ECSE-4670: Computer Communication Networks (CCN)

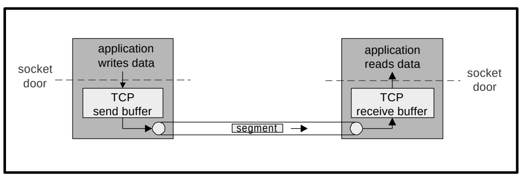
Chapter 3: TCP

Shivkumar Kalyanaraman: shivkuma@ecse.rpi.edu
Biplab Sikdar: sikdab@rpi.edu

TCP: Overview

- **Point-to-point:**
  - one sender, one receiver
- **Reliable, in-order *byte steam:***
  - no "message boundaries"
  - But TCP chops it up into segments for transmission internally
- **Pipelined (window) flow control:**
  - Window size decided by receiver and network
- ***Send & receive buffers***

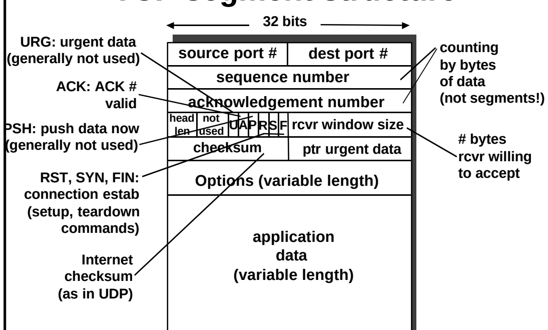
TCP: Overview



TCP: Overview

- **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 & Congestion Control:**
  - sender will not overwhelm receiver or the network

TCP segment structure



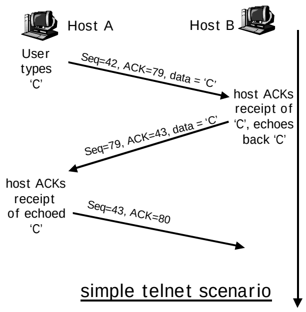
TCP seq. #'s and ACKs (I)

Sequence Numbers:
- byte stream "number" of first byte in segment's data

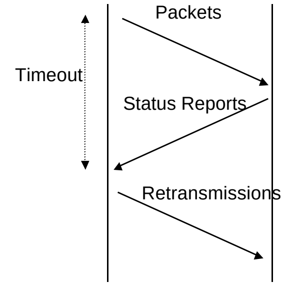
ACKs:
- seq # of next byte expected from other side
- cumulative ACK

Q: how receiver handles out-of-order segments
- A: TCP spec doesn't say, - up to implementor

TCP Seq. #'s and ACKs (II)



Temporal Redundancy Model



- Sequence Numbers
- CRC or Checksum

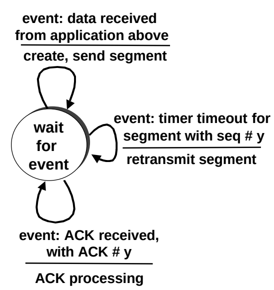
- ACKs
- NAKs,
- SACKs
- Bitmaps

- Packets
- FEC information

Status Report Design

- ***Cumulative*** **acks:**
  - ***Robust*** **to losses on the reverse channel**
  - **Can work with *go-back-N* retransmission**
  - **Cannot pinpoint *blocks* of data which are lost**
    - **The first lost packet can be pinpointed because the receiver would generate duplicate acks**

TCP: reliable data transfer (I)



- **one way data transfer**
- **no flow, congestion control**

TCP: reliable data transfer (II)

