Last time

- Reliable Data Transfer with packet loss
  - rdt3.0
- Pipelining
  - Problems with stop-and-wait
  - Go-Back-N
  - Selective-Repeat

This time

- UDP socket programming
- TCP

Chapter 2: Application layer

- 2.1 Principles of network applications
- 2.2 Web and HTTP
- 2.3 FTP
- 2.4 Electronic Mail
  - SMTP, POP3, IMAP
- 2.5 DNS
- 2.6 P2P file sharing
- 2.7 Socket programming with TCP
- 2.8 Socket programming with UDP
- 2.9 Building a Web server

Socket programming

Goal: learn how to build client/server application that communicate using sockets

Socket API

- introduced in BSD4.1 UNIX, 1981
- explicitly created, used, released by apps
- client/server paradigm
- two types of transport service via socket API:
  - unreliable datagram
  - reliable, byte stream-oriented

socket

a *host-local*, *application-created*, *OS-controlled* interface (a "door") into which application process can both send and receive messages to/from another application process

Socket programming *with UDP*

UDP: no "connection" between client and server

- no handshaking
- sender explicitly attaches IP address and port of destination to each packet
- server must extract IP address, port of sender from received packet

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

application viewpoint

UDP provides unreliable transfer of groups of bytes ("datagrams") between client and server

Client/server socket interaction: UDP

Server (running on `hostid`)

create socket,
port=**x**, for
incoming request:
serverSocket =
new DatagramSocket(x)

read request from
serverSocket

write reply to
serverSocket
specifying client
host address,
port number

Client

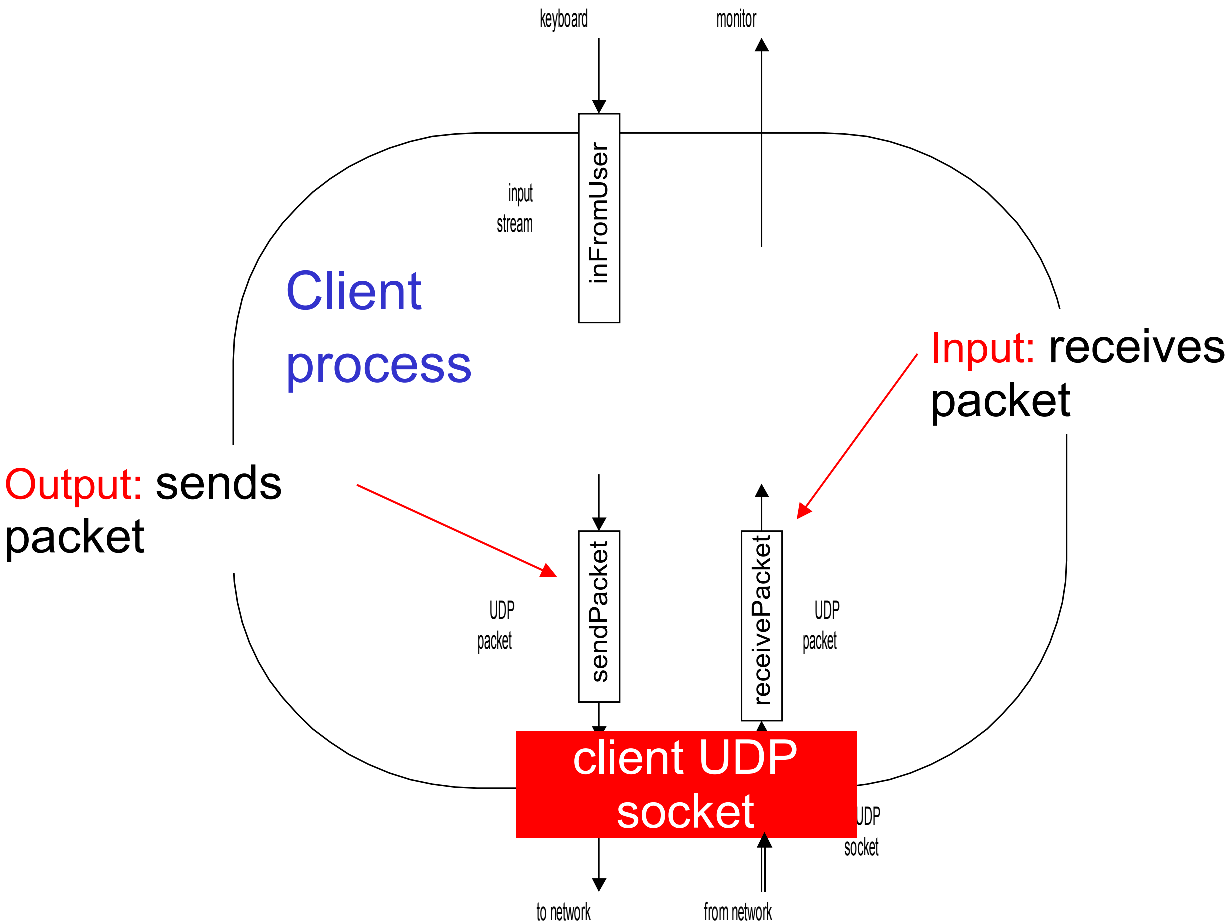
create socket,
clientSocket =
new DatagramSocket()

