Computer Network Applications Lecture 9

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Chapter 3 outline

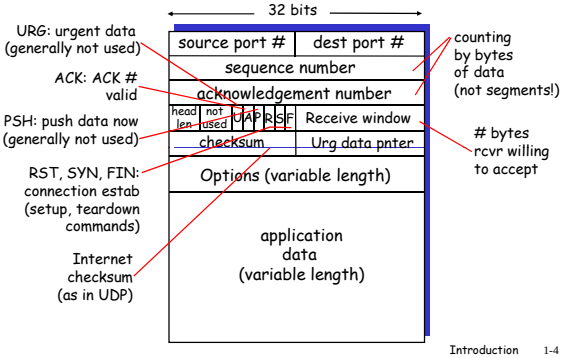
- 3.1 Transport-layer services
- 3.2 Multiplexing and demultiplexing
- 3.3 Connectionless transport: UDP
- 3.4 Principles of reliable data transfer
- 3.5 Connection-oriented transport: TCP
  - segment structure
  - reliable data transfer
  - flow control
  - connection management
- 3.6 Principles of congestion control
- 3.7 TCP congestion control

TCP: Overview

RFCs: 793, 1122, 1323, 2018, 2581

- 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) init's sender, receiver state before data exchange
- flow controlled:
  - sender will not overwhelm receiver

TCP segment structure

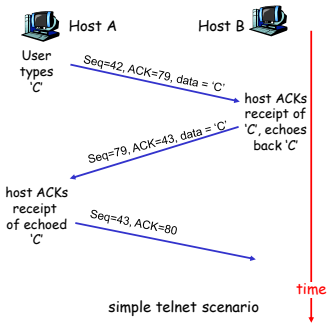


TCP seq. #'s and ACKs

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

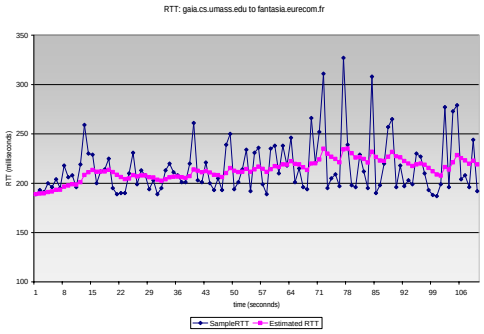


TCP Round Trip Time and Timeout

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

- Exponential weighted moving average
- influence of past sample decreases exponentially fast
- typical value: $\alpha = 0.125$

Example RTT estimation:



TCP Round Trip Time and Timeout

Setting the timeout

- **EstimtedRTT** plus "safety margin"
  - large variation in **EstimatedRTT** -> larger safety margin
- first estimate of how much SampleRTT deviates from EstimatedRTT:

```
DevRTT = (1-β)*DevRTT +
          β*|SampleRTT-EstimatedRTT|

(typically, β = 0.25)
```

Then set timeout interval:

```
TimeoutInterval = EstimatedRTT + 4*DevRTT
```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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

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`

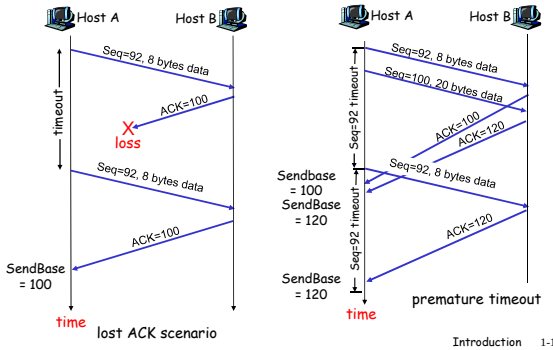
timeout:

- retransmit segment that caused timeout
- restart timer

Ack rcvd:

- If acknowledges previously unacked segments
  - update what is known to be acked
  - start timer if there are outstanding segments

TCP: retransmission scenarios



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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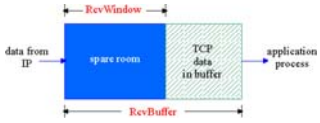
TCP Flow Control

- receive side of TCP connection has a receive buffer:

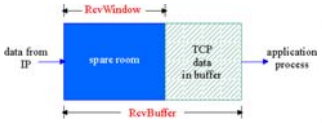


- app process may be slow at reading from buffer

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

- speed-matching service: matching the send rate to the receiving app's drain rate

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 Connection Management

Recall: TCP sender, receiver establish "connection" before exchanging data segments

- initialize TCP variables:
  - seq. #s
  - buffers, flow control info (e.g. **RcvWindow**)
- *client:* connection initiator

```
Socket clientSocket = new
Socket("hostname","port
number");
```

- *server:* contacted by client

```
Socket connectionSocket =
welcomeSocket.accept();
```

SYN flood attack!!!

