Internetværk og Web-programmering

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Internetværk og Web-programmering
TCP and Socket Programming

Lecture 12
Michele Albano

Distributed, Embedded, Intelligent Systems

Last lecture you saw:

- Transport Layer Protocols:
  - Reliable Data Transfer 1.0 to 3.0
  - Pipelined protocols
  - Go-back-N vs Selective Repeat
- TCP / UDP
  - Protocol header
  - 3 ways handshake
  - Multiplexing / demultiplexing

- Don’t forget to evaluate the course

Agenda

3.5 connection-oriented transport: TCP
- **segment structure**
- reliable data transfer
- flow control

3.6 principles of congestion control

3.7 TCP congestion control

2.7 Socket programming / [DF] **16.9** Javascript sockets

Transmission Control Protocol:

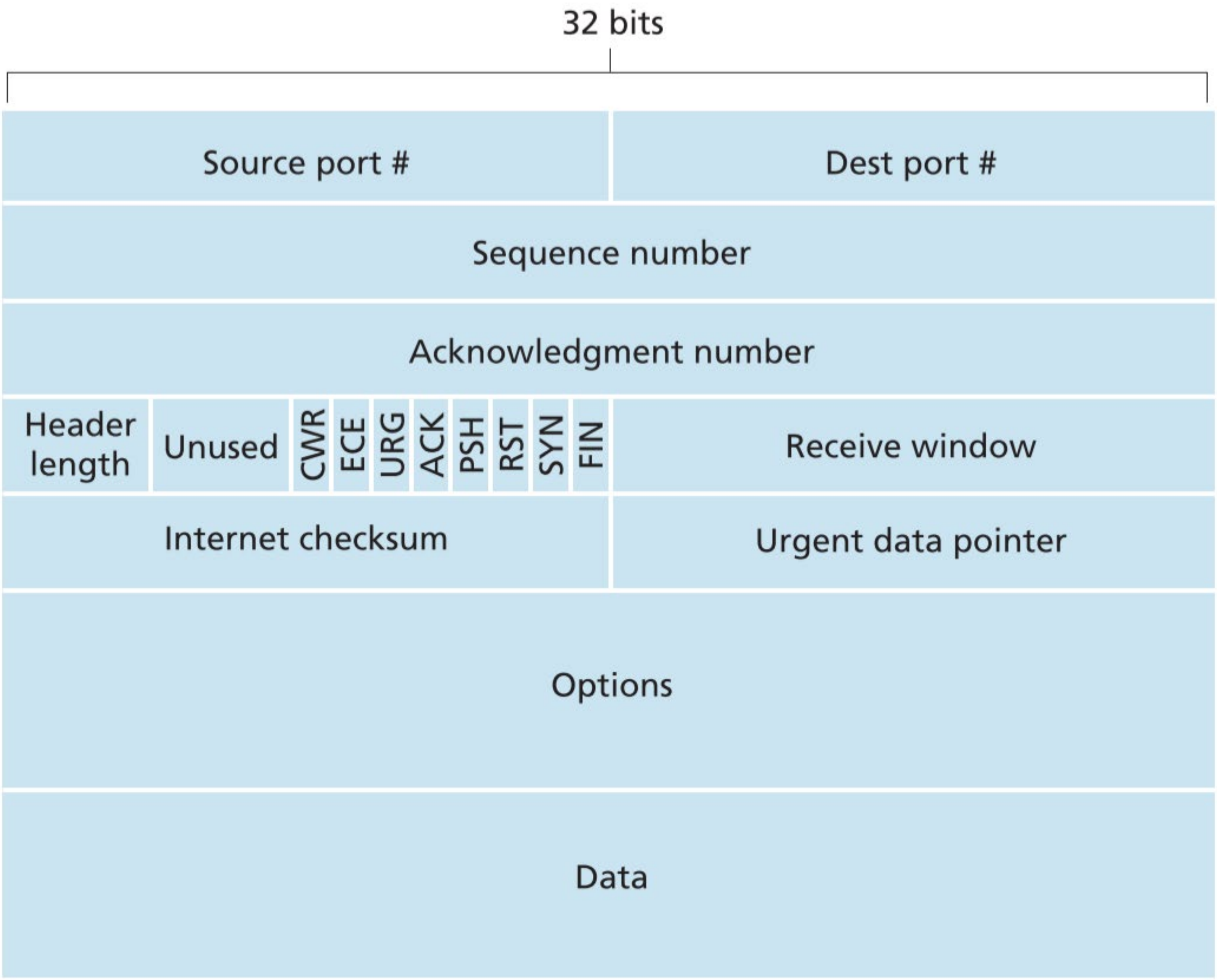
- point-to-point:
  - one sender, one receiver
- full duplex data:
  - bi-directional data flow in same connection
  - MSS: maximum segment size
- reliable, in-order byte stream:
  - no "message boundaries"
- connection-oriented:
  - handshaking (exchange of control msgs) initializes sender, receiver state before data exchange
- pipelined:
  - TCP congestion and flow control set window size
- flow controlled:
  - sender will not overwhelm receiver

TCP segment structure 1/2

- Src/dst ports
  - TCP and UDP have different namespaces
- Seq/ack numbers to the byte

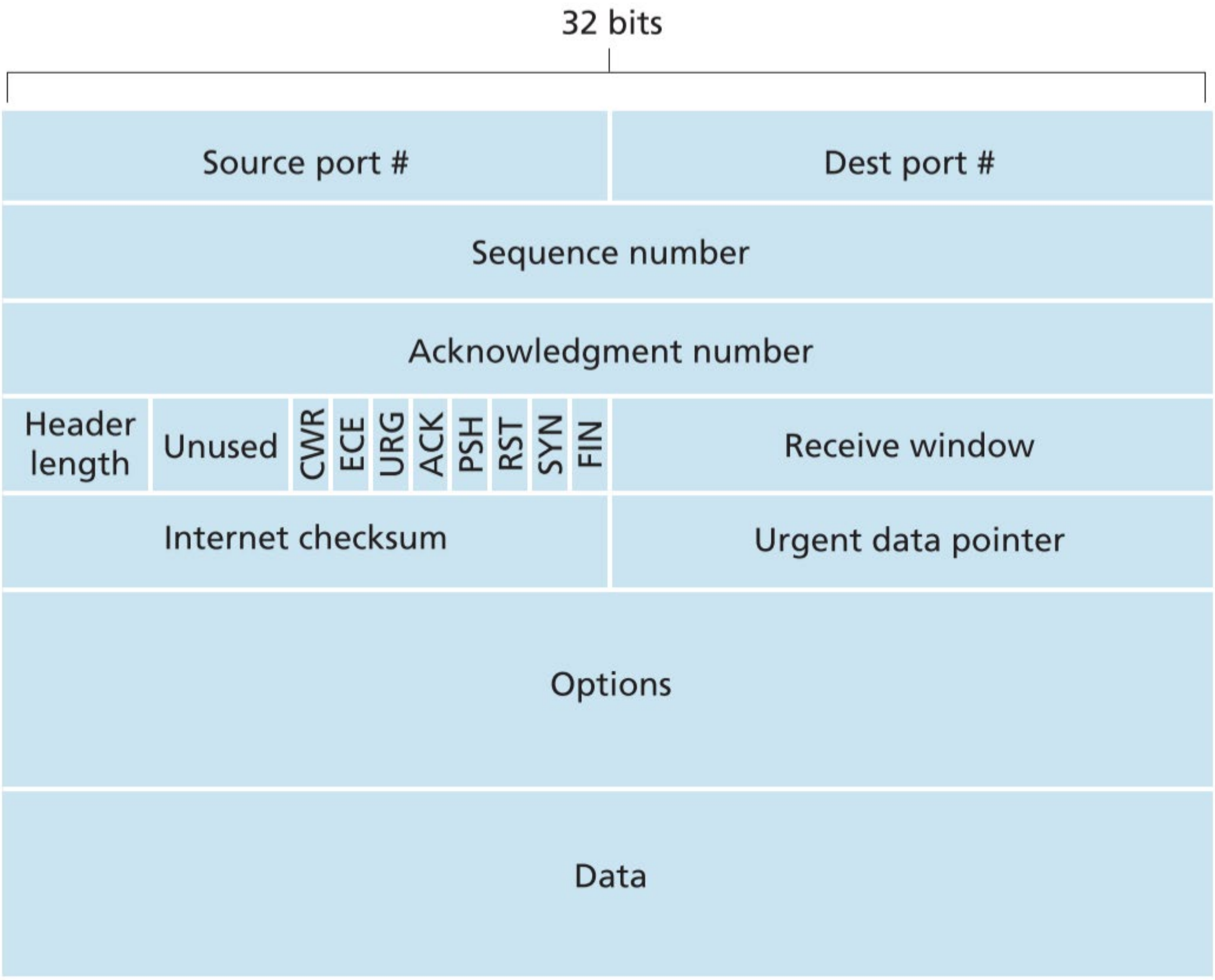
Flags:

- CWR and ECE – explicit congestion notification
- URG and PSH – urgent data
- SYN and FIN seen in lecture 11
- ACK is 1 if the ACK number is valid
- RST to kill the connection



TCP segment structure 2/2

- Header length – needed since "Options" has variable length
- Receive window: for flow control
- Internet checksum: 16-bit checksum for TCP header, payload, and some data from IP (routing layer)
- Urgent data pointer: used with the URG flag
- Options: for example to set the maximum size of the segments, or timestamp



TCP Sequence Numbers, ACKs 1/2

sequence numbers:

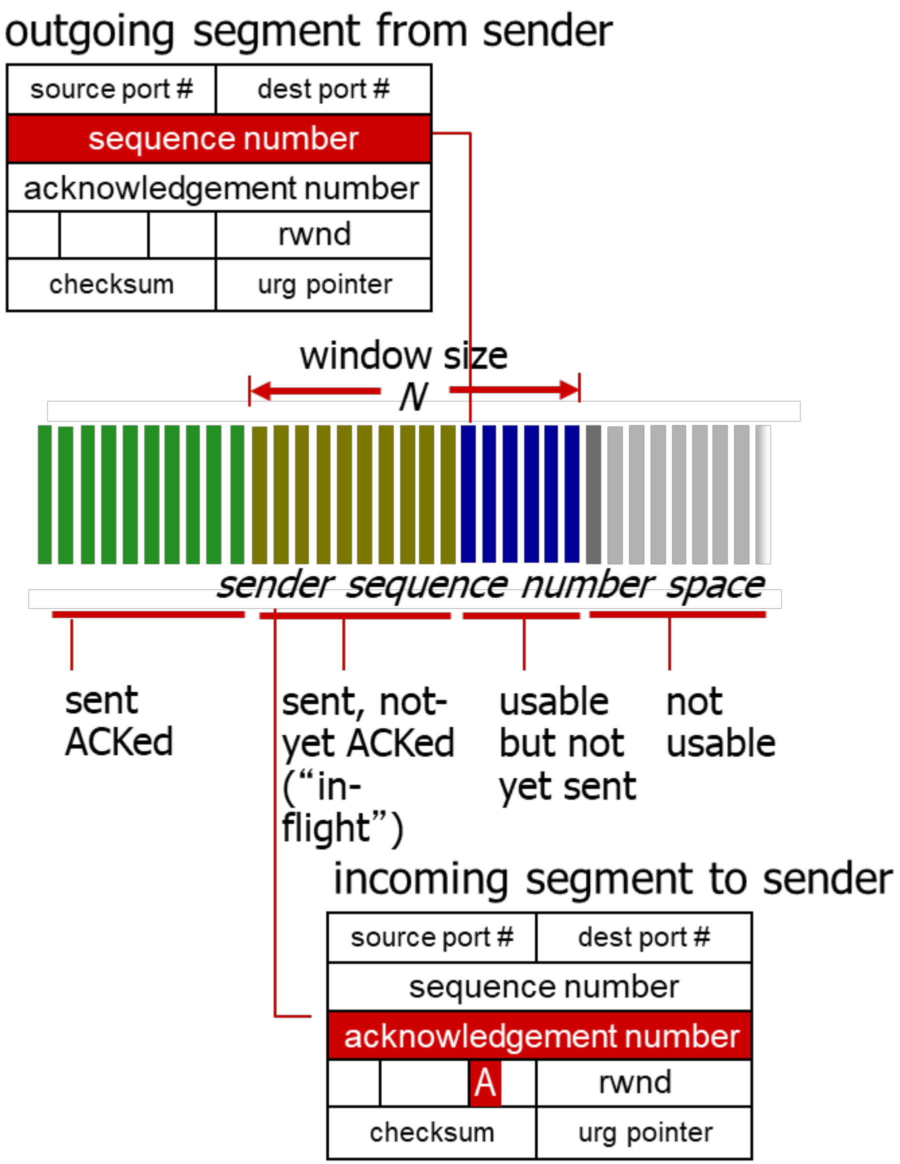
- byte stream "number" of first byte in segment's data

acknowledgements:

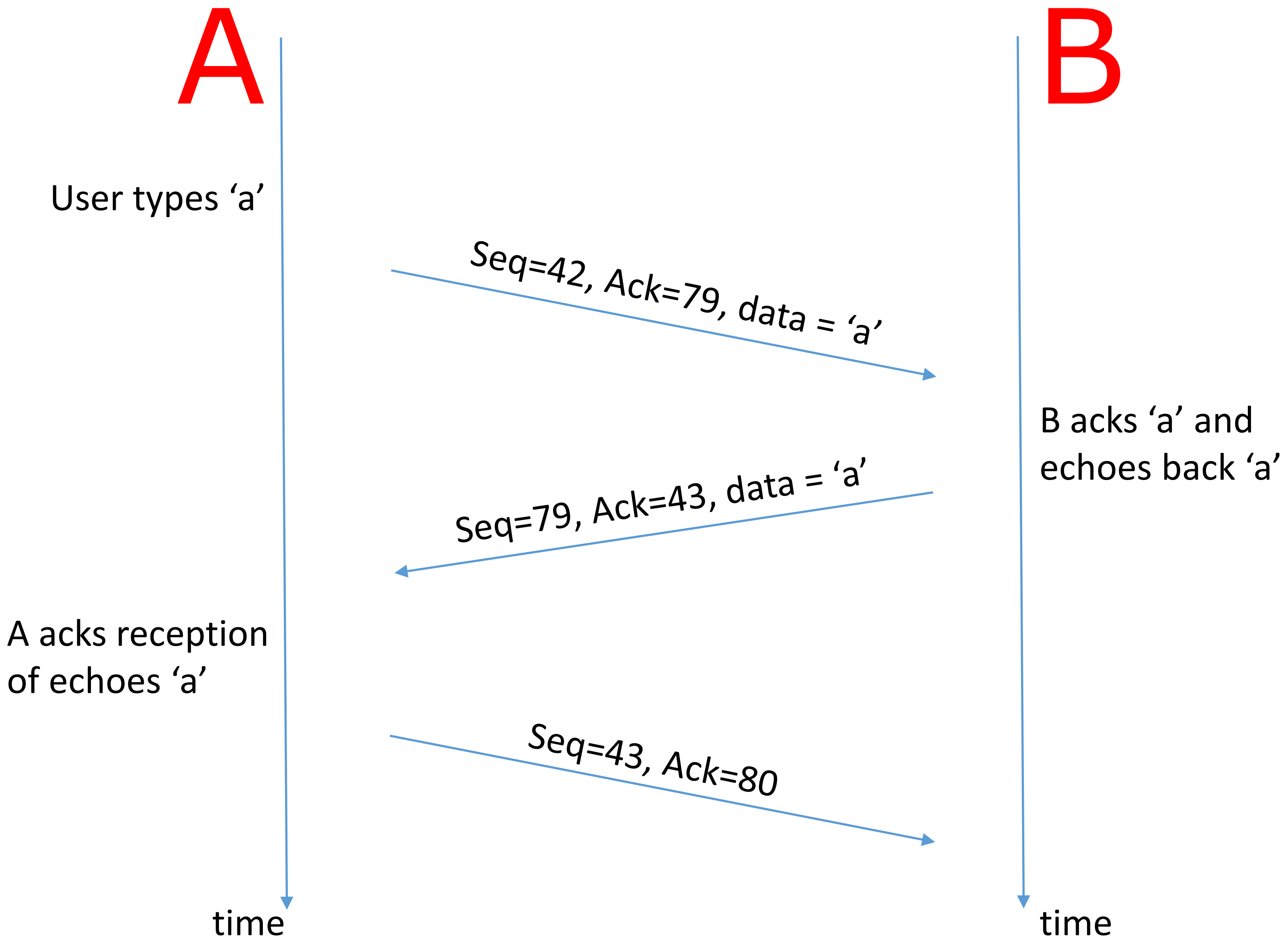
- seq # of next byte expected from other side
- cumulative ACK

How receiver handles out-of-order segments?

- TCP specs doesn't say
- implementation dependent



TCP Sequence Numbers, ACKs 2/2



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TCP Reliable Data Transfer

- TCP creates rdt service on top of IP's unreliable service
  - pipelined segments
  - cumulative acks
  - single retransmission timer
- retransmissions triggered by:
  - timeout events
  - duplicate acks
- let's initially consider simplified TCP sender:
  - ignore duplicate acks
  - ignore flow control, congestion control

Timeout: how long?

- longer than RTT (but RTT varies)
- **too short**: premature timeout, unnecessary retransmissions
- **too long**: slow reaction to segment loss

TCP Sender Events:

data received from app layer:

- 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`

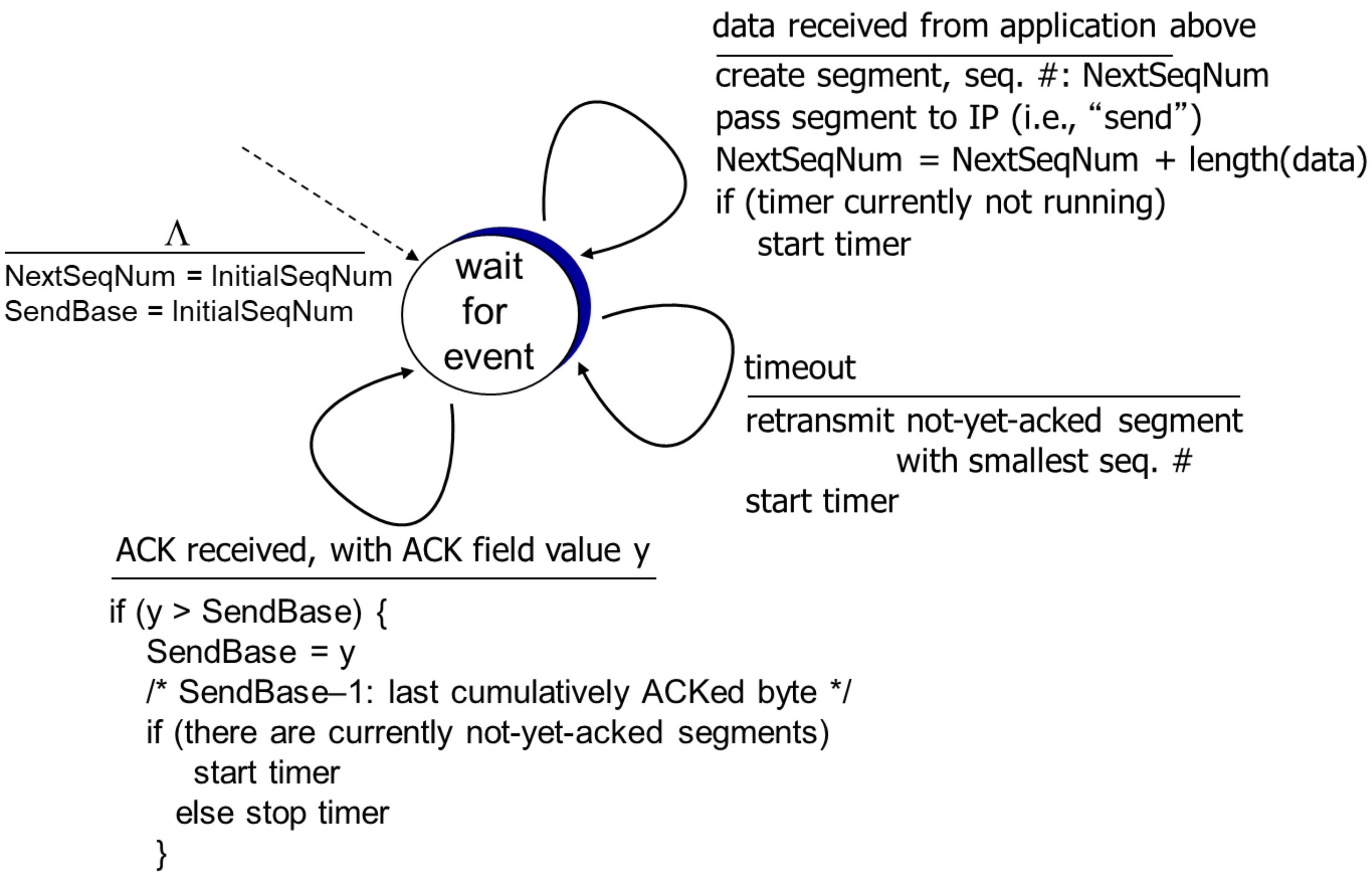
timeout:

- retransmit segment that caused timeout
- restart timer

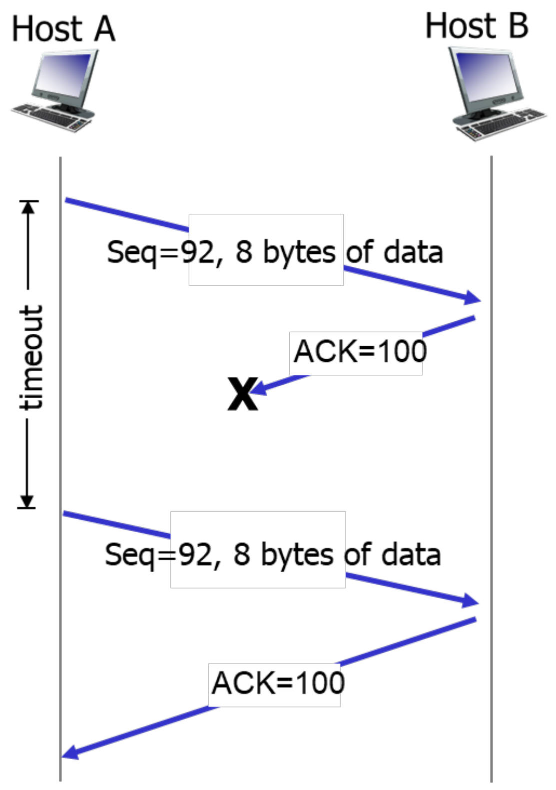
ack received:

- if ack acknowledges previously unacked segments
  - update what is known to be ACKed
  - start timer if there are still unacked segments

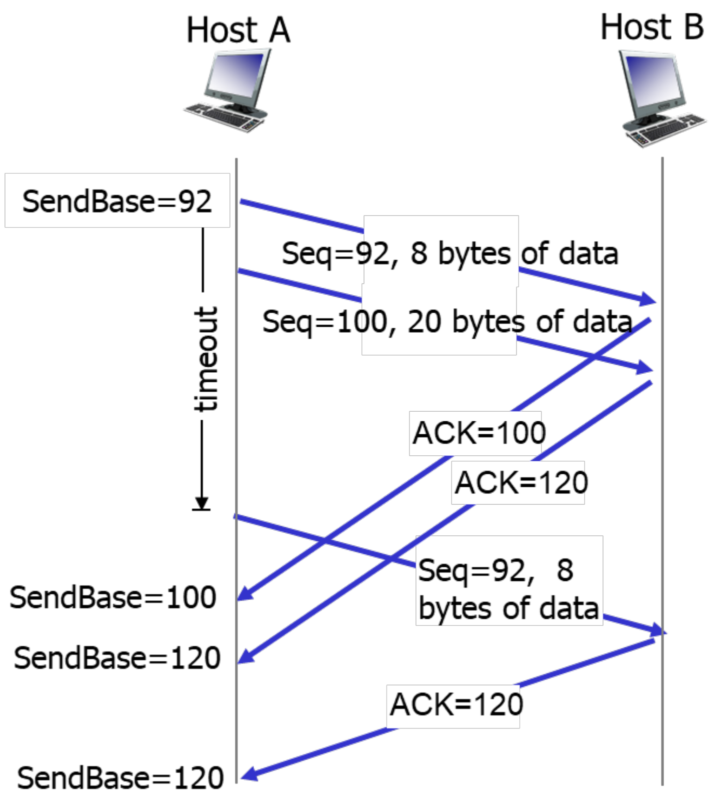
TCP Sender (Simplified)



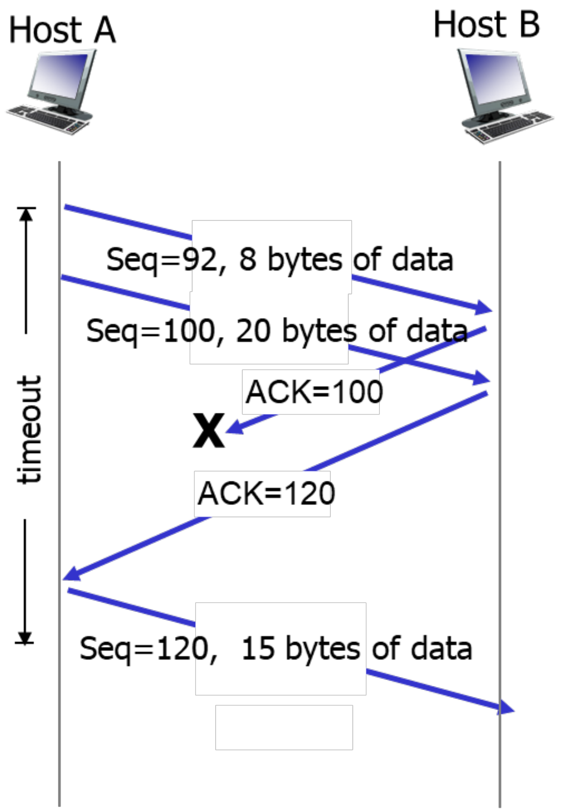
TCP Retransmission Scenarios



lost ACK scenario



premature timeout



cumulative ACK

TCP ACK Generation [RFC 1122, RFC2581]

event at receiver	**TCP receiver action**
arrival of in-order segment with expected seq #. All data up to expected seq # already ACKed	delayed ACK. Wait up to 500ms for next segment. If no next segment, send ACK
arrival of in-order segment with expected seq #. One other segment has ACK pending	immediately send single cumulative ACK, ACKing both in-order segments
arrival of out-of-order segment higher-than-expect Sequence # . Gap detected	immediately send **duplicate ACK,** indicating Sequence # of next expected byte
arrival of segment that partially or completely fills gap	immediate send ACK, provided that segment starts at lower end of gap

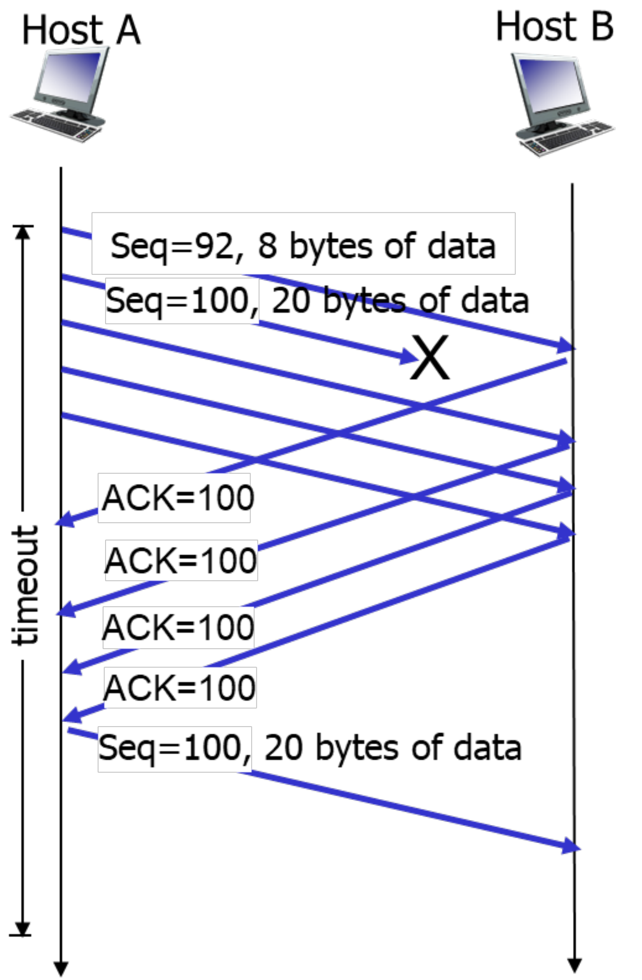
TCP Fast Retransmit

- 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

TCP fast retransmit

if sender receives 3 ACKs for same data ("triple duplicate ACKs"), resend unacked segment with smallest seq #