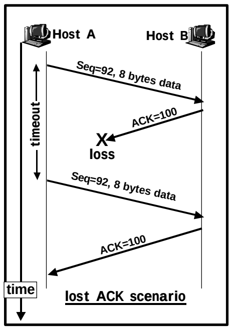
Simplified TCP sender

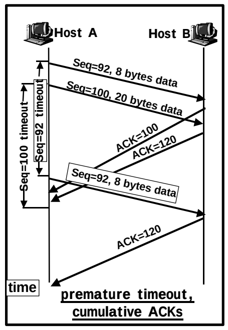
```
sendbase = initial_sequence number
 nextseqnum = initial_sequence number

 loop (forever) {
   switch(event)
   event: data received from application above
        create TCP segment with sequence number nextseqnum
        start timer for segment nextseqnum
        pass segment to IP
        nextseqnum = nextseqnum + length(data)
    event: timer timeout for segment with sequence number y
        retransmit segment with sequence number y
        compute new timeout interval for segment y
        restart timer for sequence number y
    event: ACK received, with ACK field value of y
        if (y > sendbase) { /* cumulative ACK of all data up to y */
            cancel all timers for segments with sequence numbers < y
            sendbase = y
            }
        else { /* a duplicate ACK for already ACKed segment */
            increment number of duplicate ACKs received for y
             if (number of duplicate ACKS received for y == 3) {
                /* TCP fast retransmit */
               resend segment with sequence number y
               restart timer for segment y
            }
   }  /* end of loop forever */
```


TCP ACK generation

Event	TCP Receiver action
in-order segment arrival, no gaps, everything else already ACKed	delayed ACK. Wait up to 500ms for next segment. If no next segment, send ACK
in-order segment arrival, no gaps, one delayed ACK pending	immediately send single cumulative ACK
out-of-order segment arrival higher-than-expect seq. # gap detected!	send duplicate ACK, indicating seq. # of next expected byte
arrival of segment that partially or completely fills gap	immediate ACK if segment starts at lower end of gap

TCP: retransmission scenarios



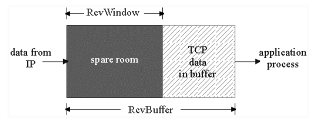


TCP Flow Control

flow control
sender won't overrun receiver's buffers by transmitting too much, too fast

RcvBuffer = size or TCP Receive Buffer

RcvWindow = amount of spare room in Buffer



receiver buffering

receiver: explicitly informs sender of free buffer space
- RcvWindow field in TCP segment

sender: keeps the amount of transmitted, unACKed data less than most recently received RcvWindow

Timeout and RTT Estimation

- Timeout: for robust detection of packet loss
- Problem: How long should timeout be ?
  - Too long => underutilization
  - Too short => wasteful retransmissions
  - Solution: ***adaptive timeout:*** *based on estimate of max RTT*

How to estimate max RTT?

- RTT = prop + queuing delay
  - Queuing delay highly variable
  - So, different samples of RTTs will give different random values of queuing delay
- Chebyshev's Theorem:
  - MaxRTT = Avg RTT + k*Deviation
  - Error probability is less than 1/(k**2)
  - Result true for ANY distribution of samples

Round Trip Time and Timeout (II)

Q: how to estimate RTT?
- SampleRTT: measured time from segment transmission until ACK receipt
  - ignore retransmissions, cumulatively ACKed segments
- SampleRTT will vary wildly => want estimated RTT "smoother"
  - use several recent measurements, not just current SampleRTT to calculate "AverageRTT"

TCP Round Trip Time and Timeout (III)

AverageRTT = (1-x)*AverageRTT + x*SampleRTT

- Exponential weighted moving average (EWMA)
- influence of given sample decreases exponentially fast; x = 0.1

Setting the timeout

- AverageRTT plus "safety margin" proportional to variation

Timeout = AverageRTT + 4*Deviation

Deviation = (1-x)*Deviation + x*|SampleRTT- AverageRTT|

TCP Connection Management - 1

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();

Rensselaer Polytechnic Institute © Shivkumar Kalvanaraman & © Biplab Sikdar 19

TCP Connection Management - 2

Three way handshake:

Step 1: client end system sends TCP SYN control segment to server
- specifies initial seq #