Three way handshake:

Step 1: client host sends TCP SYN segment to server
- specifies initial seq #
- no data

Step 2: server 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

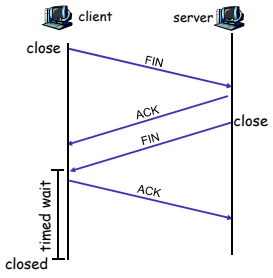
TCP Connection Management (cont.)

Closing a connection:

client closes socket:
`clientSocket.close();`

Step 1: client end system sends TCP FIN control segment to server

Step 2: server receives FIN, replies with ACK. Closes connection, sends FIN.

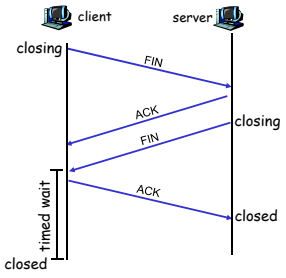


TCP Connection Management (cont.)

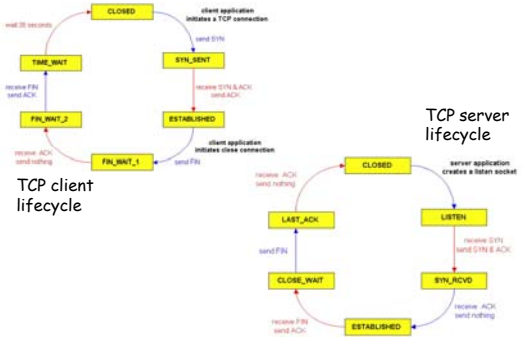
Step 3: client receives FIN, replies with ACK.
- Enters "timed wait" - will respond with ACK to received FINs

Step 4: server, receives ACK. Connection closed.

Note: with small modification, can handle simultaneous FINs.



TCP Connection Management (cont)



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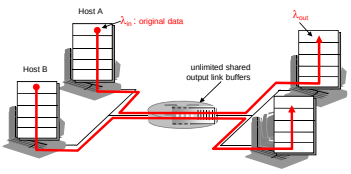
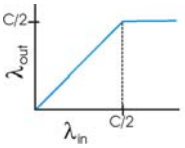
Principles of Congestion Control

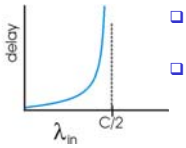
Congestion:
- informally: "too many sources sending too much data too fast for *network* to handle"
- different from flow control!
- manifestations:
  - lost packets (buffer overflow at routers)
  - long delays (queueing in router buffers)
- a top-10 problem!

Causes/costs of congestion: scenario 1

- two senders, two receivers
- one router, infinite buffers
- no retransmission



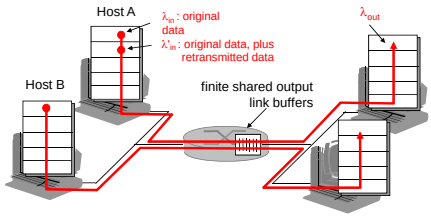




- large delays when congested
- maximum achievable throughput

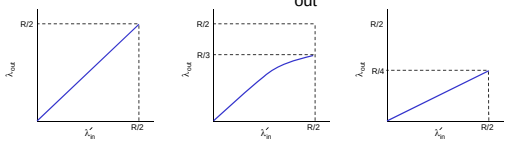
Causes/costs of congestion: scenario 2

- one router, *finite* buffers
- sender retransmission of lost packet



Causes/costs of congestion: scenario 2

- always: $\lambda_{in} = \lambda_{out}$ (goodput)
- "perfect" retransmission only when loss: $\lambda'_{in} > \lambda_{out}$
- retransmission of delayed (not lost) packet makes λ'_{in} larger (than perfect case) for same λ_{out}



a. b. c.

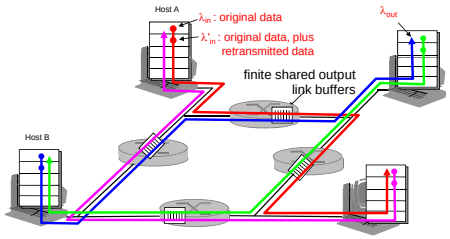
"costs" of congestion:

- more work (retrans) for given "goodput"
- unneeded retransmissions: link carries multiple copies of pkt

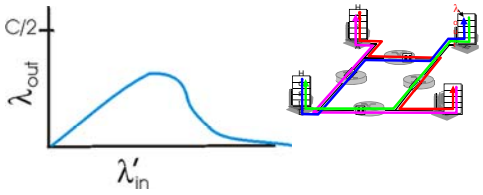
Causes/costs of congestion: scenario 3

- four senders
- multihop paths
- timeout/retransmit

Q: what happens as λ_{in} and λ'_{in} increase ?



