Communication (T

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Previously…

- Previous lecture:
  - Message-based communication is complex (e.g., routing towards destination, subject to message losses)
- Today's lecture:
  - How to avoid m
  - How to give the pens locally (not remotely)?
  - One to many communication

Outline

- The Problem of Message Loss
- The TCP/IP Solution
- Multicast commu

The Problem of Message Loss

Message loss

Cause

- Networks are in general unreliable
- Messages can be lost (never been delivered even if sent)

- Examples:
  - A server receives too many request y so it cannot treat all
  - A router drops the message because its queue is full

Message losses may impact the computation of a distributed system

Message loss

Coordinated Attack Problem

- Constraints of the problem
  - Two armies, ea e mountains surrounding a battlefield (dist
  - Can only communicate via messeng assing)
  - Messengers can be killed before r tion (message losses)
- Goal: they want to coordinate an attack
  - If they attack at different times, they both die
  - If they attack at the same time, they win

Message loss

Coordinated Attack Problem (con't)

- There is no protocols to make sure they will win!

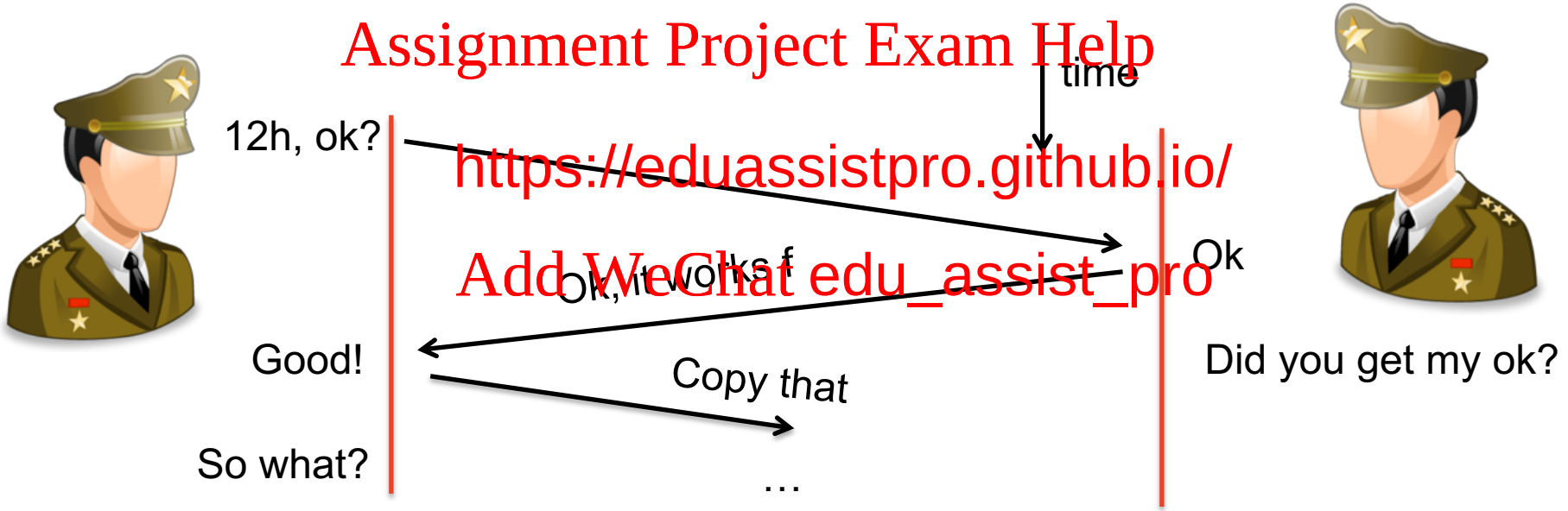
12h!

time

Message loss

Coordinated Attack Problem (con't)

- There is no protocols to make sure they will win!



Message loss

Analogy in networking

- Constraints of the problem
  - Two remote entities of a distributed system
  - Can only communicate through messages
  - The network is u pped
- Goal: they want to make sure to g simultaneously

This is impossible, even if all messages go through

TCP/IP

Communication 2/2
Week 4, COMP3221

THE UNIVERSITY OF SYDNEY

TCP: Overview RFCs: 793,1122,1323, 2018, 2581

- Point-to-point:
  - one sender, one receiver
- Connection-oriented, reliable, in-order byte stream:
- Full duplex dat
  - bi-direction ction
- Flow controlled:
  - sender will not overwhelm receiver
- Congestion controlled:
  - TCP congestion and flow control set window size

TCP seq. numbers, ACKs

sequence numbers:

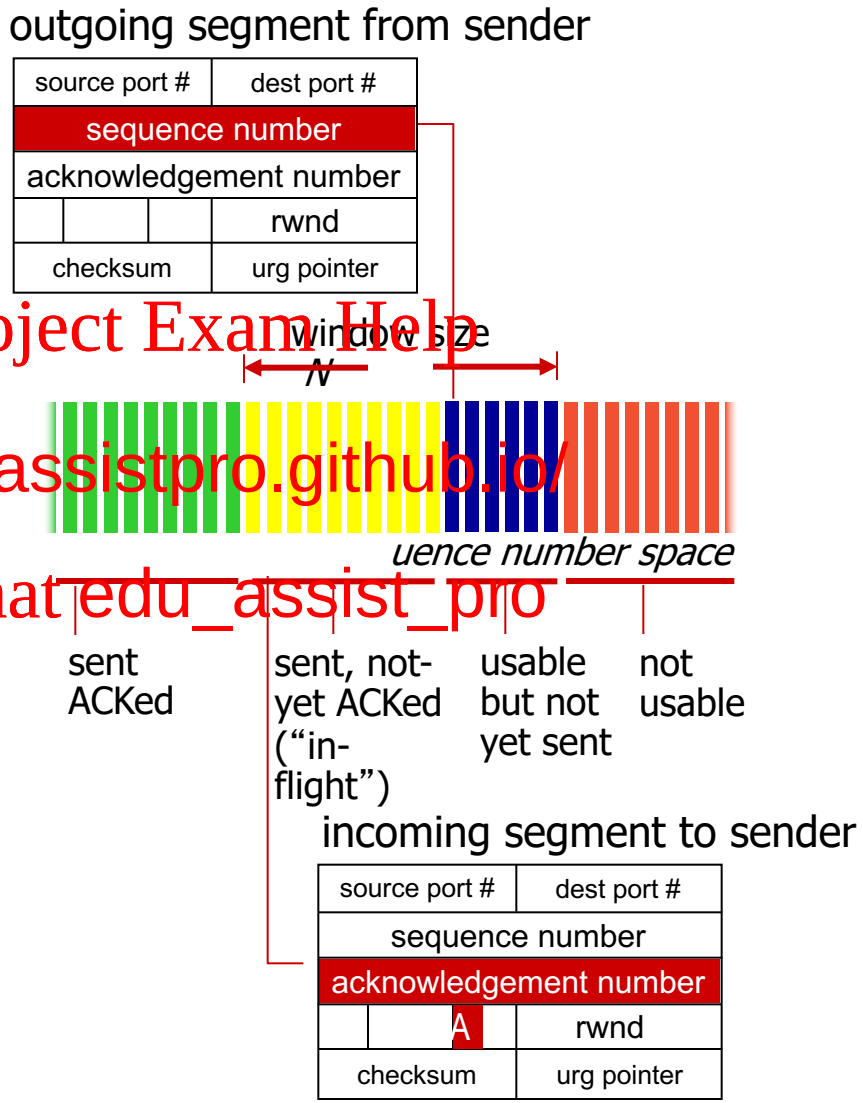
- byte stream "number" of first byte in segment's data

