Lecture 8:
Transport layer

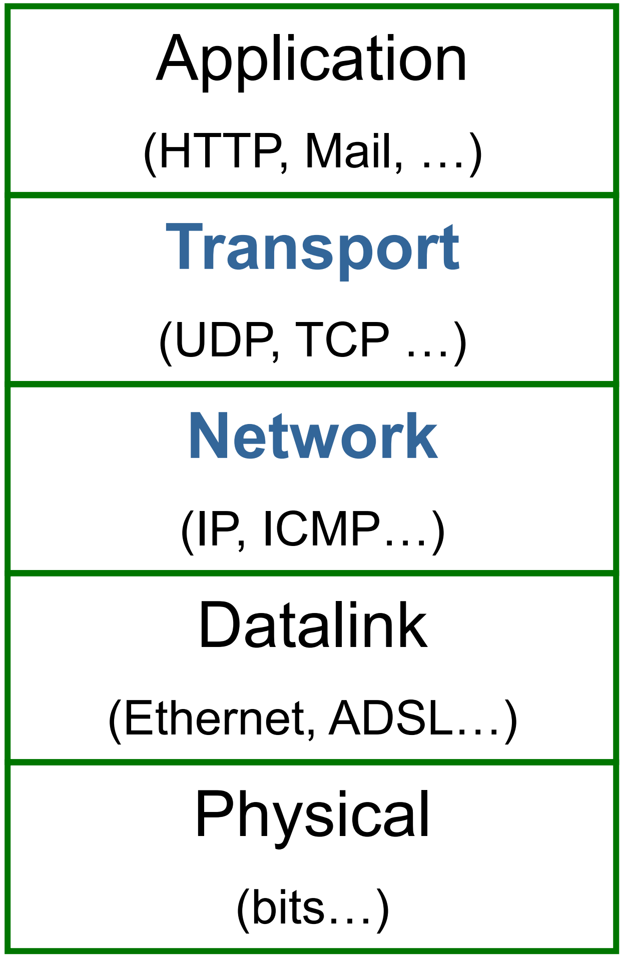
Reading 6.2, 6.3, 6.4, 6.5

Computer Networks, Tanenbaum

Contents

- Principles of transport layer
- UDP protocol
- TCP protocol

Transport layer in OSI model



Application

(HTTP, Mail, …)

Transport

(UDP, TCP …)

Network

(IP, ICMP…)

Datalink

(Ethernet, ADSL…)

Physical

(bits…)

