Chapter 3
Transport Layer

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Chapter 3 outline

3.1 Transport-layer services
3.2 Multiplexing and demultiplexing
3.3 Connectionless transport: UDP
3.4 Principles of reliable data transfer
   - Bit error: Ack, seq.#
   - Loss: Time out
   - Pipelining
   - Selective Repeat
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
Selective Repeat

- receiver *individually* acknowledges all correctly received pkts
  - buffers pkts, as needed, for eventual in-order delivery to upper layer
- sender only resends pkts for which ACK not received
  - sender timer for each unACKed pkt
- sender window
  - N consecutive seq #’s
  - again limits seq #s of sent, unACK’ed pkts
Selective repeat: sender, receiver windows

(a) sender view of sequence numbers

(b) receiver view of sequence numbers
Selective repeat

**sender**

- data from above:
  - if next available seq # in window, send pkt

**timeout(n):**
- resend pkt n, restart timer

**ACK(n) in [sendbase, sendbase+N]:**
- mark pkt n as received
- if n smallest unACKed pkt, advance window base to next unACKed seq #

**receiver**

- pkt n in [rcvbase, rcvbase+N-1]
  - send ACK(n)
  - out-of-order: buffer
  - in-order: deliver (also deliver buffered, in-order pkts), advance window to next not-yet-received pkt

- pkt n in [rcvbase-N, rcvbase-1]
  - ACK(n)

- otherwise:
  - ignore
Selective repeat in action
Selective repeat: dilemma

Example:
- seq #’s: 0, 1, 2, 3
- window size=3

- receiver sees no difference in two scenarios!
- incorrectly passes duplicate data as new in (a)

Q: what relationship between seq # size and window size?
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TCP: Overview

- **point-to-point:**
  - one sender, one receiver

- **reliable, in-order byte steam:**
  - 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) inits sender, receiver state before data exchange

- **flow controlled:**
  - sender will not overwhelm receiver
TCP segment structure

- Source port #
- Dest port #
- Sequence number
- Acknowledgement number
- Receive window
- Checksum
- Urg data pointer
- Options (variable length)
- Application data (variable length)

- 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)
- Internet checksum (as in UDP)
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

User types ‘C’
Seq=42, ACK=79, data = ‘C’
Seq=79, ACK=43, data = ‘C’
Seq=43, ACK=80

Host A

Host B

host ACKs receipt of ‘C’, echoes back ‘C’
host ACKs receipt of echoed ‘C’
simple telnet scenario

Transport Layer 3-11
TCP Round Trip Time and Timeout

Q: how to set TCP timeout value?
  ❖ longer than RTT
    ▪ but RTT varies
  ❖ 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
  ❖ SampleRTT will vary, want estimated RTT “smoother”
    ▪ average several recent measurements, not just current SampleRTT
TCP Round Trip Time and Timeout

EstimatedRTT = (1− α)*EstimatedRTT + α*SampleRTT

- Exponential weighted moving average
- influence of past sample decreases exponentially fast
- typical value: α = 0.125
Example RTT estimation:

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

![Graph showing RTT estimation over time]
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:

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

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

Then set timeout interval:

\[
\text{TimeoutInterval} = \text{EstimatedRTT} + 4 \times \text{DevRTT}
\]