acknowledgement

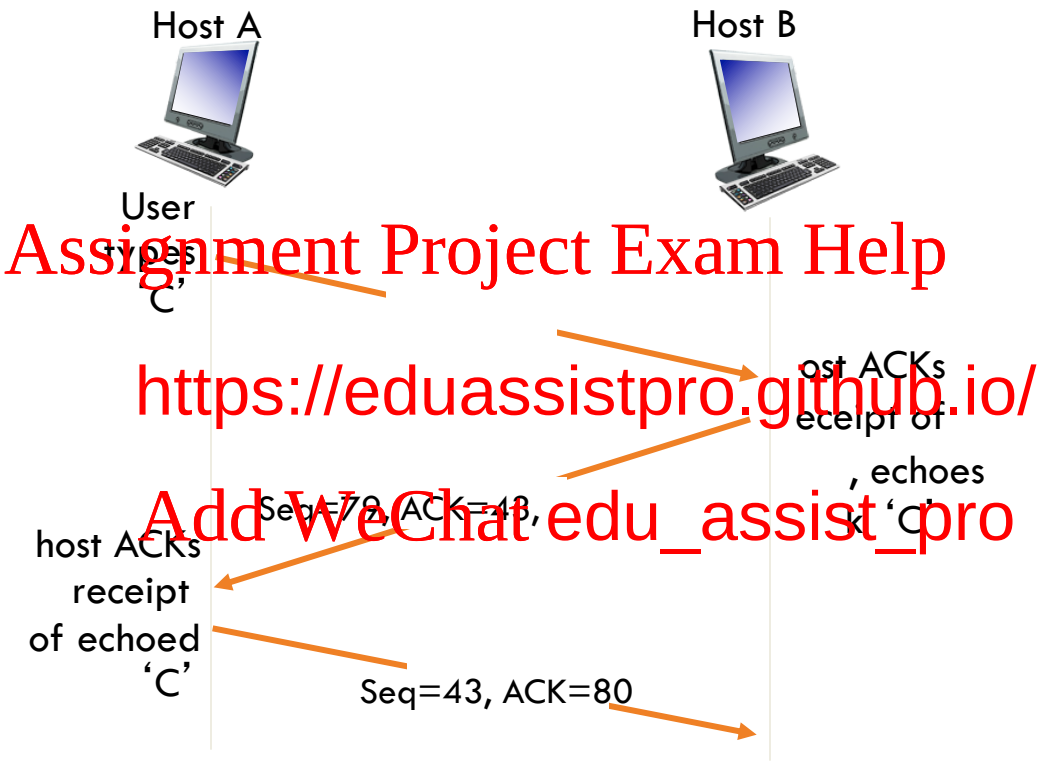
- seq # of next b expected from other side
- cumulative ACK

Q: how receiver handles out-of-order segments

- A: TCP spec doesn't say, - up to implementor



TCP seq. numbers, ACKs



simple telnet scenario

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 running
  - think of timer as for oldest unacked segment
  - expiration interval: `TimeOutInterval`

timeout:

- retransmit segment that caused timeout
- restart timer

ack rcvd:

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

TCP round trip time, timeout

Q: how to set TCP timeout value?

- longer than RTT
  - but RTT vari
- *too short:* prem timeout, unnecessary retransmissions
- *too long:* slow reaction to segment loss