- likely that unacked segment lost, so don't wait for timeout



fast retransmit after sender receipt of triple duplicate ACK

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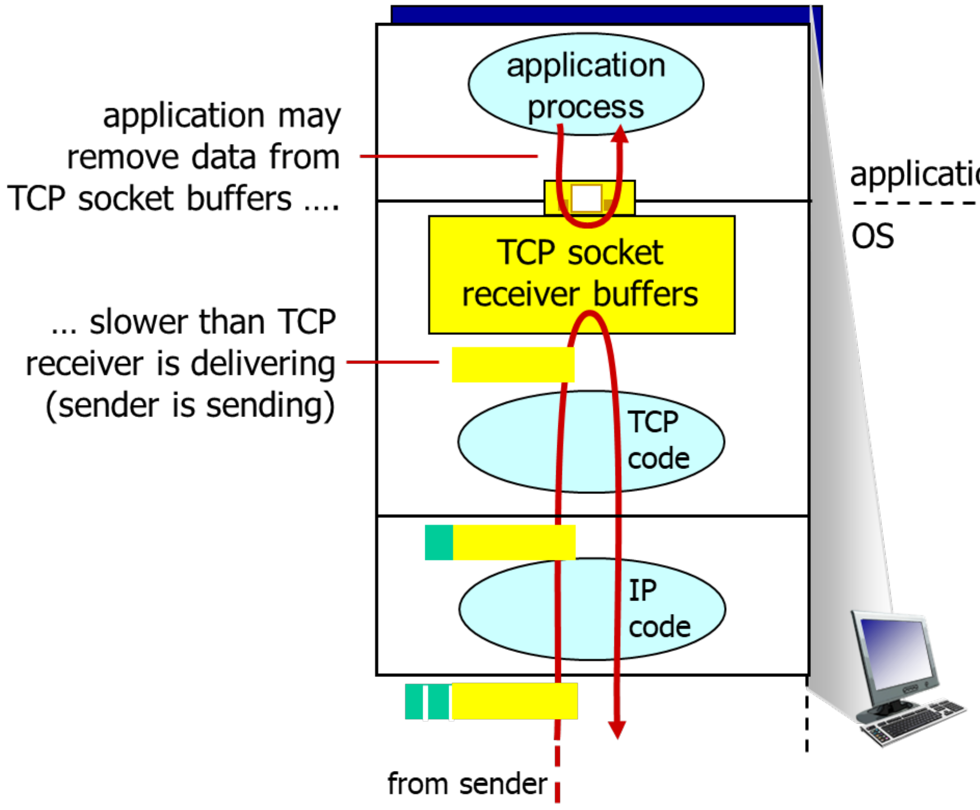
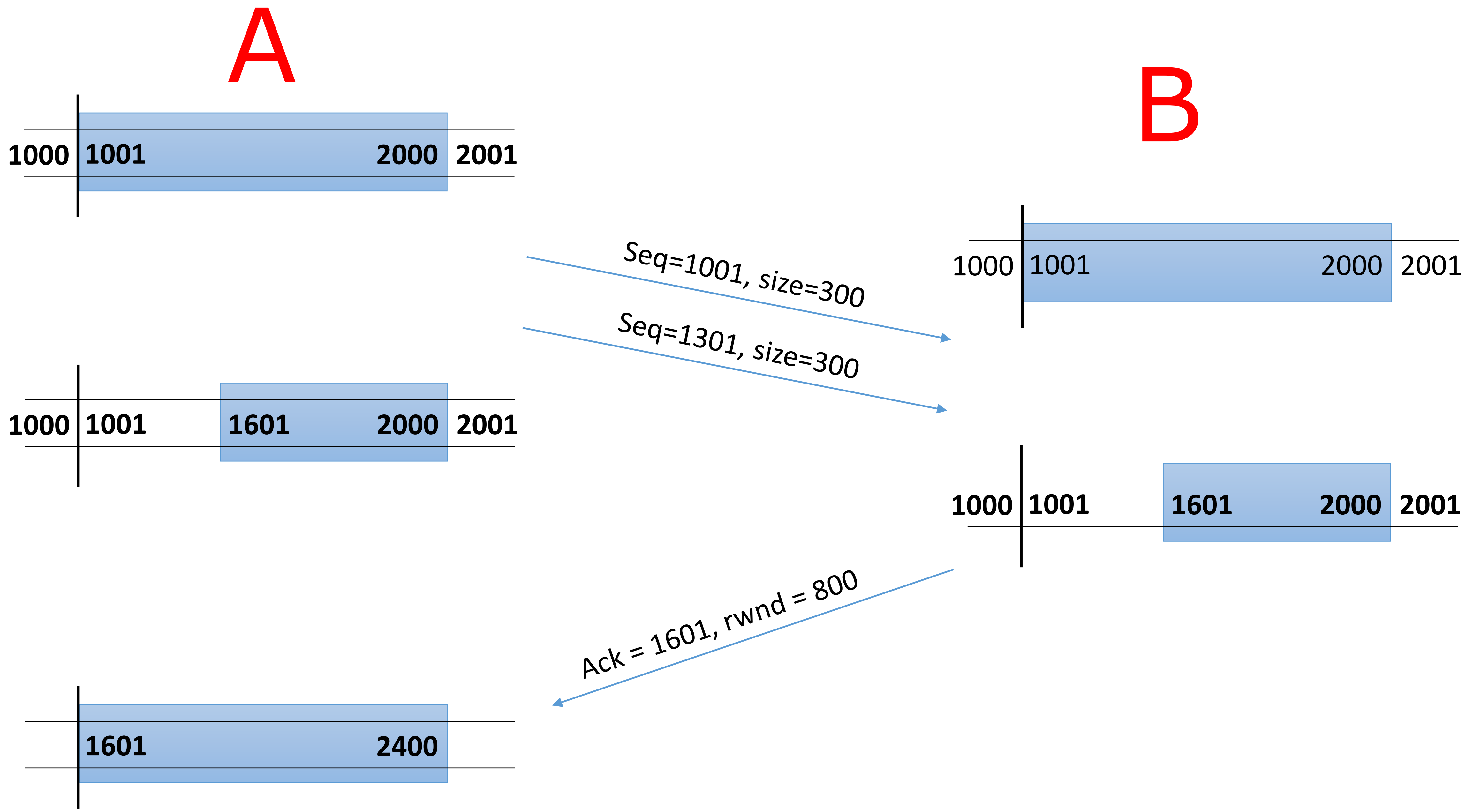
TCP Flow Control

- receiver controls sender, so sender won't overflow receiver's buffer by transmitting too much, too fast
- receiver "advertises" free buffer space by including **`rwnd`** value in TCP header of receiver-to-sender segments
  - **`RcvBuffer`** size set via socket options (typical default is 4096 bytes)
  - many operating systems autoadjust **`RcvBuffer`**
- sender limits amount of unacked ("in-flight") data to receiver's **`rwnd`** value
- guarantees receive buffer will not overflow



receiver protocol stack


A
B
1000
1001
2000
2001
Seq=1001, size=300
Seq=1301, size=300
1000
1001
2000
2001
1000
1001
1601
2000
2001
1000
1001
1601
2000
2001
Ack = 1601, rwnd = 800
1601
2400

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Principles of Congestion Control

congestion:

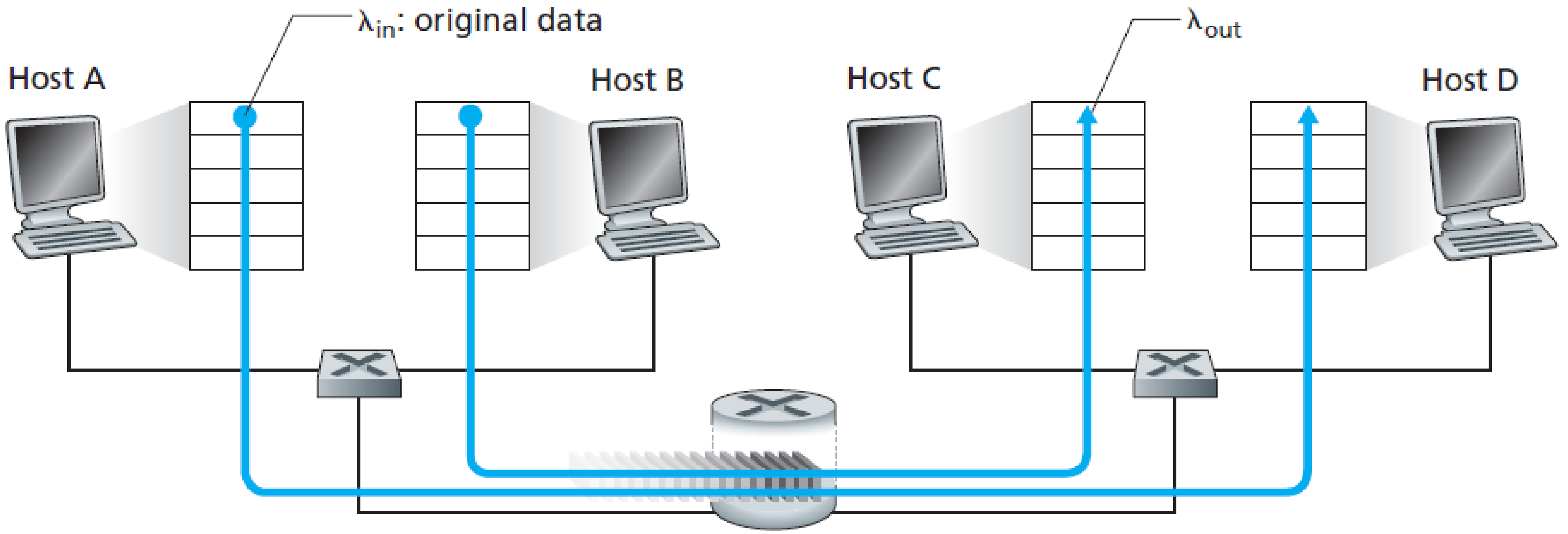
- informally: "too many sources sending too much data too fast for **network** to handle"
- manifestations:
  - lost packets (buffer overflow at routers)
  - long delays (queueing in router buffers)