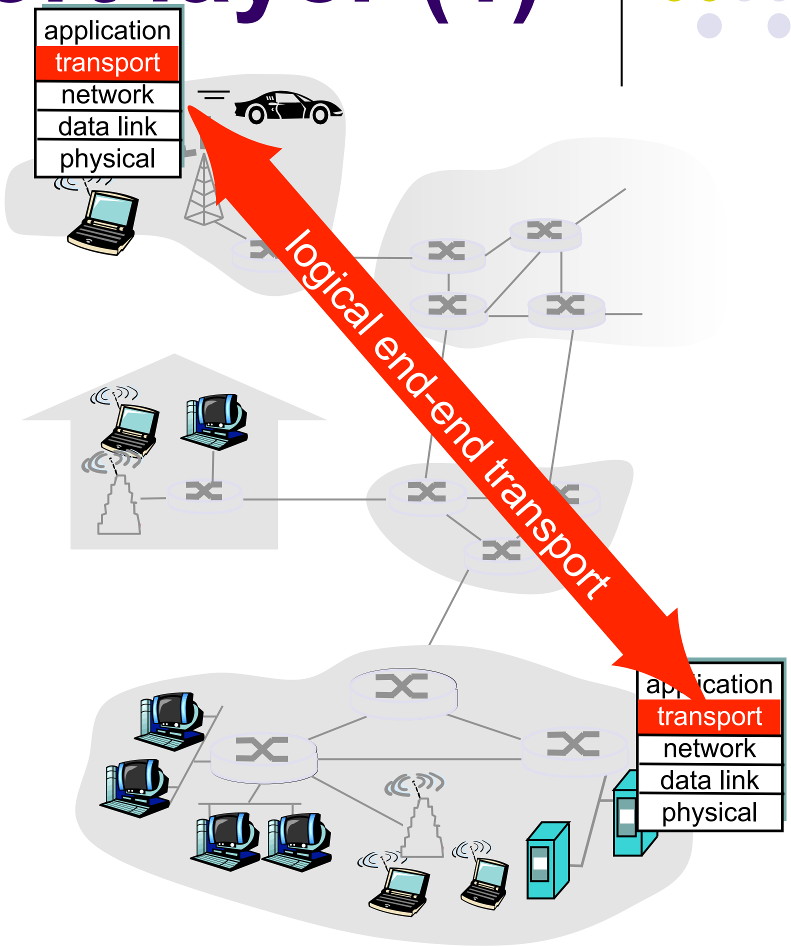
Support applications

Transferring data between applications

Routing and forwarding data between hosts

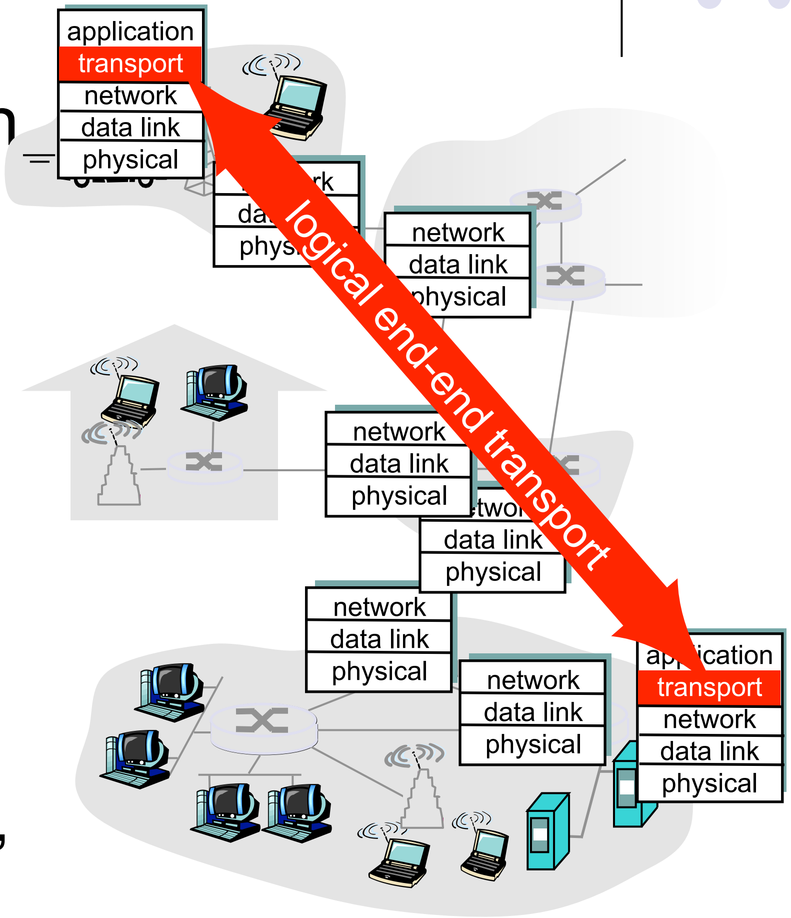
Principle of transport layer (1)

- Provide transport means between end applications
- Sender:
  - Receives data from application
  - Place data in segments and give to network layer
  - If the data size is too big, it is divided into many segments
- Receiver:
  - Receives segments from network layer
  - Reconstitute data from segments and deliver to the application



Principle of transport layer (2)

- Transport layer is installed in end systems
  - Not installed in routers, switches…
- Two kinds of transport layer services
  - Reliable, connection-oriented, e.g TCP
  - Not reliable, connectionless, e.g. UDP



Why two kinds of service?

- Requirements from application layer are various
- Some applications need transport service with 100% fiability such as mail, web…
  - Should use TCP transport service
- Some applications need to transmit data as fast as possible, with some fault tolerance, e.g. VoIP, Video Streaming
  - Should use UDP transport service