Create, address (**hostid, port=x**),
send datagram request
using clientSocket

read reply from
clientSocket

close
clientSocket

Example: Java client (UDP)



Example: Java client (UDP)

Create input stream

Create client socket

Translate hostname to IP address using DNS

```
import java.io.*;
import java.net.*;

class UDPClient {
    public static void main(String args[]) throws Exception
    {

      BufferedReader inFromUser =
        new BufferedReader(new InputStreamReader(System.in));

      DatagramSocket clientSocket = new DatagramSocket();

      InetAddress IPAddress = InetAddress.getByName("hostname");

      byte[] sendData = new byte[1024];
      byte[] receiveData = new byte[1024];

      String sentence = inFromUser.readLine();

      sendData = sentence.getBytes();
```

Example: Java client (UDP), cont.

Create datagram with data-to-send, length, IP addr, port

Send datagram to server

Read datagram from server

```
DatagramPacket sendPacket =
   new DatagramPacket(sendData, sendData.length, IPAddress, 9876);

clientSocket.send(sendPacket);

DatagramPacket receivePacket =
   new DatagramPacket(receiveData, receiveData.length);

clientSocket.receive(receivePacket);

String modifiedSentence =
    new String(receivePacket.getData());

System.out.println("FROM SERVER:" + modifiedSentence);
clientSocket.close();
}
}
```

Example: Java server (UDP)

```
import java.io.*;
import java.net.*;

class UDPServer {
  public static void main(String args[]) throws Exception
    {

      DatagramSocket serverSocket = new DatagramSocket(9876);

      byte[] receiveData = new byte[1024];
      byte[] sendData  = new byte[1024];

      while(true)
        {

          DatagramPacket receivePacket =
             new DatagramPacket(receiveData, receiveData.length);
          serverSocket.receive(receivePacket);
```

Create datagram socket at port 9876

Create space for received datagram

Receive datagram

Example: Java server (UDP), cont

Get IP addr port #, of sender

Create datagram to send to client

Write out datagram to socket

```
        String sentence = new String(receivePacket.getData());

        InetAddress IPAddress = receivePacket.getAddress();

        int port = receivePacket.getPort();

                String capitalizedSentence = sentence.toUpperCase();

        sendData = capitalizedSentence.getBytes();

        DatagramPacket sendPacket =
           new DatagramPacket(sendData, sendData.length, IPAddress,
                              port);

        serverSocket.send(sendPacket);
      }
    }
}
```