- flow control vs congestion control
  - buffers of the receiver vs network resources
  - knowing the state of the receiving buffers vs estimating the state of the network

Causes/Costs of Congestion: Scenario 1

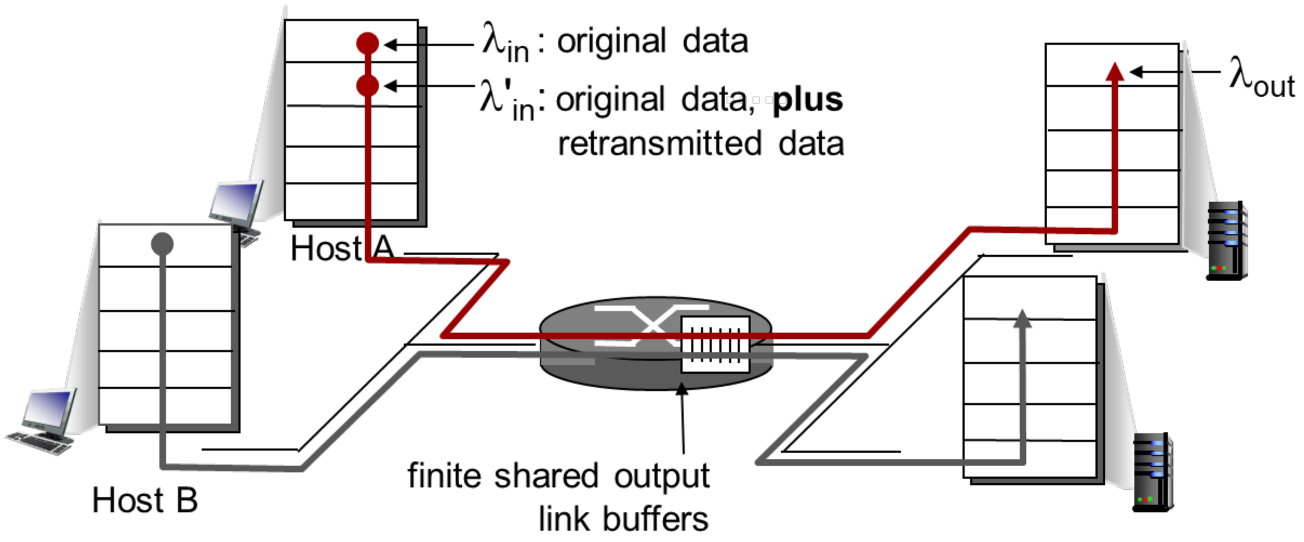
- two senders, two receivers
- one router, infinite buffers
- output link capacity: R
- no retransmission

The capacity of the link must be split to all the data flows



Causes/Costs of Congestion: Scenario 2 (1/2)

- one router, **finite** buffers
- sender retransmission of timed-out packet
  - application-layer input = application-layer output: $\lambda_{in} = \lambda_{out}$
  - transport-layer input includes **retransmissions** : $\lambda'_{in} \geq \lambda_{in}$



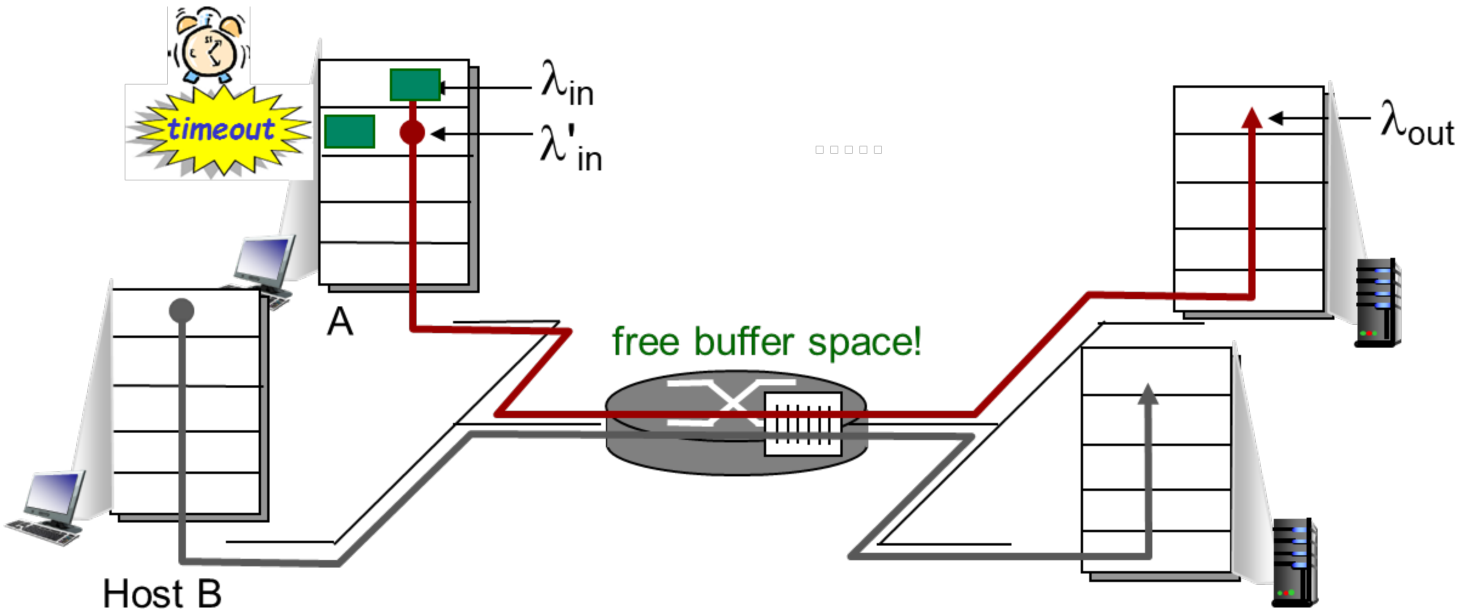
Idealizations:

- **perfect knowledge**
  - sender sends only when router buffers available
- **known loss** packets can be lost, dropped at router due to full buffers
  - sender only resends if packet **known** to be lost

Causes/Costs of Congestion: Scenario 2 (2/2)

Realistic: duplicates

- packets can be lost, dropped at router due to full buffers
- sender times out prematurely, sending **two** copies, both of which are delivered



"costs" of congestion:

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

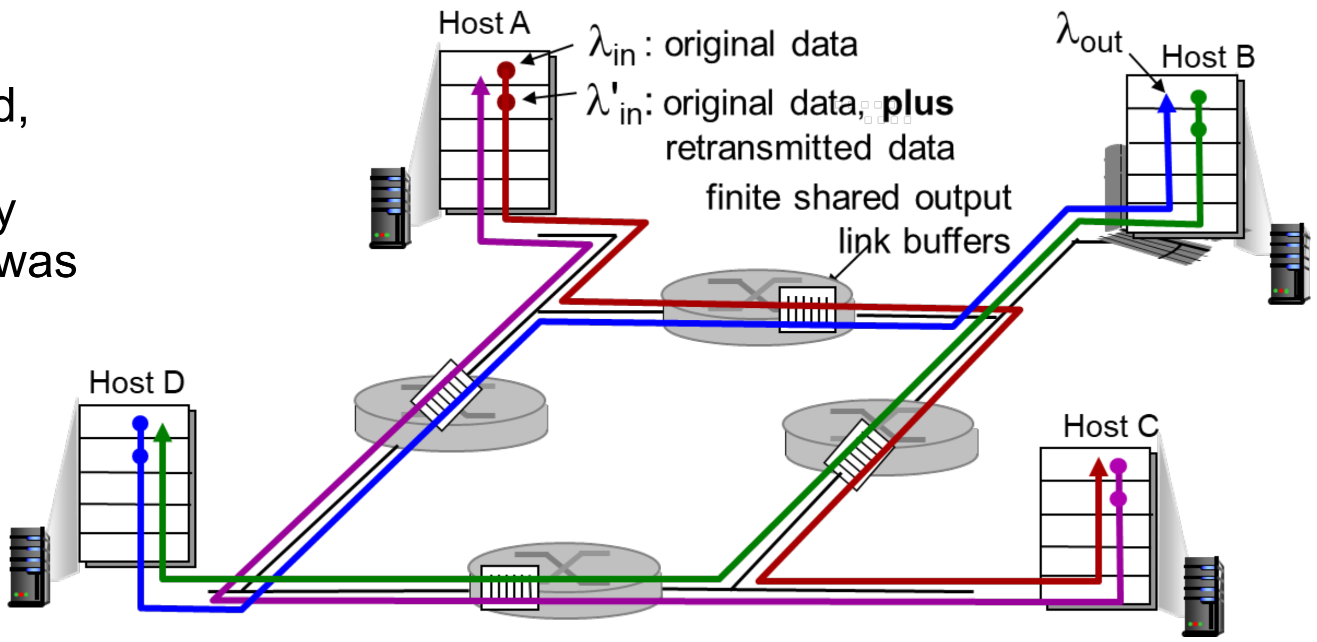
Causes/Costs of Congestion: Scenario 3

- four senders
- multihop paths
- timeout/retransmit

- **Q:** what happens as λ_{in} and λ_{in}' increase **?**
- **A:** as red λ_{in}' increases, all arriving blue pkts at upper queue are dropped, blue throughput $\rightarrow 0$

"cost" of congestion:

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



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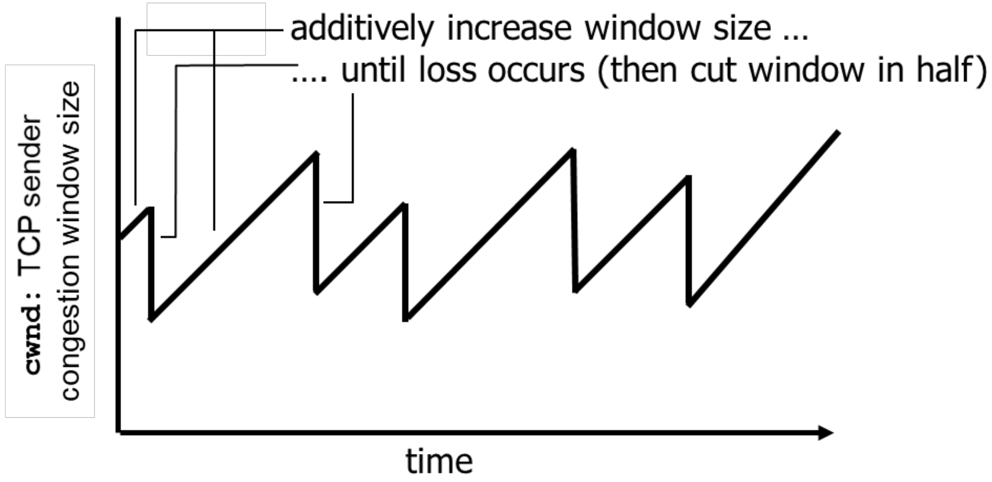
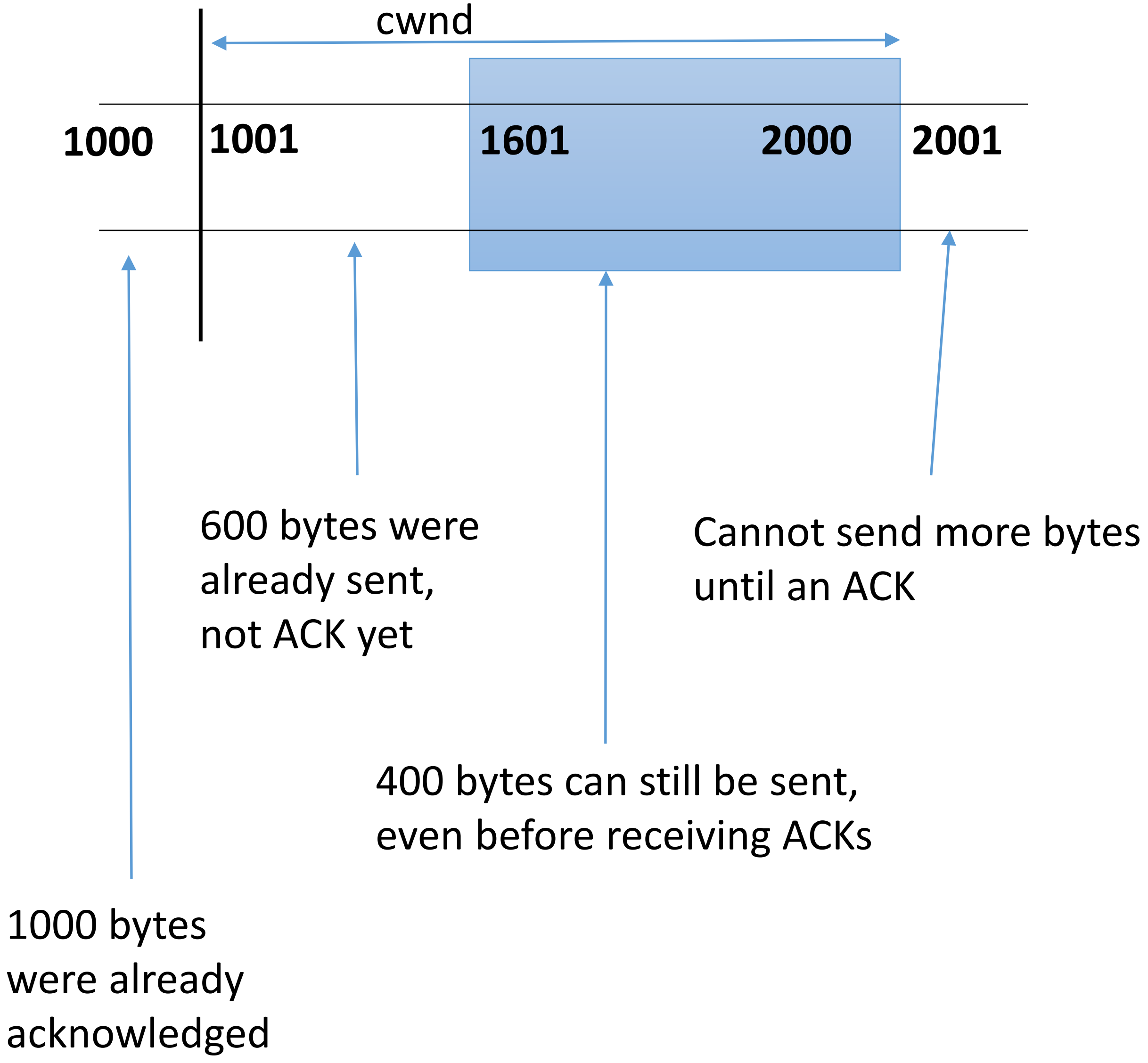
TCP Congestion Control: Additive Increase Multiplicative Decrease

- **Mechanism:** changing `cwnd`
- sender *increases* the window size, probing for usable bandwidth, until loss occurs
  - **additive increase***:* increase **`cwnd`** by 1 MSS every RTT until loss detected
  - **multiplicative decrease**: cut **`cwnd`** in half after loss

AIMD saw tooth behavior: probing for bandwidth



TCP Congestion Control: the sender



- sender limits transmission
- **cwnd** is dynamic, function of perceived network congestion

TCP sending rate:

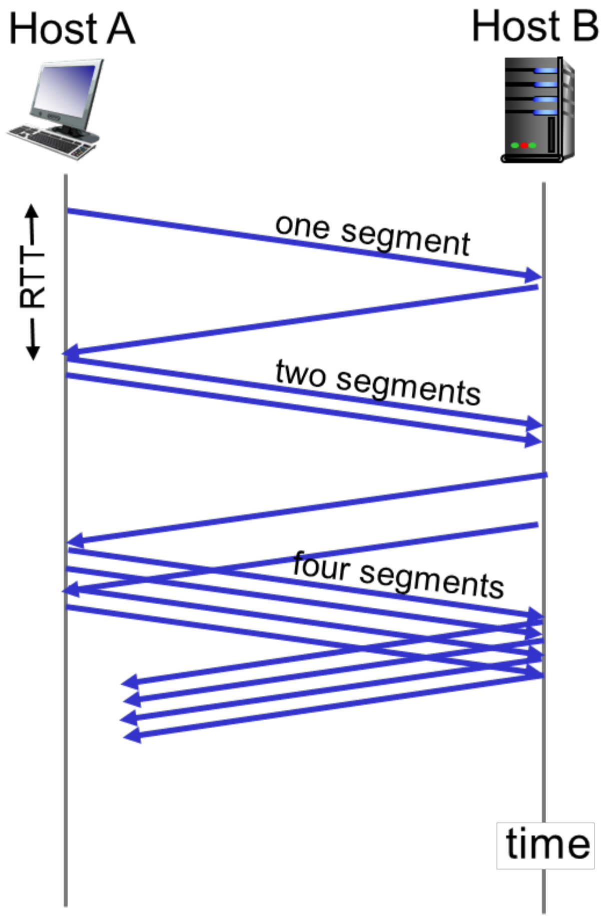
- **roughly:** send cwnd bytes, wait RTT for ACKS, then send more bytes

$$\boxed{\text{rate} \approx \frac{\text{cwnd}}{\text{RTT}} \text{ bytes/sec}}$$

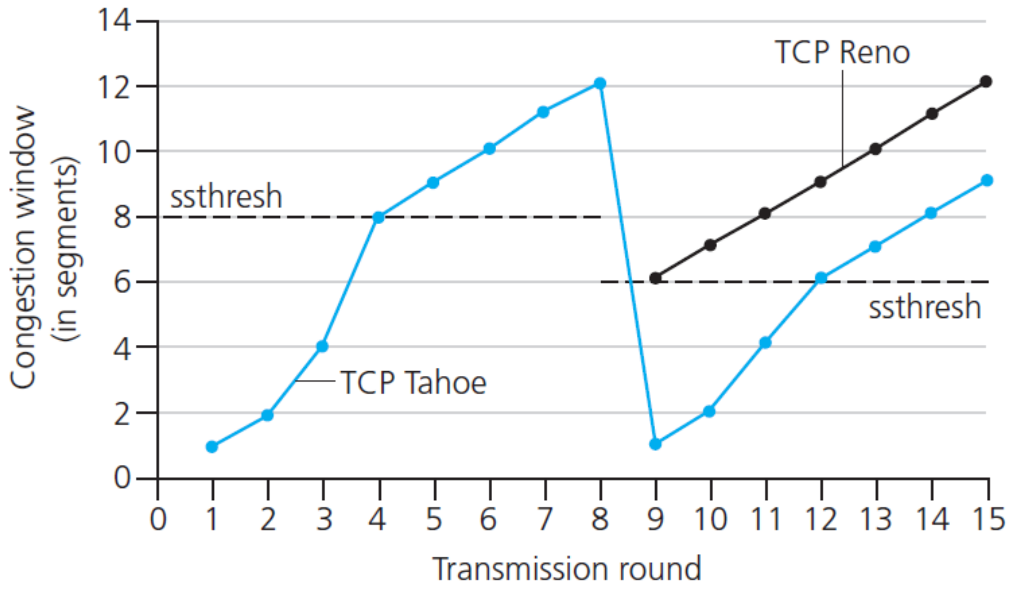
$$\boxed{\texttt{LastByteSent-LastByteAcked} \leq \texttt{cwnd}}$$