Step 2: server end system receives SYN, replies with SYNACK control segment
- ACKs received SYN
- allocates buffers
- specifies server-> receiver initial seq. #

Rensselaer Polytechnic Institute © Shivkumar Kalvanaraman & © Biplab Sikdar 20

TCP Connection Management - 3

Closing a connection:

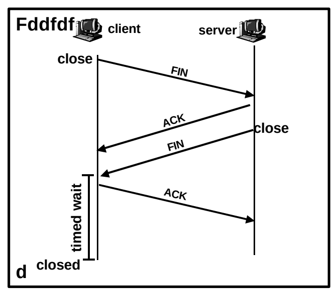
client closes socket: clientSocket.close();

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

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

Rensselaer Polytechnic Institute © Shivkumar Kalvanaraman & © Biplab Sikdar 21

TCP Connection Management - 4



Rensselaer Polytechnic Institute © Shivkumar Kalvanaraman & © Biplab Sikdar 22

TCP Connection Management - 5

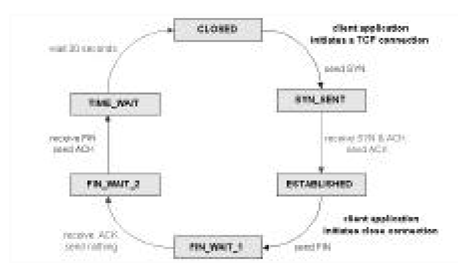
Step 3: client receives FIN, replies with ACK.
- Enters "timed wait" - will respond with ACK to received FINs

Step 4: server, receives ACK. Connection closed.

Note: with small modification, can handle simultaneous FINs.

Rensselaer Polytechnic Institute © Shivkumar Kalvanaraman & © Biplab Sikdar 23

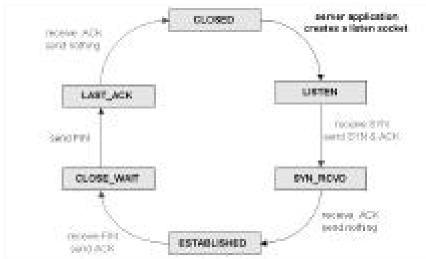
TCP Connection Management - 6



TCP client lifecycle

Rensselaer Polytechnic Institute © Shivkumar Kalvanaraman & © Biplab Sikdar 24

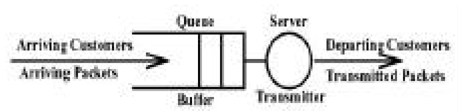
TCP Connection Management - 7



TCP server lifecycle

Recap: Stability of a Multiplexed System

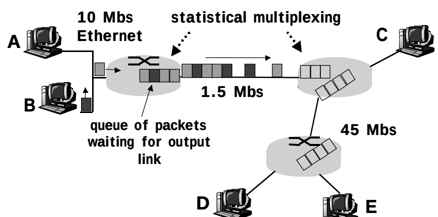
Average Input Rate > Average Output Rate
=> system is unstable!



How to ensure stability ?

1. Reserve enough capacity so that demand is less than reserved capacity
2. Dynamically detect overload and adapt either the demand or capacity to resolve overload

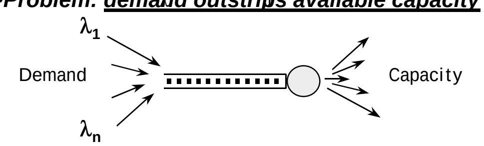
Congestion Problem in Packet Switching



- Cost: self-descriptive header per-packet, buffering and delays for applications.
- Need to *either reserve resources or dynamically detect/adapt to overload for stability*

The Congestion Problem

•*Problem: demand outstrips available capacity*



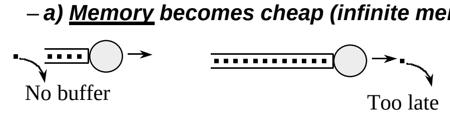
- If information about λ_i , λ and μ is known in a central location where control of λ_i or μ can be effected with zero time delays,
  - the congestion problem is solved!