End of while loop, loop back and wait for another datagram

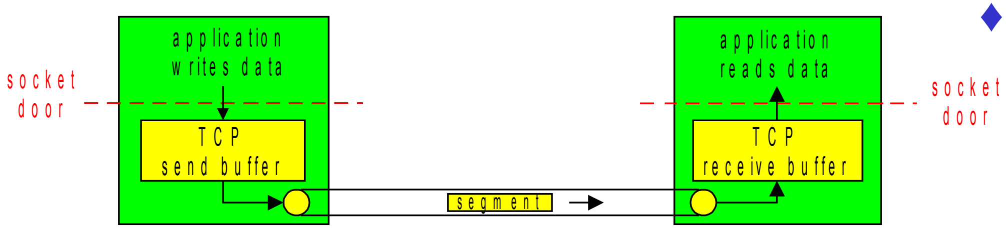
Chapter 3 outline

- 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

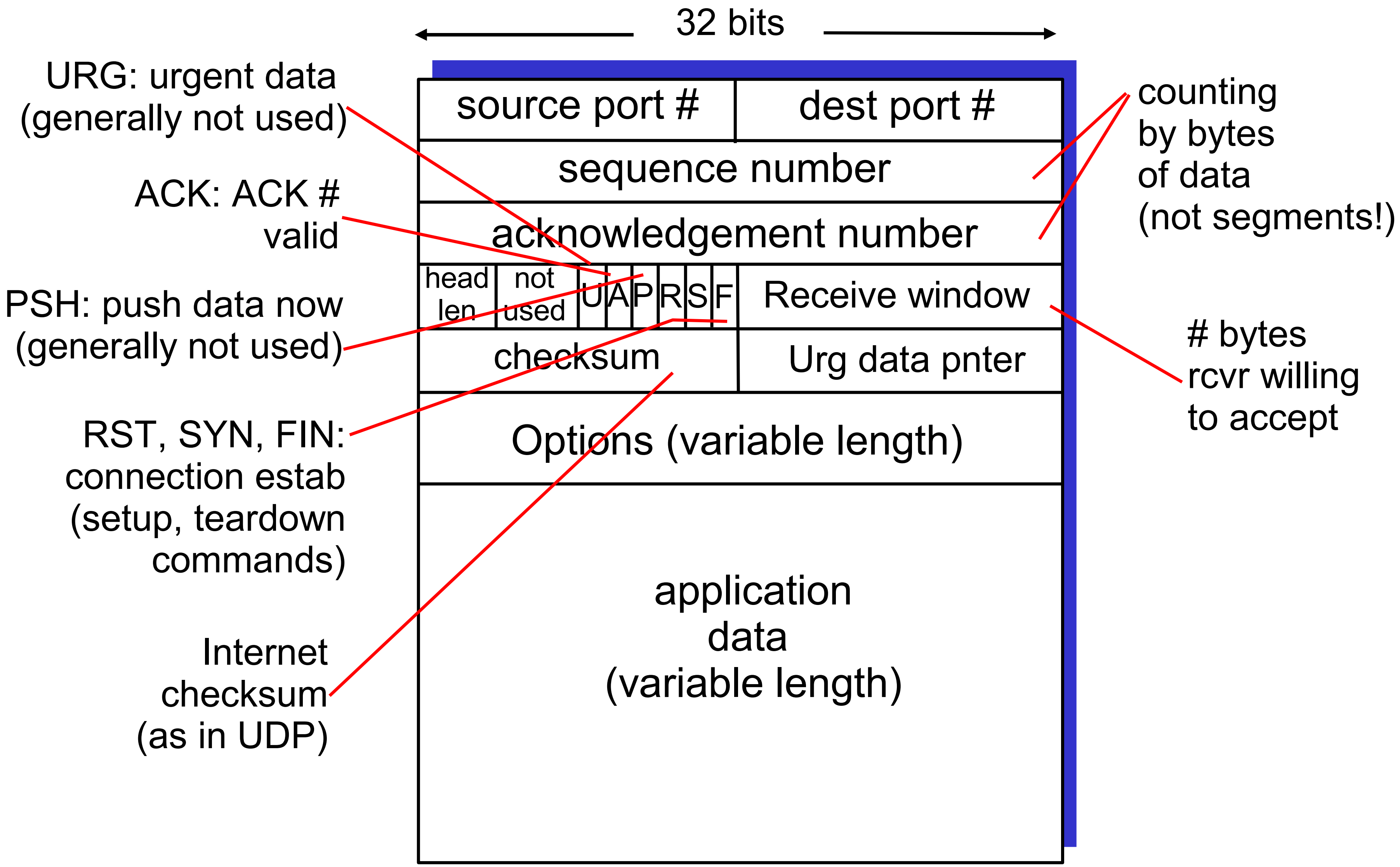
TCP: Overview

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) inits sender, receiver state before data exchange
- Flow controlled:
  - sender will not overwhelm receiver



TCP segment structure



TCP seq. #'s and ACKs

Seq. #'s:

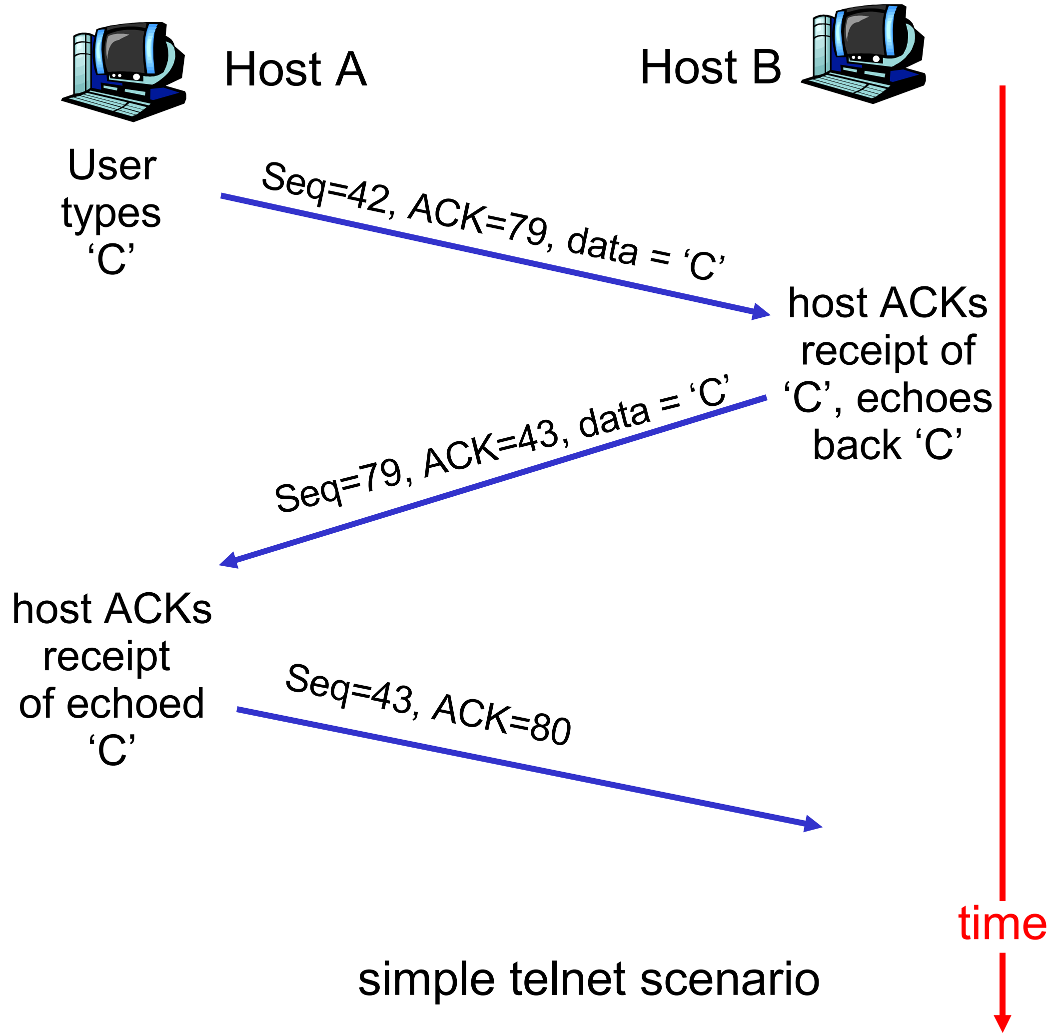
- byte stream "number" of first byte in segment's data

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



simple telnet scenario

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?