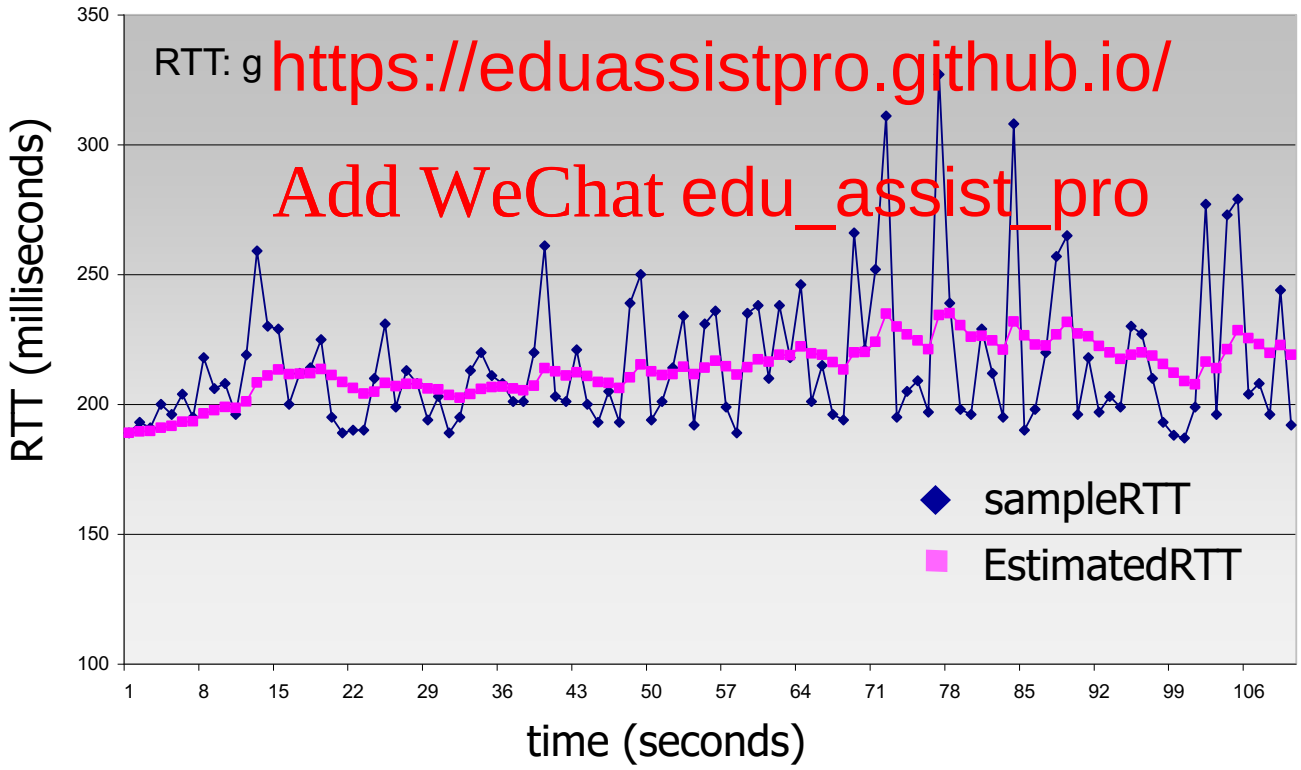
Q: how to estimate RTT?

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

TCP round trip time, timeout

`EstimatedRTT = (1- α)*EstimatedRTT + α*SampleRTT`

- exponential weighted moving average
- influence of past sample decreases exponentially fast
- typical value: α = 0.125



TCP round trip time, timeout

- timeout interval: **`EstimatedRTT`** plus "safety margin"
  - large variation in **`EstimatedRTT`** -> larger safety margin
- estimate SampleRTT deviation from EstimatedRTT:


```
DevRTT = (
```