The Congestion Problem (Continued)

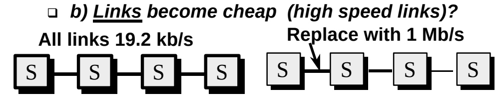
- Problems:
  - Incomplete information (eg: loss indications)
  - Distributed solution required
  - Congestion and control/measurement locations different
  - Time-varying, heterogeneous time-delay

The Congestion Problem

- Static fixes *may not* solve congestion
  - *a) Memory becomes cheap (infinite memory)*



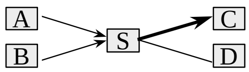
- *b) Links become cheap (high speed links)?*



File Transfer time = 5 mins *File Transfer Time = 7 hours*

The Congestion Problem (Continued)

- c) ***Processors*** become cheap (fast routers & switches)



Scenario: All links 1 Gb/s.
A & B send to C
=> "high-speed" congestion!!
(lose more packets faster!)

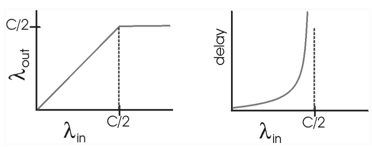
Principles of Congestion Control

Congestion:

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

Causes/costs of congestion: scenario 1

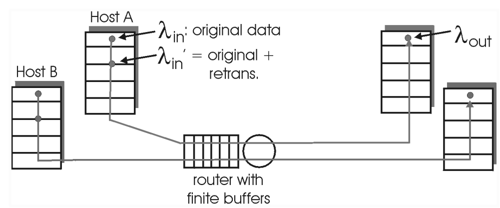
- **two senders, two receivers**
- **one router, infinite buffers**
- **no retransmission**



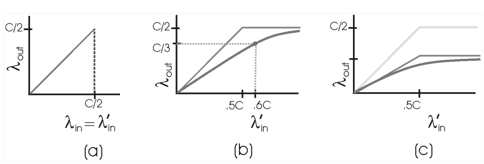
- **large delays when congested**
- **maximum achievable throughput**

Causes/costs of congestion: scenario 2

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



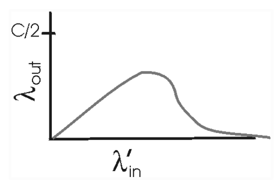
Causes/costs of congestion: scenario 2 (continued)

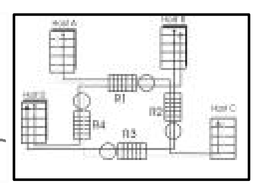


"Costs" of congestion:

- **More work (*retrans*) for given "goodput"**
- ***Unneeded retransmissions*: link carries multiple copies of pkt due to *spurious timeouts***

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 - 1

- 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**

Approaches towards congestion control - 2

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**

TCP congestion control - 1

- **end-end control (no network assistance)**
- **transmission rate limited by congestion window size, `Congwin`, over segments:**

TCP congestion control - 2

- **w segments, each with MSS bytes sent in one RTT:**

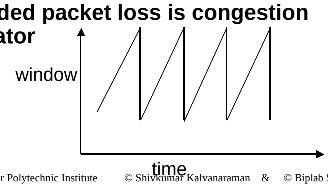
$$\text{throughput} = \frac{w * MSS}{RTT} \text{ Bytes/sec}$$

TCP congestion control - 3

- **"Probing" for usable bandwidth:**
  - **Window flow control: avoid <u>receiver</u> overrun**
  - **Dynamic window congestion control*:* avoid/control <u>network</u> overrun**
- **<u>Policy</u>*:***
  - ***Increase* Congwin until loss (congestion)**
  - **Loss => *decrease* Congwin, then begin probing (increasing) again**