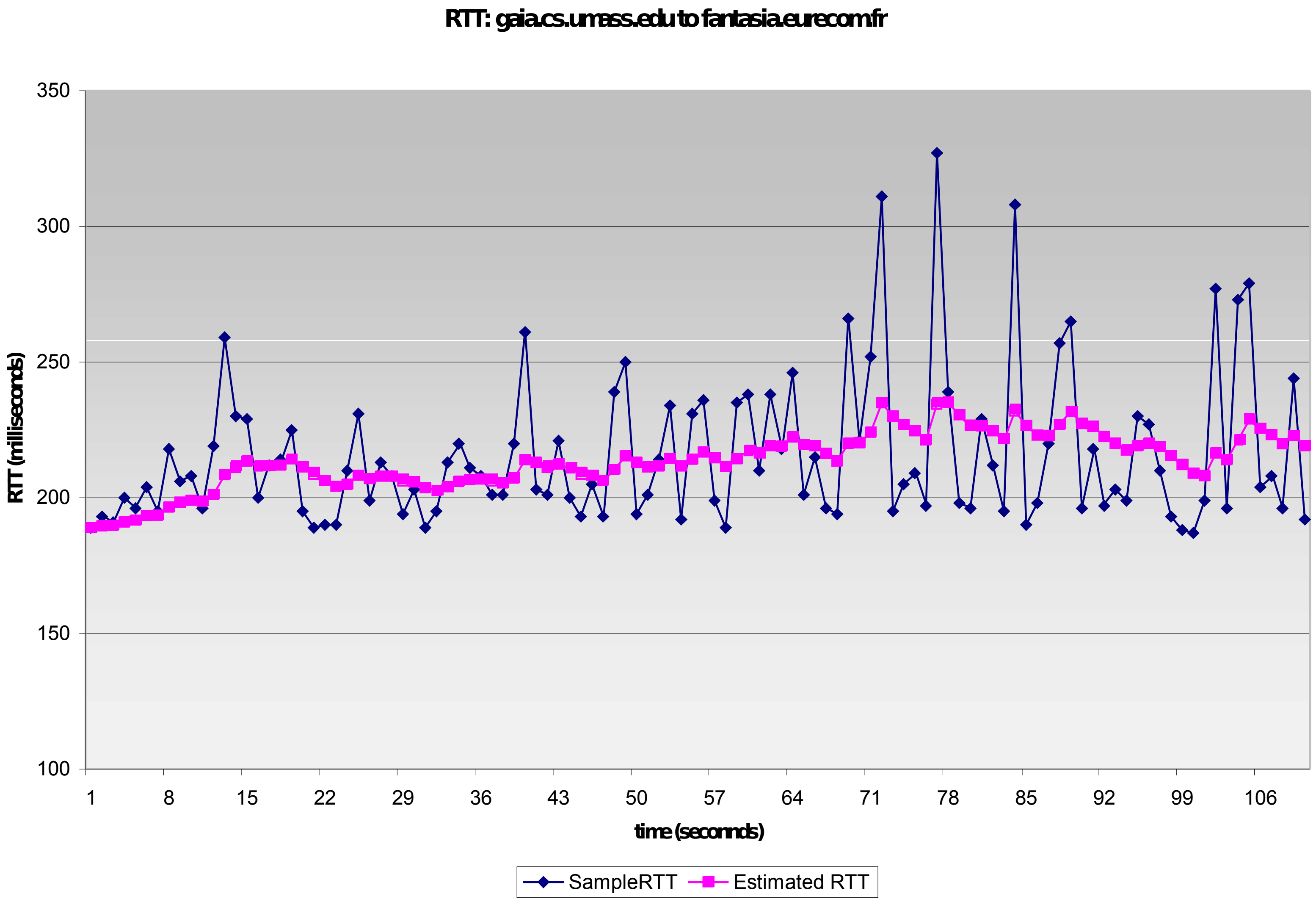
- **`SampleRTT`**: measured time from segment transmission until ACK receipt
  - ignore retransmissions
- **`SampleRTT`** will vary, want estimated RTT "smoother"
  - average several recent measurements, not just current **`SampleRTT`**