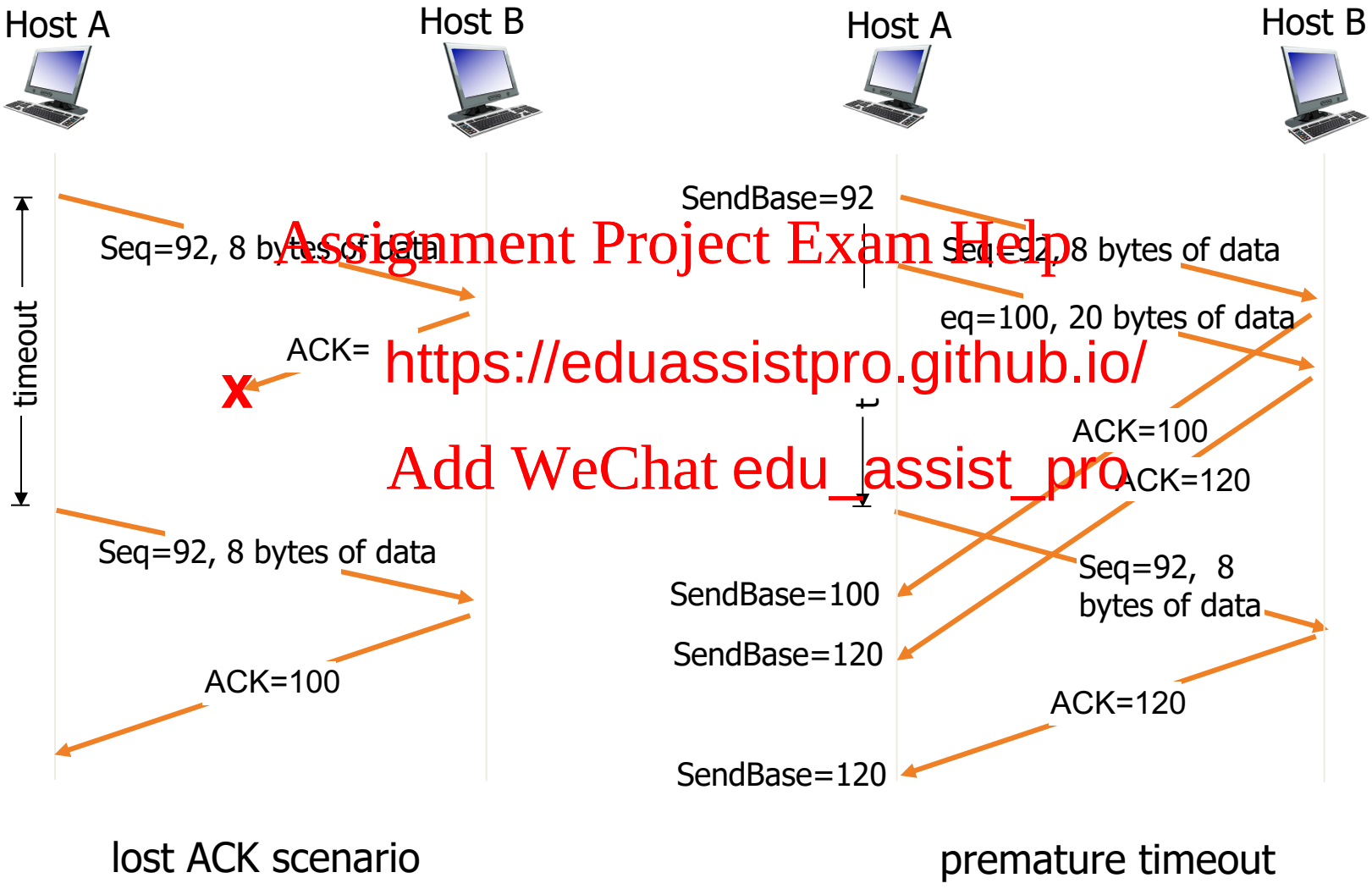
```
    (typically,
```


`TimeoutInterval = EstimatedRTT + 4*DevRTT`

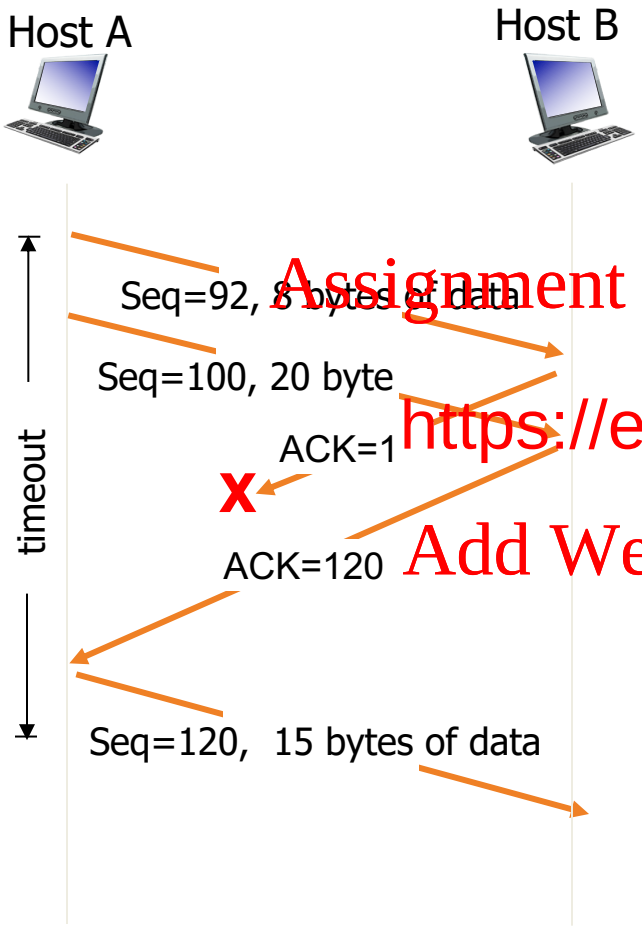
estimated RTT

"safety margin"

TCP: retransmission scenarios



TCP: retransmission scenarios



cumulative ACK

TCP fast retransmit

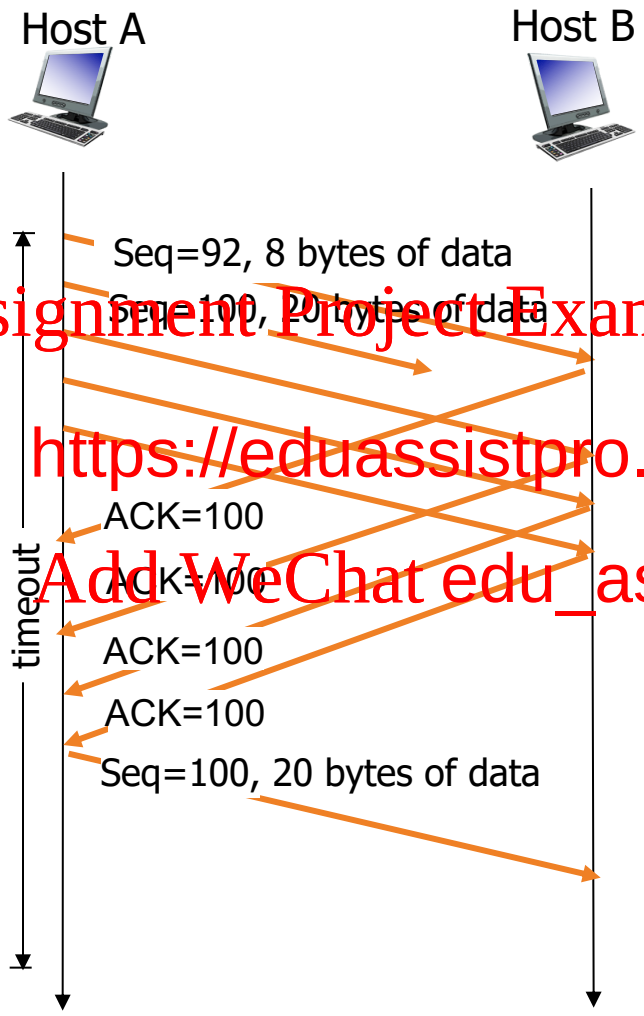
- time-out period often relatively long:
  - long delay before resending a lost packet
- detect lost segm duplicate ACKs.
  - sender often sends many segments back-to-back
  - if segment is lost, there will likely be many duplicate ACKs.