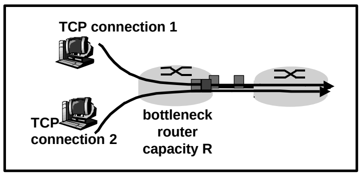
Additive Increase/Multiplicative Decrease (AIMD) Policy

- **For stability:**
  - **rate-of-decrease > rate-of-increase**
  - **Decrease performed "enough" times as long as congestion exists**
- **AIMD policy satisfies this condition, provided packet loss is congestion indicator**

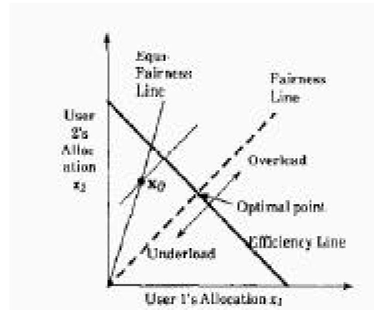


Fairness

Fairness goal: if N TCP sessions share same bottleneck link, each should get 1/N of link capacity



Fairness Analysis



AIMD Converges to Fairness

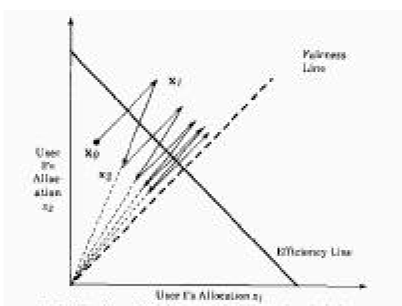


Fig. 5. Additive Increase/Multiplicative Decrease converges to the optimal point.

TCP congestion control - 4

- **TCP uses AIMD policy in "steady state"**
- **Two "phases"**
  - **Transient phase: aka "Slow start"**
  - **Steady State: aka "Congestion avoidance"**
- **Important variables:**
  - **Congwin**
  - **threshold: defines threshold between two slow start phase, congestion avoidance phase**

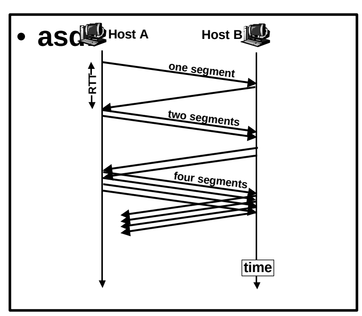
TCP Slowstart - 1

Slowstart algorithm

```
initialize: Congwin = 1
for (each segment ACKed)
     Congwin++
until (loss event OR
       CongWin > threshold)
```

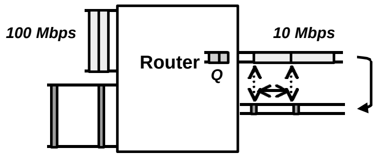
- ***Exponential increase (per RTT)* in window size (not so slow!)**
- ***Loss event:* timeout (Tahoe TCP) and/or three duplicate ACKs (Reno TCP)**

TCP Slowstart - 2



TCP Dynamics

- Rate of acks determines rate of packets : *“Self-clocking”* property.



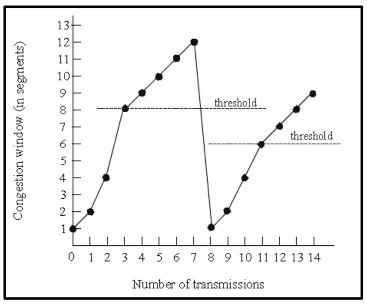
TCP Congestion Avoidance - 1

Congestion avoidance

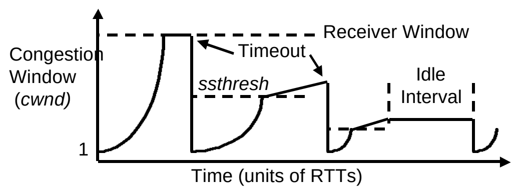
```
/* slowstart is over       */
/* Congwin > threshold */
Until (loss event) {
  every w segments ACKed:
     Congwin++
  }
threshold = Congwin/2
Congwin = 1
perform slowstart [1]
```

1: TCP Reno skips slowstart (aka *fast recovery*) after three duplicate ACKs and performs close to AIMD

TCP Congestion Avoidance - 2



TCP window dynamics (more)



TCP latency modeling - 1

<u>Q:</u> How long does it take to receive an object from a Web server after sending a request?