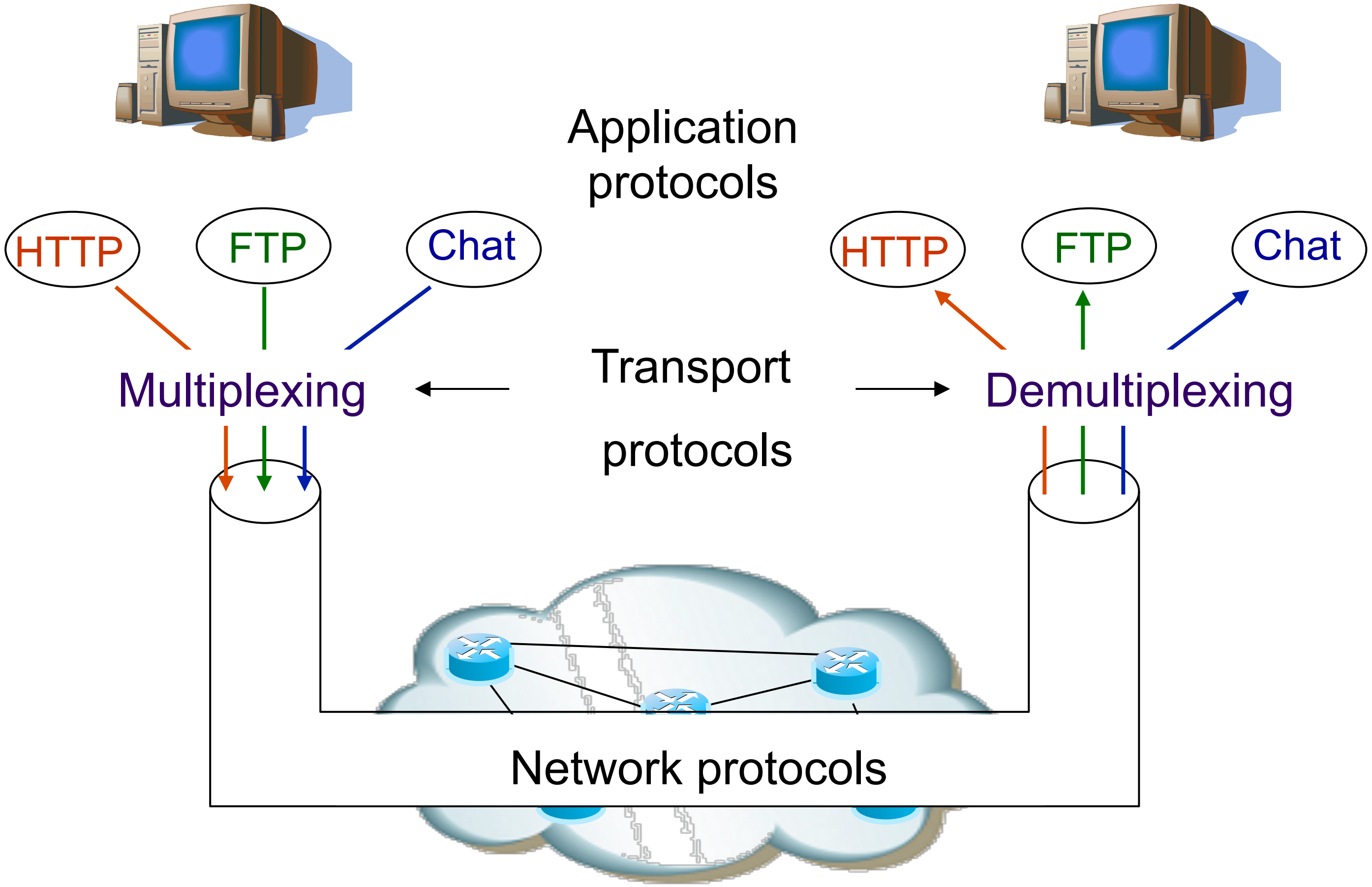
Applications and transport services

Application	Application protocols	Transport protocols
e-mail	SMTP	TCP
remote terminal access	Telnet	TCP
Web	HTTP	TCP
file transfer	FTP	TCP
streaming multimedia	Specific protocols (e.g. RealNetworks)	TCP or UDP
Internet telephony	Specific protocols (e.g., Vonage,Dialpad)	Usually UDP