TCP fast retransmit

if sender receives 3 ACKs for same data uplicate ACKs"), nacked ith smallest

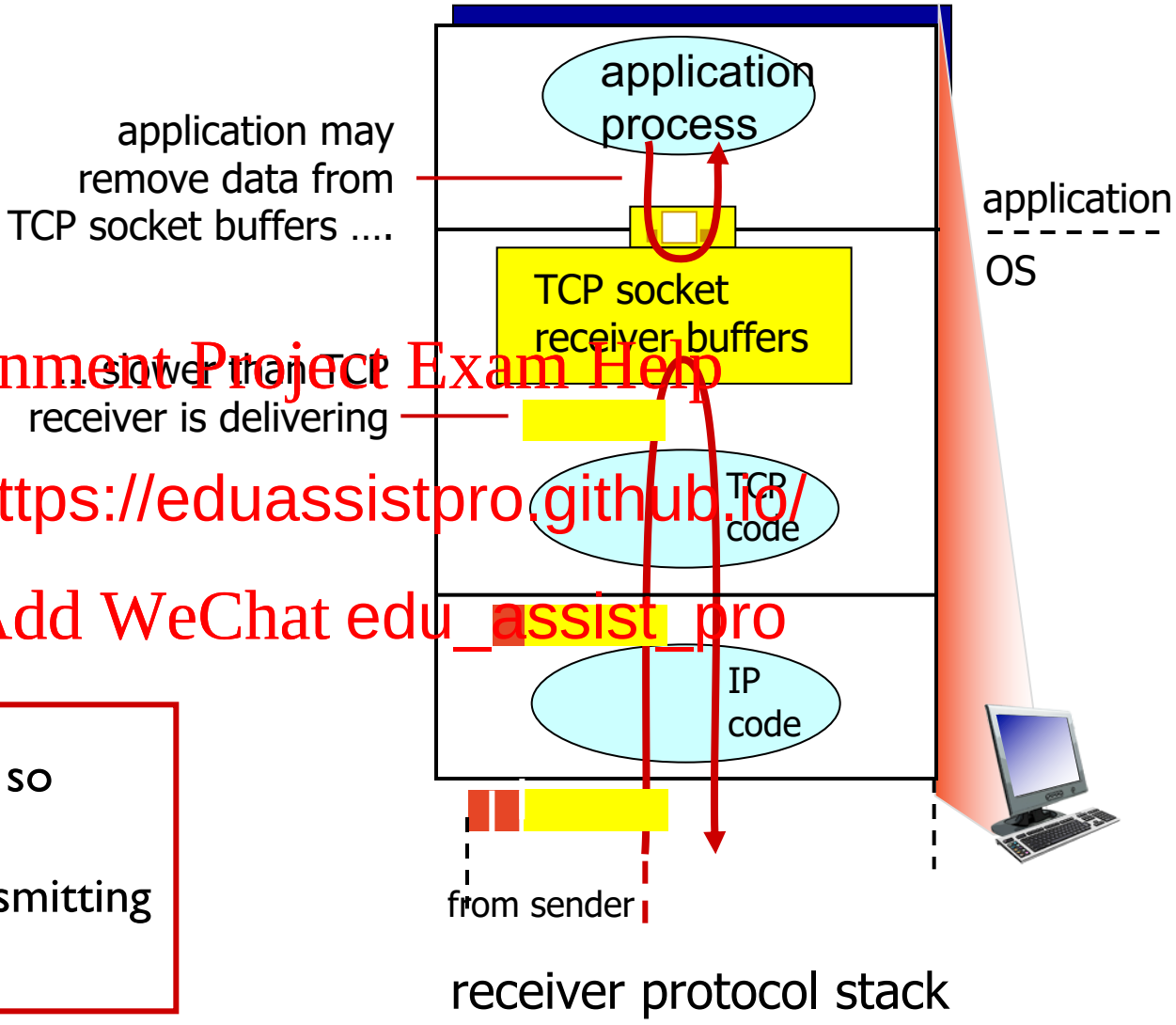
- likely that unacked segment lost, so don't wait for timeout

TCP fast retransmit



fast retransmit after sender receipt of triple duplicate ACK

TCP flow control

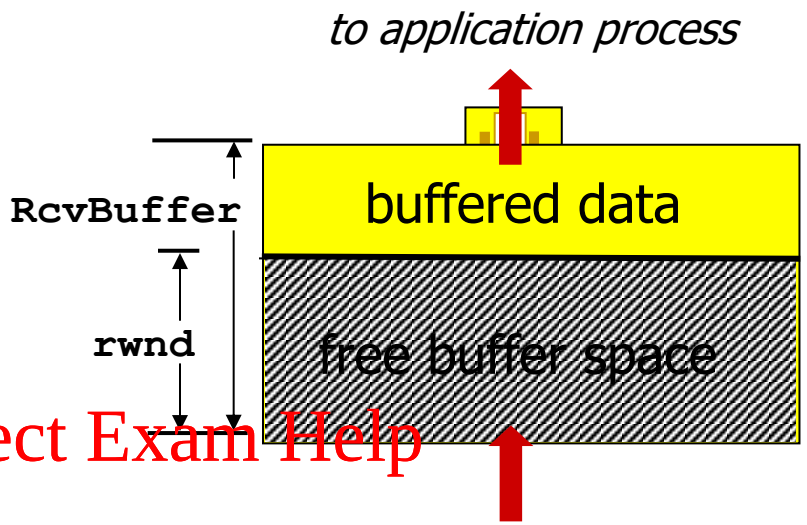


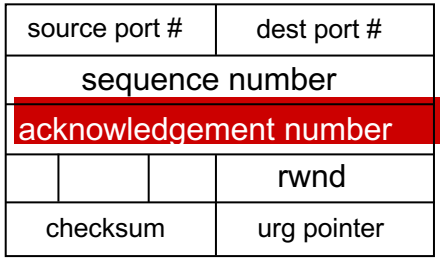
flow control

receiver controls sender, so sender won't overflow receiver's buffer by transmitting too much, too fast

TCP flow control

- receiver "advertises" free buffer space by including **rwnd** value in TCP header of receiver-to-sender segments
  - **RcvBuffer** si socket options (t 4096 bytes)
  - many operating systems auto adjust **RcvBuffer**
- sender limits amount of unacked ("in-flight") data to receiver's **rwnd** value
- guarantees receive buffer will not overflow





