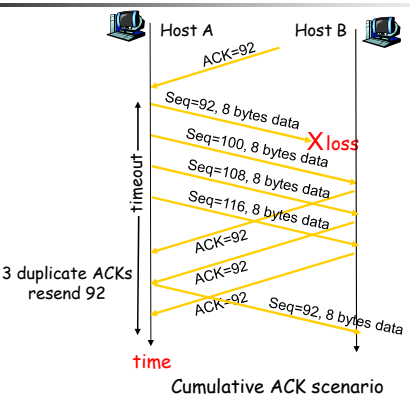


Cumulative ACK scenario

Fast Retransmission

- Time-out period often relatively long:
  - long delay before resending lost packet
- When receiver receives out-of-order segments, it re-ACKs 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



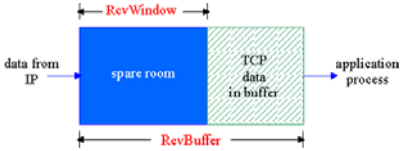
Cumulative ACK scenario

Outline

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

TCP Flow Control

- receive side of TCP connection has a receive buffer:

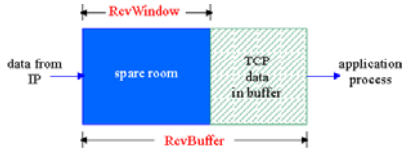


- app process may be slow at reading from buffer

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

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

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`**)
- Teardown:
  - freeing up resources after mutually close

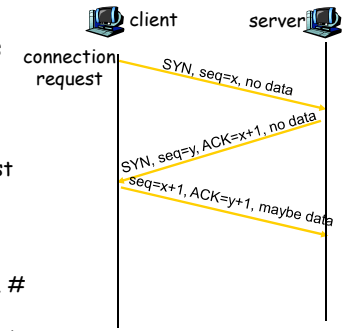
TCP Connection Establishment

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);**

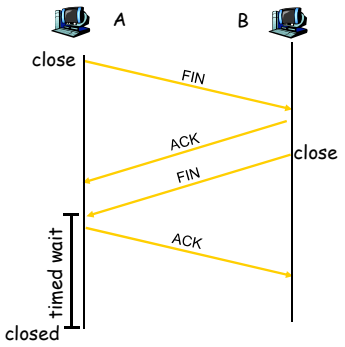
Step 1: A (active closing host) sends TCP FIN control segment to server

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

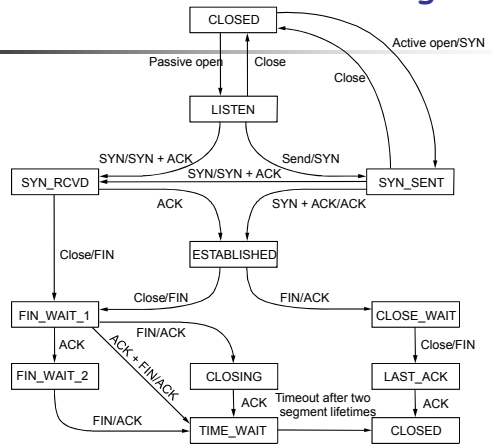
Step 3: A receives FIN, replies with ACK.

- 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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