TCP Round Trip Time and Timeout

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

- Exponentially 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

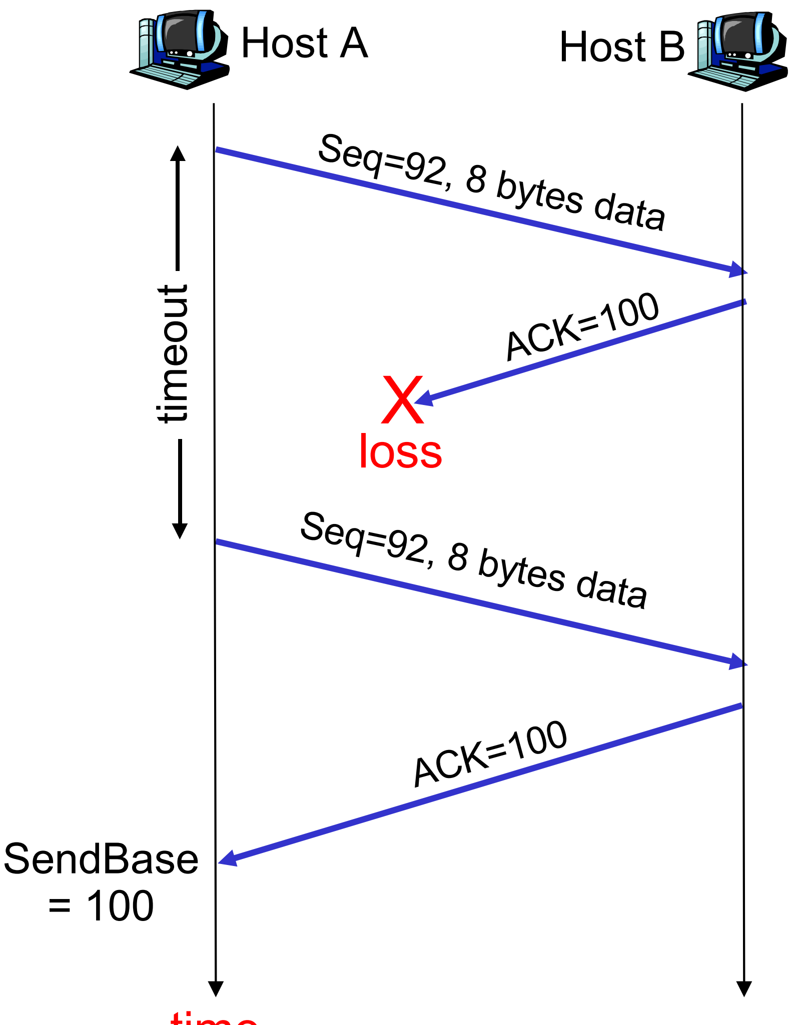
TCP sender (simplified)

```
NextSeqNum = InitialSeqNum
SendBase = InitialSeqNum

loop (forever) {
  switch(event)

  event: data received from application above
      create TCP segment with sequence number NextSeqNum
      if (timer currently not running)
          start timer
      pass segment to IP
      NextSeqNum = NextSeqNum + length(data)

   event: timer timeout
      retransmit not-yet-acknowledged segment with
           smallest sequence number
      start timer

   event: ACK received, with ACK field value of y
      if (y > SendBase) {
          SendBase = y
         if (there are currently not-yet-acknowledged segments)
               start timer
         }

 }  /* end of loop forever */
```

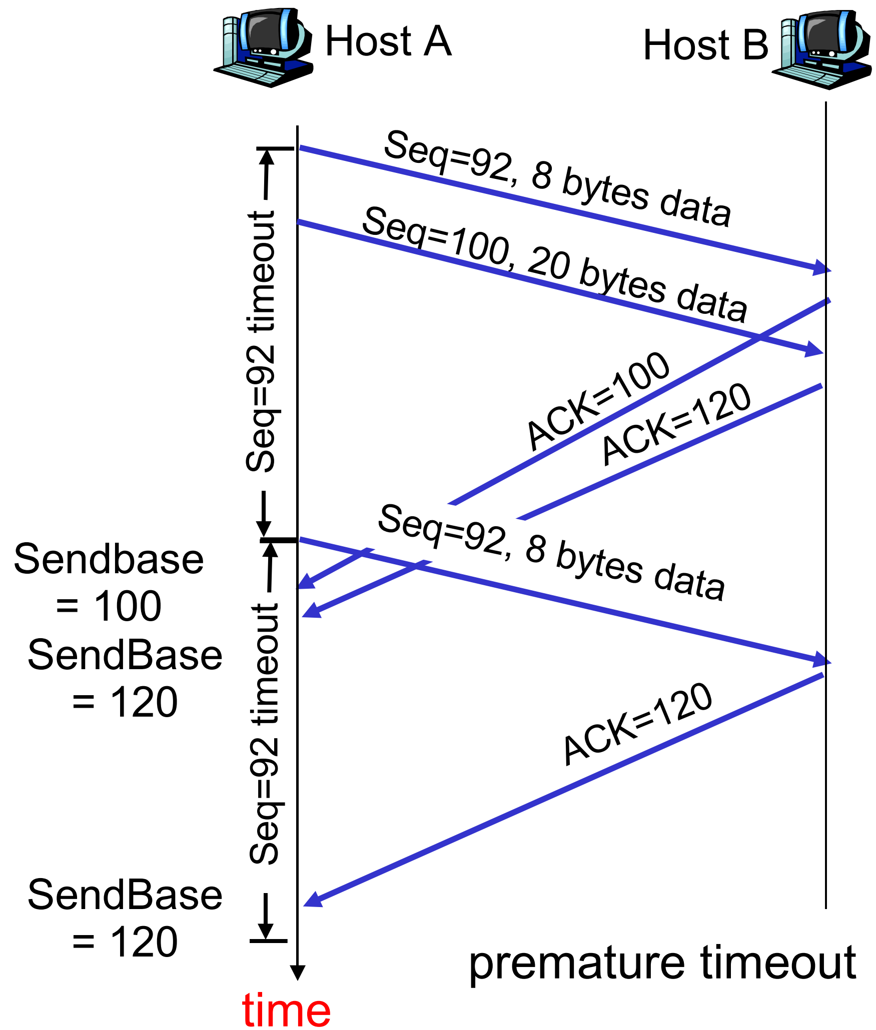
Comment:
- SendBase-1: last cumulatively ack'ed byte