Principles of congestion control

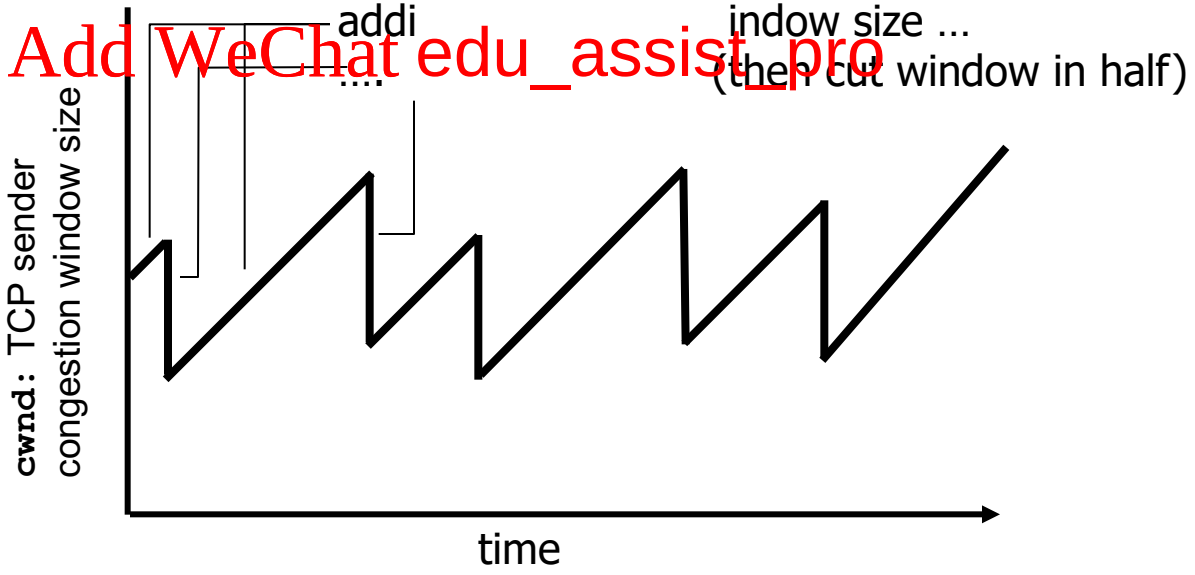
congestion:

- informally: "too many sources sending too much data too fast for *network* to handle"
- different from f
- manifestations:
  - lost packets (buffer over ters)
  - long delays (queueing in router buffers)
- a top-10 problem!

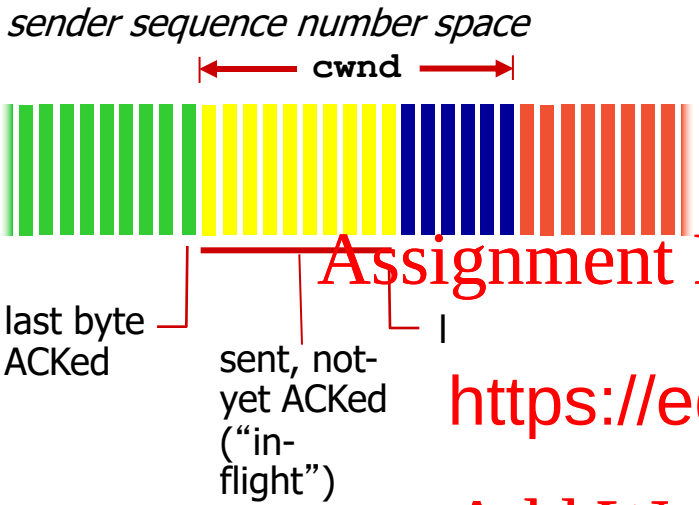
TCP congestion control

- *approach:* sender increases transmission rate (window size), probing for usable bandwidth, until loss occurs
  - *additive increase:* increase **cwnd** by 1 MSS (Maximum Segment Size) every RTT until loss detected
  - *multiplicativ* n half after loss

AIMD saw tooth behavior: probing for bandwidth



TCP Congestion Control: details



TCP sending rate:

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

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

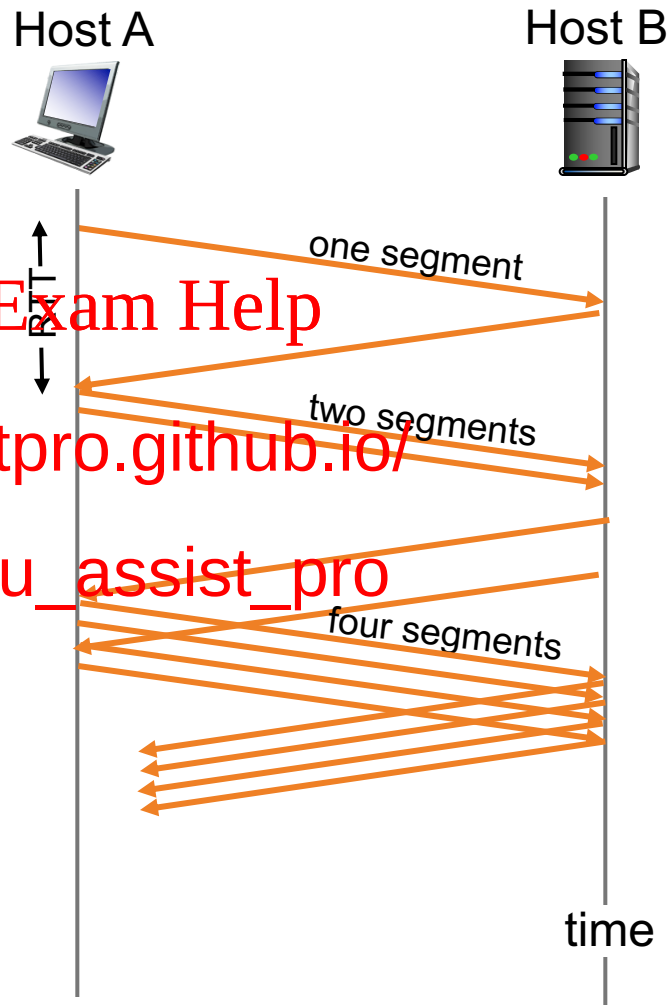
- sender limits transmission:

$$\text{LastByteSent} - \text{LastByteAcked} \leq \text{cwnd}$$

- Congestion window **(cwnd)** is dynamic function of perceived network congestion

TCP Slow Start

- when connection begins, increase rate exponentially until first loss event:
  - initially **cwnd** = 1 MSS (Maximum S
  - double **cwn**
  - done by incrementing **cwnd** for every ACK received
- *summary:* 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, then grows linearly

- Loss indicated b
  - dup ACKs indicate network capa ing some segments
  - **cwnd** is cut in half window then

- TCP Tahoe always sets **cwnd** to 1 (timeout or 3 duplicate acks)

TCP: switching from slow start to Congestion Avoidance

Q: when should the exponential increase switch to linear?

A: when **cwnd** gets to 1/2 of its value before timeout.