- **TCP connection establishment**
- **data transfer delay**

TCP latency modeling - 2

Notation, assumptions:

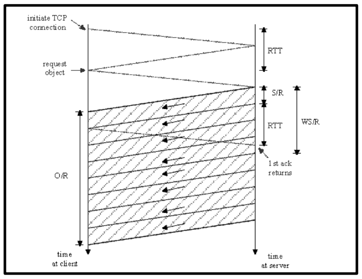
- **Assume one link between client and server of rate R**
- **Assume: fixed congestion window, W segments**
- **S: MSS (bits)**
- **O: object size (bits)**
- **no retransmissions (no loss, no corruption)**

TCP latency modeling - 3

<u>Two cases to consider:</u>

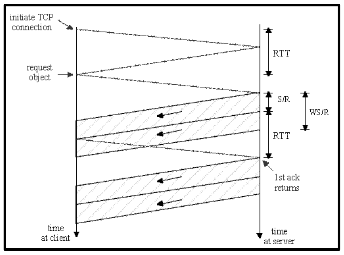
- **WS/R > RTT + S/R: ACK for first segment in window returns before window's worth of data sent**
- **WS/R < RTT + S/R: wait for ACK after sending window's worth of data sent**

TCP latency modeling - 4



Case 1: latency = 2RTT + O/R

TCP latency modeling - 5



K = O/WS

Case 2: latency = 2RTT + O/R + (K-1)[S/R + RTT - WS/R]

TCP latency modeling: slow start - 1

- **Now suppose window grows according to slow start.**
- **Will show that the latency of one object of size O is:**

where *P* is the number of times TCP stalls at server:

TCP latency modeling: slow start - 2

- where Q is the number of times the server would stall if the object were of infinite size.

- and K is the number of windows that cover the object.

TCP latency modeling: slow start - 3

Example:

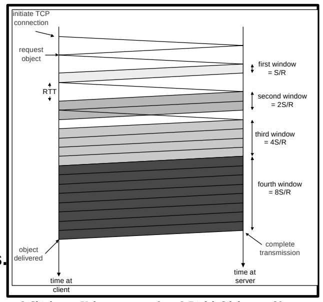
O/S = 15 segments

K = 4 windows

Q = 2

P = min{K-1,Q} = 2

Server stalls P=2 times.



TCP latency modeling: slow start - 4

Rensselaer Polytechnic Institute © Shivkumar Kalvanaraman & © Biplab Sikdar 61

TCP latency modeling: slow start - 5

Rensselaer Polytechnic Institute © Shivkumar Kalvanaraman & © Biplab Sikdar 62

Sample Results

R	O/R	P	Minimum Latency: O/R + 2 RTT	Latency with slow start
28 Kbps	28.6 sec	1	28.8 sec	28.9 sec
100 Kbps	8 sec	2	8.2 sec	8.4 sec
1 Mbps	800 msec	5	1 sec	1.5 sec
10 Mbps	80 msec	7	0.28 sec	0.98 sec

Rensselaer Polytechnic Institute © Shivkumar Kalvanaraman & © Biplab Sikdar 63

Summary: Chapter 3

- **Principles behind transport layer services:**
  - **multiplexing/demultiplexing**
  - **reliable data transfer**
  - **flow control**
  - **congestion control**
- **Instantiation and implementation in the Internet**
  - **UDP, TCP**

Rensselaer Polytechnic Institute © Shivkumar Kalvanaraman & © Biplab Sikdar 64