Example:
- SendBase-1 = 71; y= 73, so the rcvr wants 73+ ; y > SendBase, so that new data is acked

TCP: retransmission scenarios

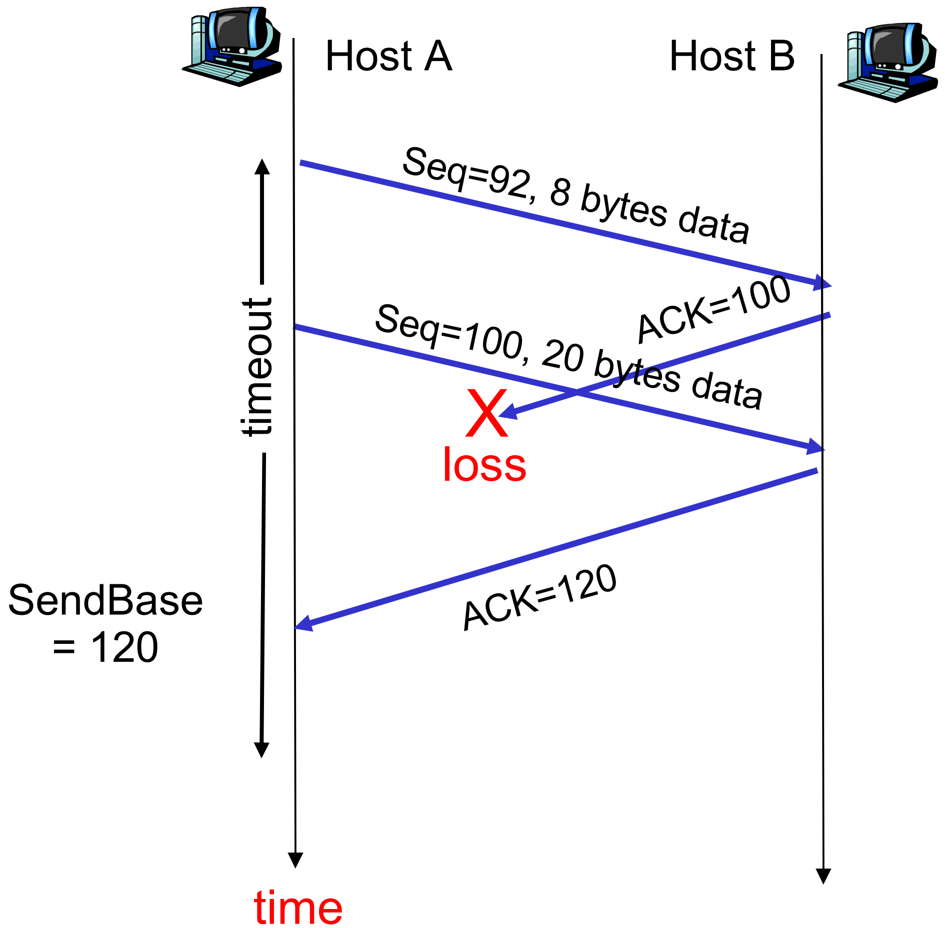


lost ACK scenario



premature timeout

TCP retransmission scenarios (more)



Cumulative ACK scenario

TCP ACK generation [RFC 1122, RFC 2581]

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 seq. # . Gap detected	Immediately send *duplicate ACK*, indicating seq. # of next expected byte
Arrival of segment that partially or completely fills gap	Immediate send ACK, provided that segment startsat lower end of gap

Recap

- UDP socket programming
  - DatagramSocket, DatagramPacket
- TCP
  - Sequence numbers, ACKs
  - RTT, DevRTT, timeout calculations
  - Reliable data transfer algorithm

Next time

- TCP
  - Fast retransmit
  - Flow control
  - Connection management
  - Congestion control