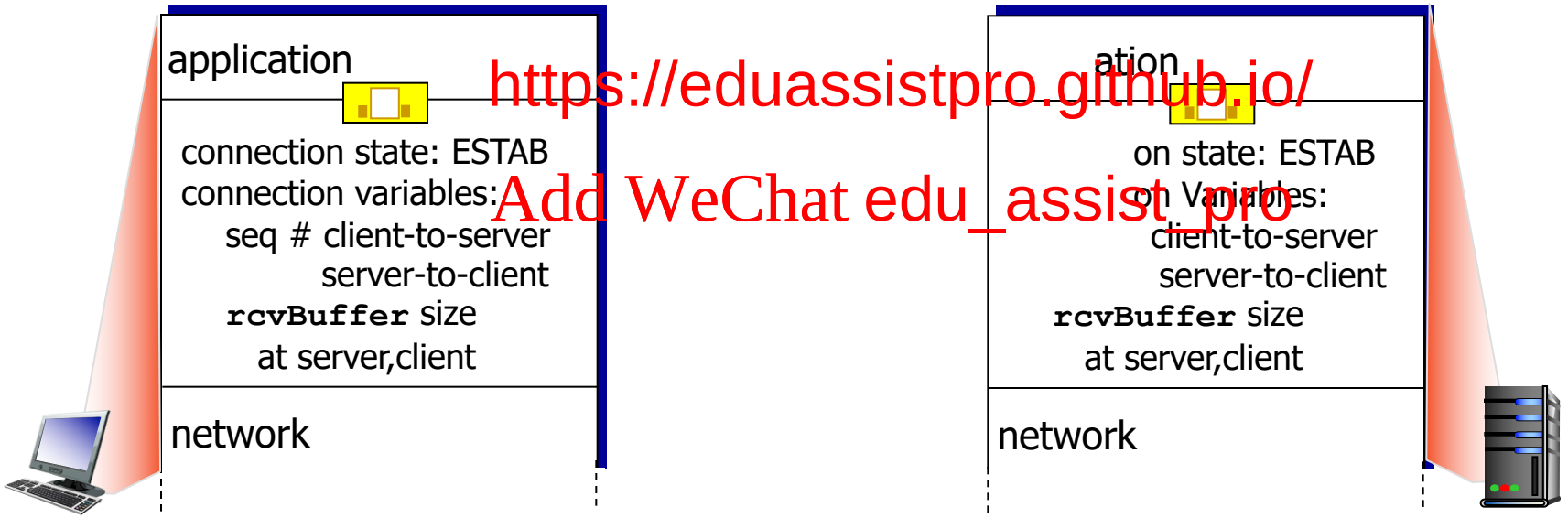
Implementation:

- variable **ssthresh**
- on loss event, **ssthresh** is set to 1/2 of **cwnd** just before loss event

Connection Management

before exchanging data, sender/receiver "handshake":

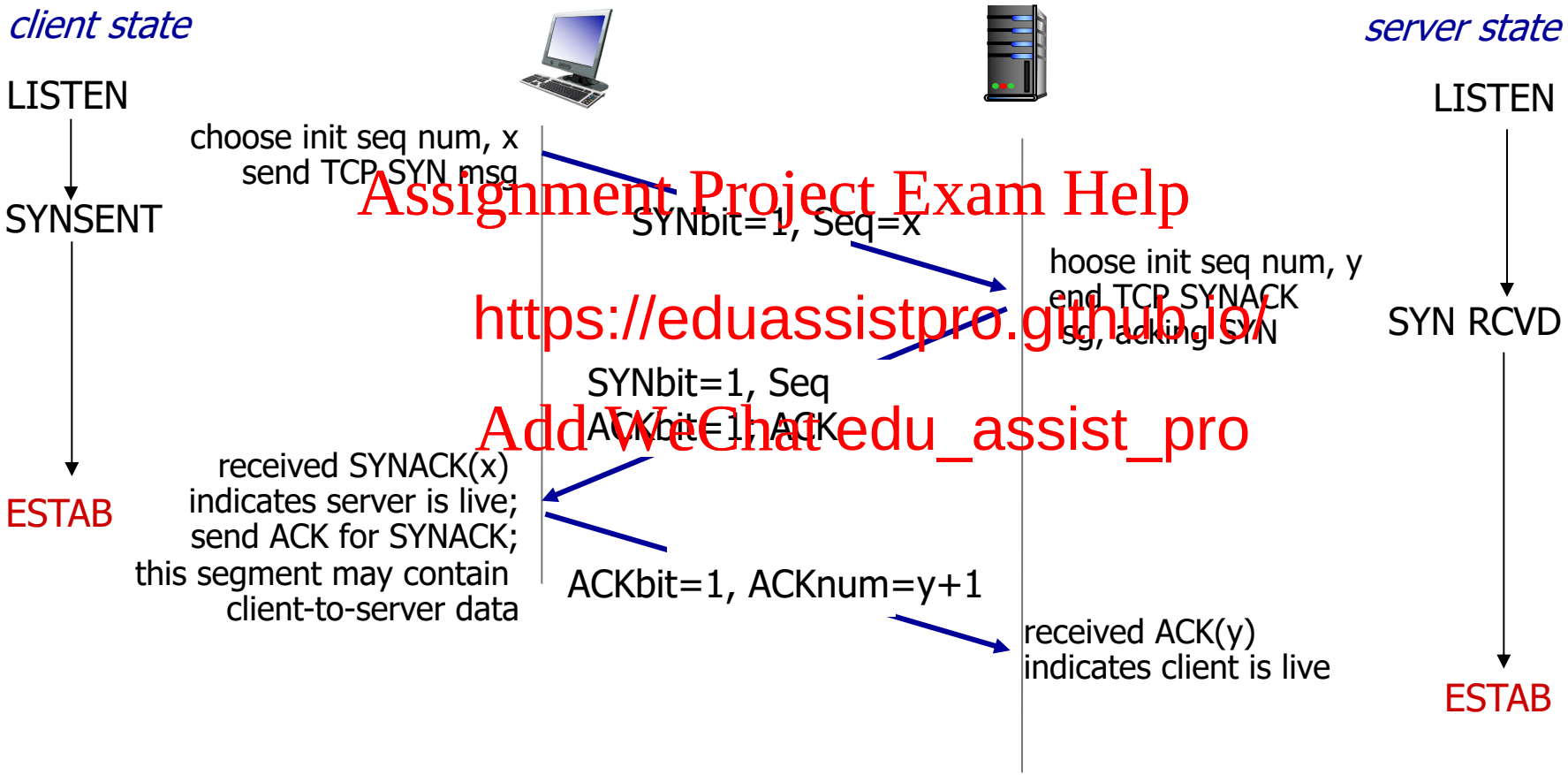
- agree to establish connection (each knowing the other willing to establish connection)
- agree on connection parameters, e.g. MSS, rwnd, etc.
- connection-oriented