TCP Slow Start

- when connection begins, increase rate exponentially until first loss event:
  - initially **`cwnd`** = 1 MSS
  - double **`cwnd`** every RTT
  - done by incrementing **`cwnd`** for every ACK received
- initial rate is slow but ramps up exponentially fast



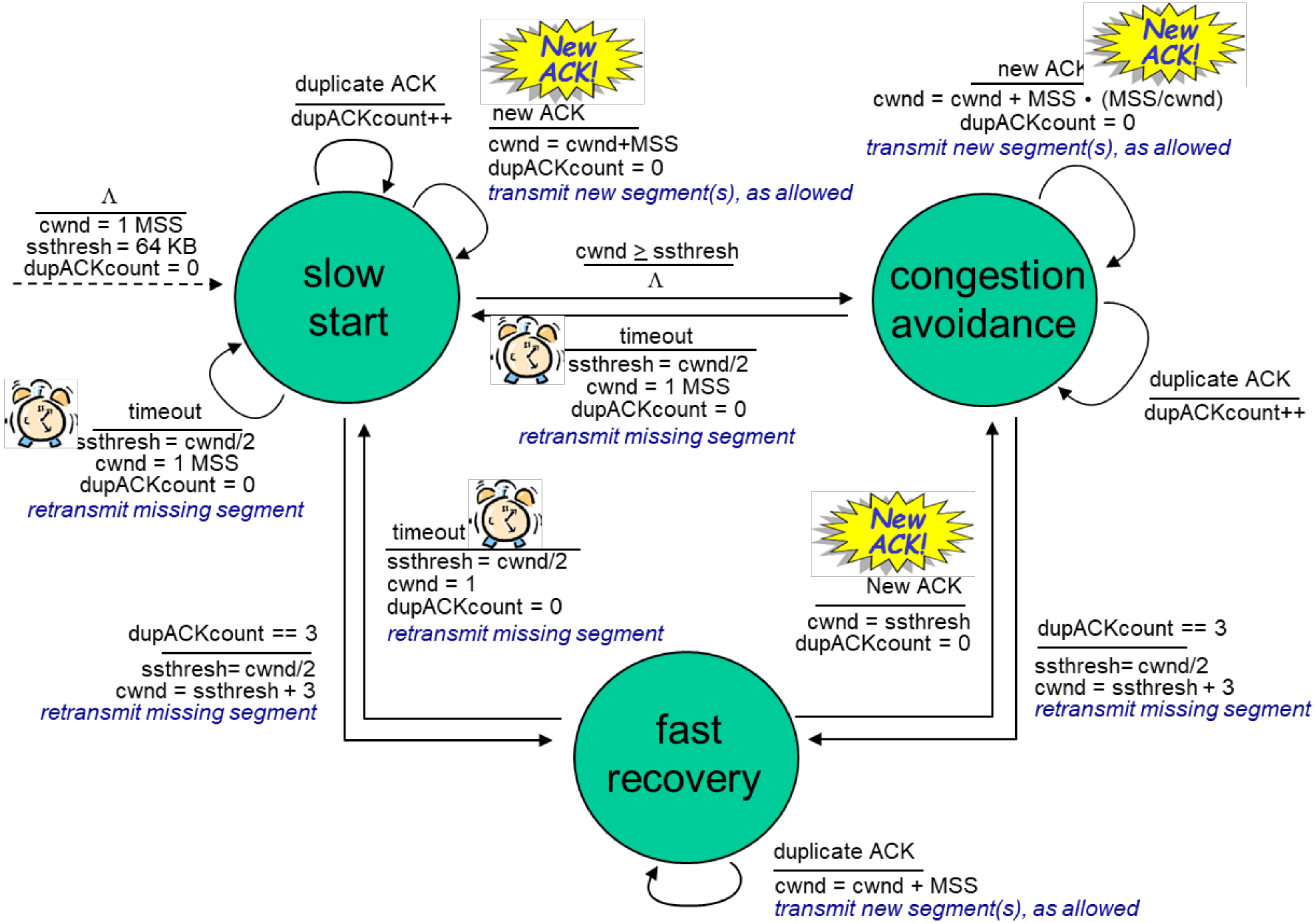
TCP: Detecting, Reacting to Loss



- loss indicated by timeout:
  - **`cwnd`** set to 1 MSS;
  - window then grows exponentially (as in slow start) to threshold (a configuration parameter, controlled by the sender*), then grows linearly
- loss indicated by 3 duplicate ACKs: TCP RENO
  - dup ACKs indicate network capable of delivering some segments
  - **`cwnd`** is cut in half window then grows linearly
- TCP Tahoe always sets **`cwnd`** to 1 (timeout or 3 duplicate acks)

* Check out the online interactive exercises for more examples: http://gaia.cs.umass.edu/kurose_ross/interactive/

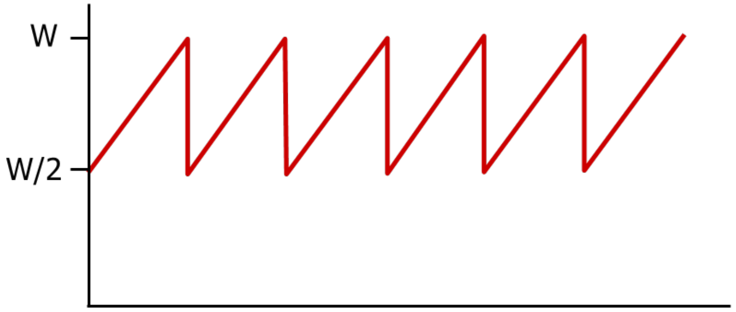
Summary: TCP Congestion Control



TCP Throughput

- average TCP thruput as function of window size, RTT?
  - ignore slow start, assume always data to send
- W: window size (measured in bytes) where loss occurs
  - average window size (# in-flight bytes) is $\frac{3}{4}W$
  - average thruput is $\frac{3}{4}W$ per RTT

$$\text{avg TCP thruput} = \frac{3}{4}\frac{W}{RTT} \text{ bytes/sec}$$



Fairness

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

Fairness and UDP

- multimedia apps often do not use TCP
  - do not want rate throttled by congestion control
- instead use UDP:
  - send audio/video at constant rate, tolerate packet loss

Fairness, parallel TCP connections

- application can open multiple parallel connections between two hosts
- web browsers do this
- example, link of rate R with 9 existing connections:
  - new app asks for 1 TCP, gets rate $\frac{R}{10}$
  - new app asks for 11 TCPs, gets $\frac{R}{2}$

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The Socket

- Formed by the concatenation of a port value and an IP address
  - Unique throughout the Internet
- Used to define an API
  - Generic communication interface for writing programs that use TCP or UDP
- Stream sockets
  - All blocks of data sent between a pair of sockets are guaranteed for delivery and arrive in the order that they were sent
- Datagram sockets
  - Delivery is not guaranteed, nor is order necessarily preserved
- Raw sockets
  - Allow direct access to lower-layer protocols

Defining a socket:

- Two main things to do
  - Addressing
    - Specifying a host and a service (IP + port)
    - It is a tuple (“www.cs.aau.dk”, 80)
  - Data transport
    - Mainly TCP (SOCK_STREAM) or UDP (SOCK_DGRAM)

Socket Programming with UDP

UDP: no “connection” between client & server

- no handshaking before sending data
- sender explicitly attaches IP destination address and port # to each packet
- receiver extracts sender IP address and port# from received packet

UDP: transmitted data may be lost or received out-of-order

Application viewpoint:

- UDP provides **unreliable** transfer of groups of bytes (“datagrams”) between client and server

Client/Server Socket Interaction: UDP

server (running on serverIP)

create socket, port= x:
serverSocket =
socket(AF_INET,SOCK_DGRAM)

read datagram from
serverSocket

write reply to
serverSocket
specifying
client address,
port number

client

create socket:
clientSocket =
socket(AF_INET,SOCK_DGRAM)

Create datagram with server IP and
port=x; send datagram via
clientSocket

read datagram from
clientSocket

close
clientSocket

Socket Programming with TCP

client must contact server

- server process must first be running
- server must have created socket (door) that welcomes client's contact

client contacts server by:

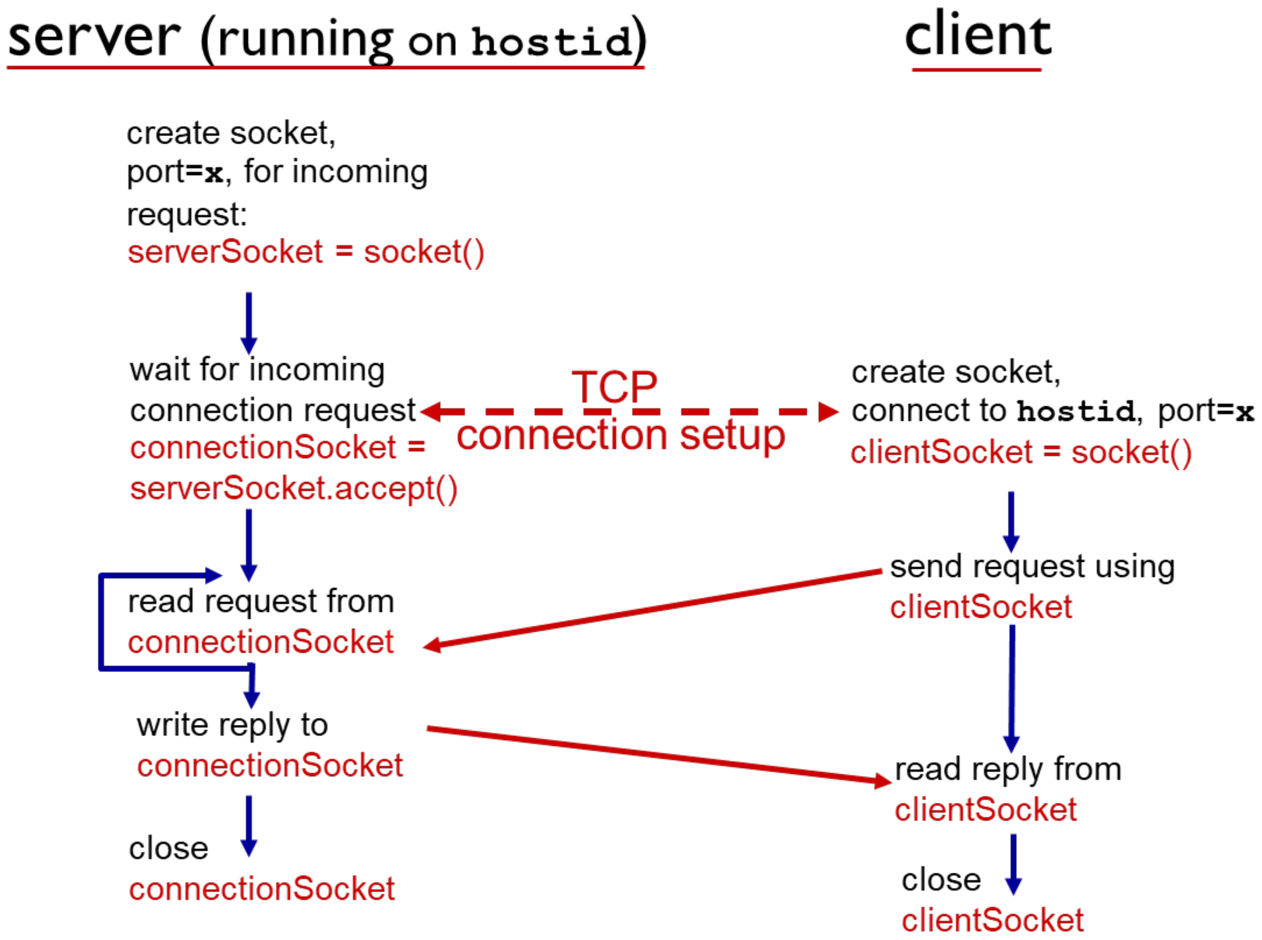
- Creating TCP socket, specifying IP address, port number of server process
- **when client creates socket:** client TCP establishes connection to server TCP

- when contacted by client, **server TCP creates new socket** for server process to communicate with that particular client
  - allows server to talk with multiple clients
  - source port numbers used to distinguish clients (more in Chap 3)

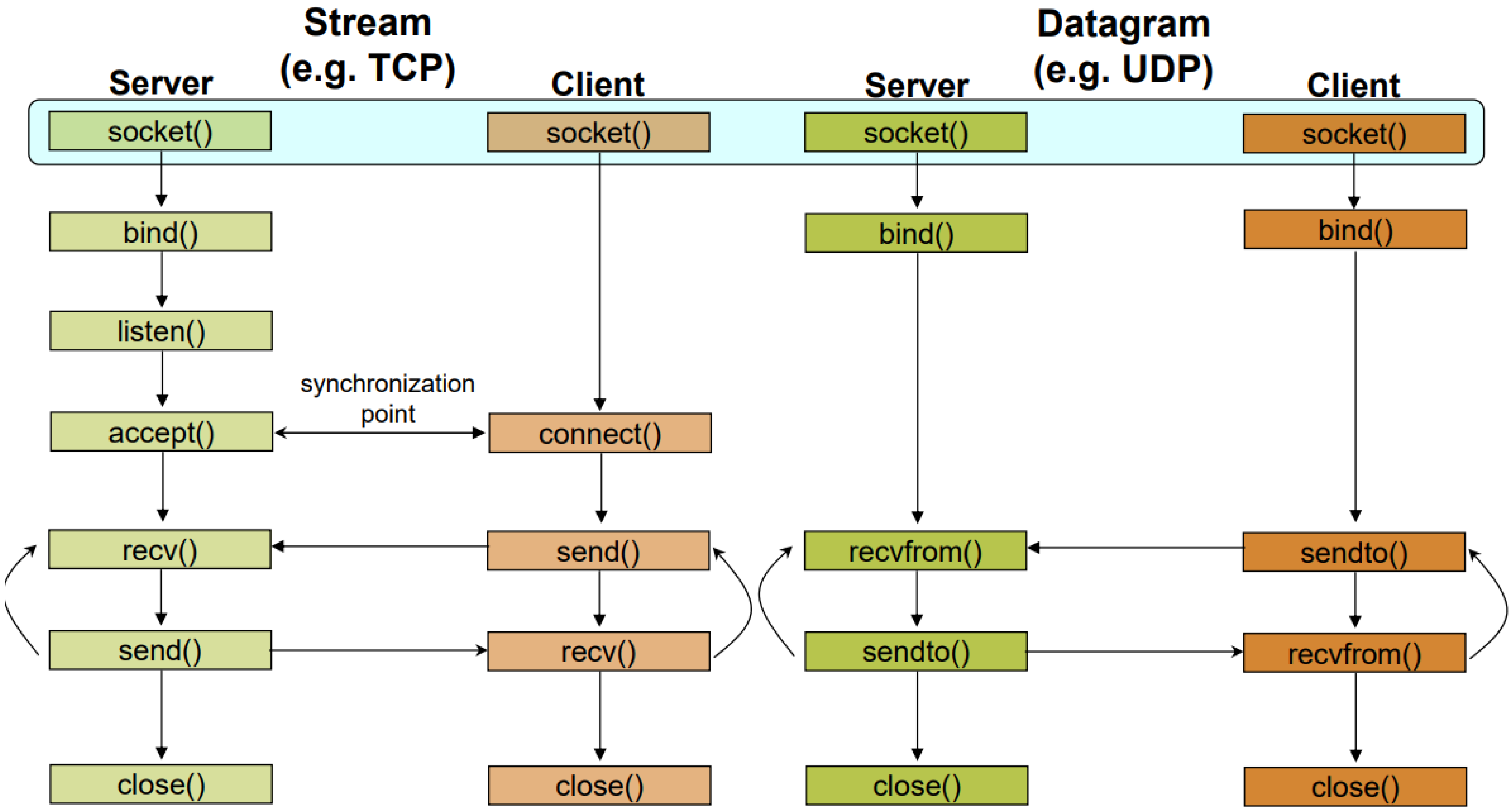
application viewpoint:

TCP provides reliable, in-order byte-stream transfer ("pipe") between client and server

Client/Server Socket Interaction: TCP



The Socket API



Always check the return value of these calls !!!!

C Server

1) Create a socket, bind it to a port, make it listen:

```
s = socket(AF_INET, SOCK_STREAM)
s.bind(my_address, 9000)
listen(s, 5) # up to 5 pending connections
```

2) Accept an incoming connection to have a connected socket:

```
c = accept(s, (struct sockaddr*)&client_addr,
&addrlen);
print("connected with %s", client_addr)
```

3) Communicate:

```
send(c, "you are connected\n", 19)
```

6) Close the Socket:

```
close(c);
```

7) Go to Step 2.

C Client

1) Create a socket, connect it:

```
s = socket(AF_INET, SOCK_STREAM)
connect(s, destination)
```

3) Communicate (receive data):

```
recv(d, data, 1024)
printf("Received %s", data)
```

4) Close the socket:

```
close(s);
```

Javascript Server

1) Import module “net”, create a socket, define what must be done all the time there is a connection:

```
const net = require('net');
const server = net.createServer((socket) => { … });
```

2) The function is in an implicit infinite loop. Extract the address of the client, log, and answer to the client:

```
net.createServer((socket) => {
  addr = socket.address();
  console.log("%s:%d connected", addr.address, addr.port);
  socket.end("you are " + addr.address +
      ":" + addr.port + "\n");
});
```

3) Bind and listen on the server socket:

```
server.listen(9999);
```

Javascript Client

1) Import module “net”, create a socket:

```
const net = require('net');
const client = new net.Socket();
```

2) Connect the socket to the host provided by the command line:

```
client.connect({ port: 9999, host: process.argv[2] });
```

3) Communicate (receive data) and log:

```
client.on('data', (data) => {
  console.log("Received:\n" + data.toString('utf-8'));
});
```

Windows programming

- Microsoft "version" of socket programming is called Winsock
- Same philosophy, implementation differences:
  - `WINSOCK_API_LINKAGE SOCKET WSAAPI socket( int af, int type, int protocol )`
    - It's not returning an integer!
- Need to initialize the Winsock subsystem:
  - `WSADATA wsaData;`
  - `iResult = WSAStartup(MAKEWORD(2,2), &wsaData);`
  - `if (iResult != 0) { exit(-1); }`

Issues:

- Reading/writing to a socket may involve partial data transfer
  - send() returns actual bytes sent
  - recv() length is only a maximum limit
- For TCP, the data stream is continuous---no concept of records, etc.
  - One recv() can return the strings sent in two send()
  - A send() can provide strings to two recv()s with a small maximum limit
- How to tell if there is no more data, in the sense that the connection has been (half-)closed?
  - recv() will return empty string
  - ret = s.recv(1000)
    - … and ret == “”

Issues with the recv()

- If a peer is doing a cycle like:

```
while (l = s.recv()):
    do_something(l)
```

- There is the problem of termination:
  - Until `recv()` provides data, it doesn't end
  - When there is no more data, the program is stuck on the blocking `recv()` call
- Four solution:
  - Doing a `close()` on the other side
  - Send a *special* termination string
  - Use a timeout
  - Send the number of character that will be sent in the beginning

Drawbacks of each solution

- Doing a `close()` on the other side
  - The `read()` will return (with a “” string)
  - But the socket cannot be used anymore
- Send a *special* termination string
  - But if it was part of the normal communication flow, it could quit the application too early
- Use a timeout
  - The application gets slow since it needs to wait until the timeout
  - Should the network be slow, the timeout could end the application too early
- Send the number of character that will be sent in the beginning
  - In some applications, you don’t know the number of characters in advance

SLUT