Causes/costs of congestion: scenario 3



Another "cost" of congestion:

- when packet dropped, any "upstream transmission capacity used for that packet was wasted!

Approaches towards congestion control

Two broad approaches towards congestion control:

End-end congestion control:

- no explicit feedback from network
- congestion inferred from end-system observed loss, delay
- approach taken by TCP

Network-assisted congestion control:

- routers provide feedback to end systems
  - single bit indicating congestion (SNA, DECbit, TCP/IP ECN, ATM)
  - explicit rate sender should send at

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TCP Congestion Control

- end-end control (no network assistance)
- sender limits transmission: `LastByteSent-LastByteAcked ≤ CongWin`
- Roughly,

$$\text{rate} = \frac{\text{CongWin}}{\text{RTT}} \text{ Bytes/sec}$$

- **CongWin** is dynamic, function of perceived network congestion

How does sender perceive congestion?

- loss event = timeout *or* 3 duplicate acks
- TCP sender reduces rate (**CongWin**) after loss event

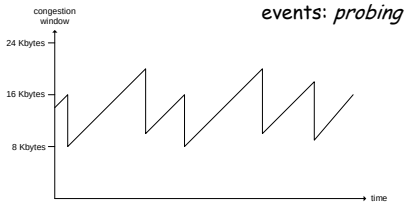
three mechanisms:

- AIMD
- slow start
- conservative after timeout events

TCP AIMD

multiplicative decrease: cut **CongWin** in half after loss event

additive increase: increase **CongWin** by 1 MSS every RTT in the absence of loss events: *probing*



Long-lived TCP connection

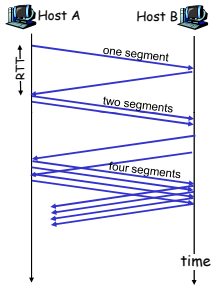
TCP Slow Start

- When connection begins, **CongWin** = 1 MSS
  - Example: MSS = 500 bytes & RTT = 200 msec
  - initial rate = 20 kbps
- available bandwidth may be >> MSS/RTT
  - desirable to quickly ramp up to respectable rate

- When connection begins, increase rate exponentially fast until first loss event

TCP Slow Start (more)

- When connection begins, increase rate exponentially until first loss event:
  - double **CongWin** every RTT
  - done by incrementing **CongWin** for every ACK received
- Summary: initial rate is slow but ramps up exponentially fast



Refinement

- After 3 dup ACKs:
  - **CongWin** is cut in half
  - window then grows linearly
- But after timeout event:
  - **CongWin** instead set to 1 MSS;
  - window then grows exponentially
  - to a threshold, then grows linearly

Philosophy:

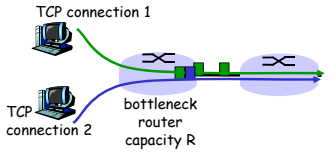
• 3 dup ACKs indicates network capable of delivering some segments
• timeout before 3 dup ACKs is "more alarming"

Summary: TCP Congestion Control

- When **CongWin** is below **Threshold**, sender in slow-start phase, window grows exponentially.
- When **CongWin** is above **Threshold**, sender is in congestion-avoidance phase, window grows linearly.
- When a triple duplicate ACK occurs, **Threshold** set to **CongWin/2** and **CongWin** set to **Threshold**.
- When timeout occurs, **Threshold** set to **CongWin/2** and **CongWin** is set to 1 MSS.

TCP Fairness

Fairness goal: if K TCP sessions share same bottleneck link of bandwidth R, each should have average rate of R/K



Fairness (more)

Fairness and UDP

- Multimedia apps often do not use TCP
  - do not want rate throttled by congestion control
- Instead use UDP:
  - pump audio/video at constant rate, tolerate packet loss
- Research area: TCP friendly

Fairness and parallel TCP connections

- nothing prevents app from opening parallel connections between 2 hosts.
- Web browsers do this
- Example: link of rate R supporting 9 connections;
  - new app asks for 1 TCP, gets rate R/10
  - new app asks for 11 TCPs, gets R/2 !

Chapter 3: Summary

- principles behind transport layer services:
  - multiplexing, demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- instantiation and implementation in the Internet
  - UDP
  - TCP

Next:

- leaving the network "edge" (application, transport layers)
- into the network "core"