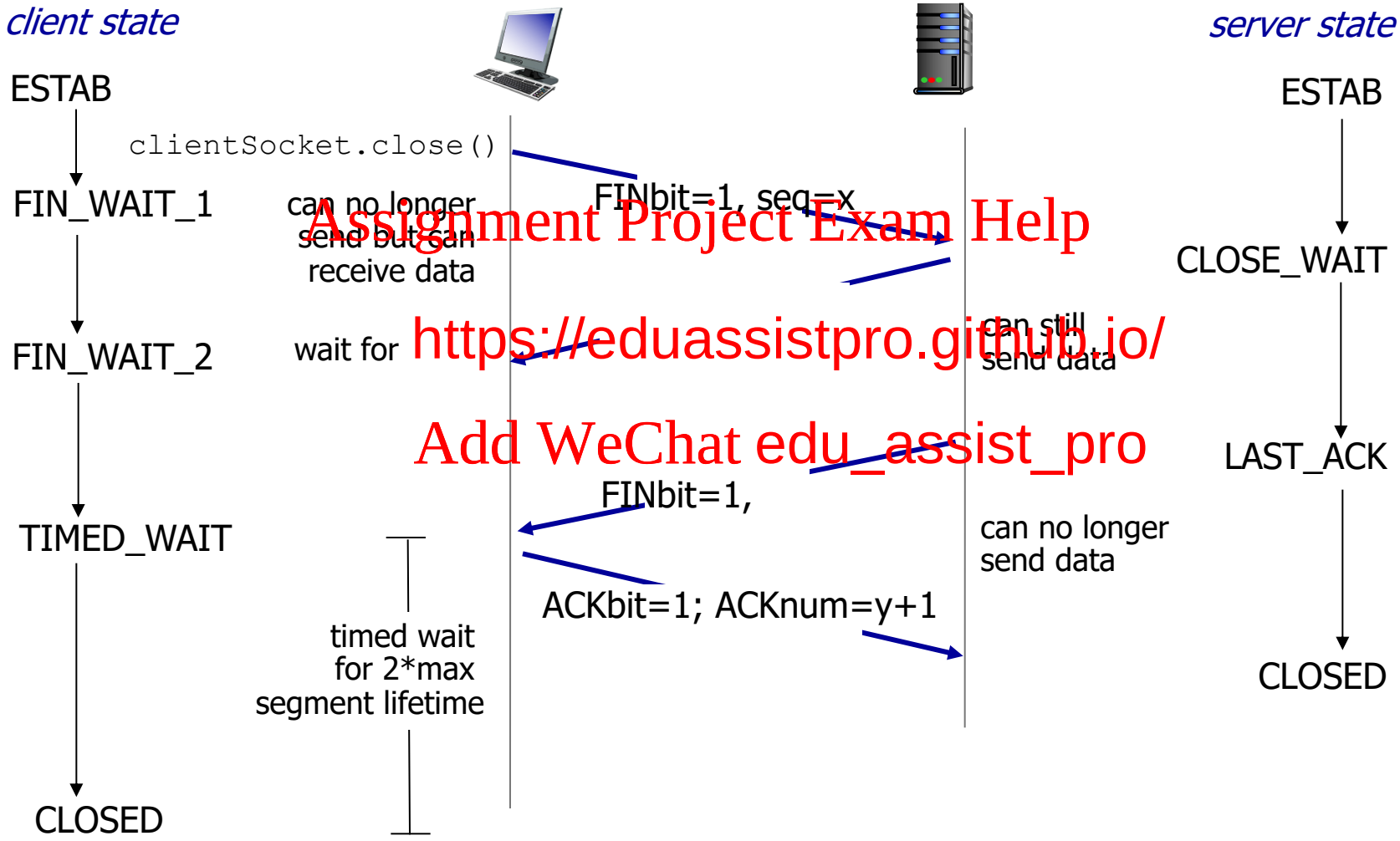

```
Socket clientSocket =
  newSocket("hostname","port
   number");
```

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

TCP 3-way handshake



TCP: closing a connection



HTTP Overview

HTTP: hypertext transfer protocol

- Web's application layer protocol
- client/server model
  - *client:* browser t receives, (using and "displays"
  - *server:* Web server sends (using HTTP protocol) objects in response to requests

HTTP overview

Uses TCP:

- client initiates TCP connection (creates socket) to server, port 80
- server accepts T connection from
- HTTP messages (application-layer protocol messages) exchanged between browser (HTTP client) and Web server (HTTP server)
- TCP connection closed

HTTP is "stateless"

- server maintains no information about past client requests

TTPS ?

ver TLS
ort Layer Security), e.g. over SSL

Related terminology

- Broadcast: one-to-all communication
  - Action of sending a message to all nodes of the system
  - Typically used for relatively small systems, like IP broadcast as part of Ethernet in LANs
- Unicast: one-to-one communication
  - In contrast with b of distribution to a single receiver
  - Used generally i eaming applications
- Anycast: one-to-random-one commu
  - Send a message to an IP address ran am a response from any potential receiver, whose IP address belongs to this range
  - Used with UDP and TCP
- Multicast: one-to-many communication
  - Action of sending a message to multiple nodes of the system (not necessarily all nodes)
  - This term is used in many contexts (network, algorithm)

Conclusion (Transportation layer)

- Message losses can have dramatic consequences
- TCP/IP protocol suite hides these losses from the application level
- Socket, RPC, use
  - Sockets are complex
  - RPC is more transparent for the client
- Multicast communication scales well for large distributed systems