Functionalities

MUX/DEMUX

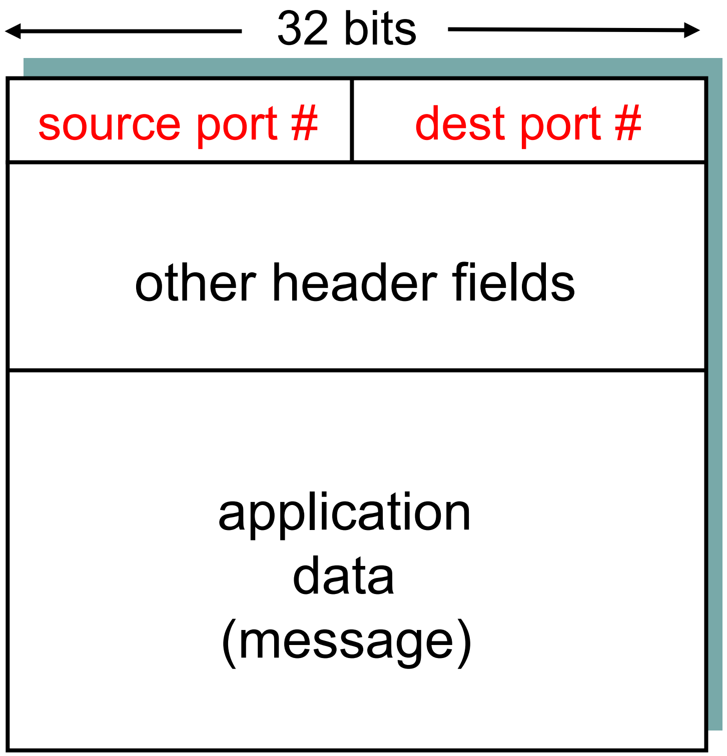
Mux/Demux



Application protocols
HTTP
FTP
Chat
HTTP
FTP
Chat
Multiplexing
Transport protocols
Demultiplexing
Network protocols

How does it Mux/Demux?

- How to distinguish applications running in the same hosts?
  - Use an identifier called port number (16 bits)
  - Each process is assigned a port
- Socket: A pair of IP address and port
  - Socket identifies an unique application process all over the world



TCP/UDP segment format

Checksum

- Phát hiện lỗi bit trong các đoạn tin/gói tin
- Nguyên lý giống như checksum (16 bits) của giao thức IP
- Ví dụ:

```
            1 1 1 0 0 1 1 0 0 1 1 0 0 1 1 0
            1 1 0 1 0 1 0 1 0 1 0 1 0 1 0 1

          1 1 0 1 1 1 0 1 1 1 0 1 1 1 0 1 1
          └───────────────────────────────┘

Tổng        1 0 1 1 1 0 1 1 1 0 1 1 1 1 0 0
Checksum    0 1 0 0 0 1 0 0 0 1 0 0 0 0 1 1
```

UDP
User Datagram Protocol

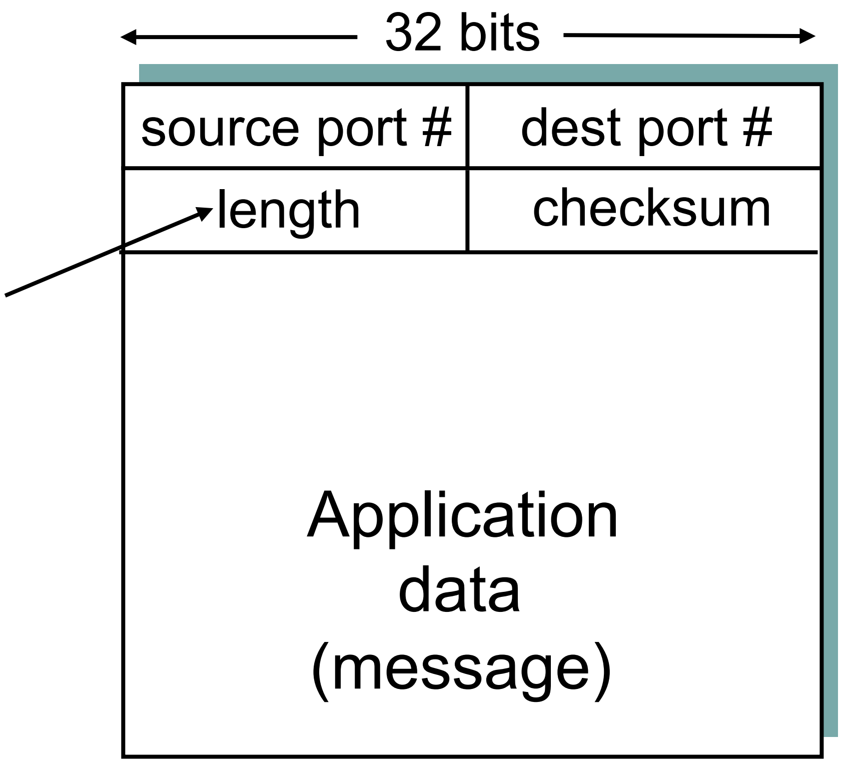
“Best effort” protocols

- Why UDP?
  - No need to establish connection (cause delay)
  - Simple
  - Small header
  - No congestion control → send data as fast as possible

- Main functionality of UDP?
  - MUX/DEMUX
  - Detect error by checksum

Datagram format

- Data unit in UDP is called datagram



32 bits

source port # | dest port #

length | checksum

Application data (message)

Length of the datagram in byte

Issues of UDP

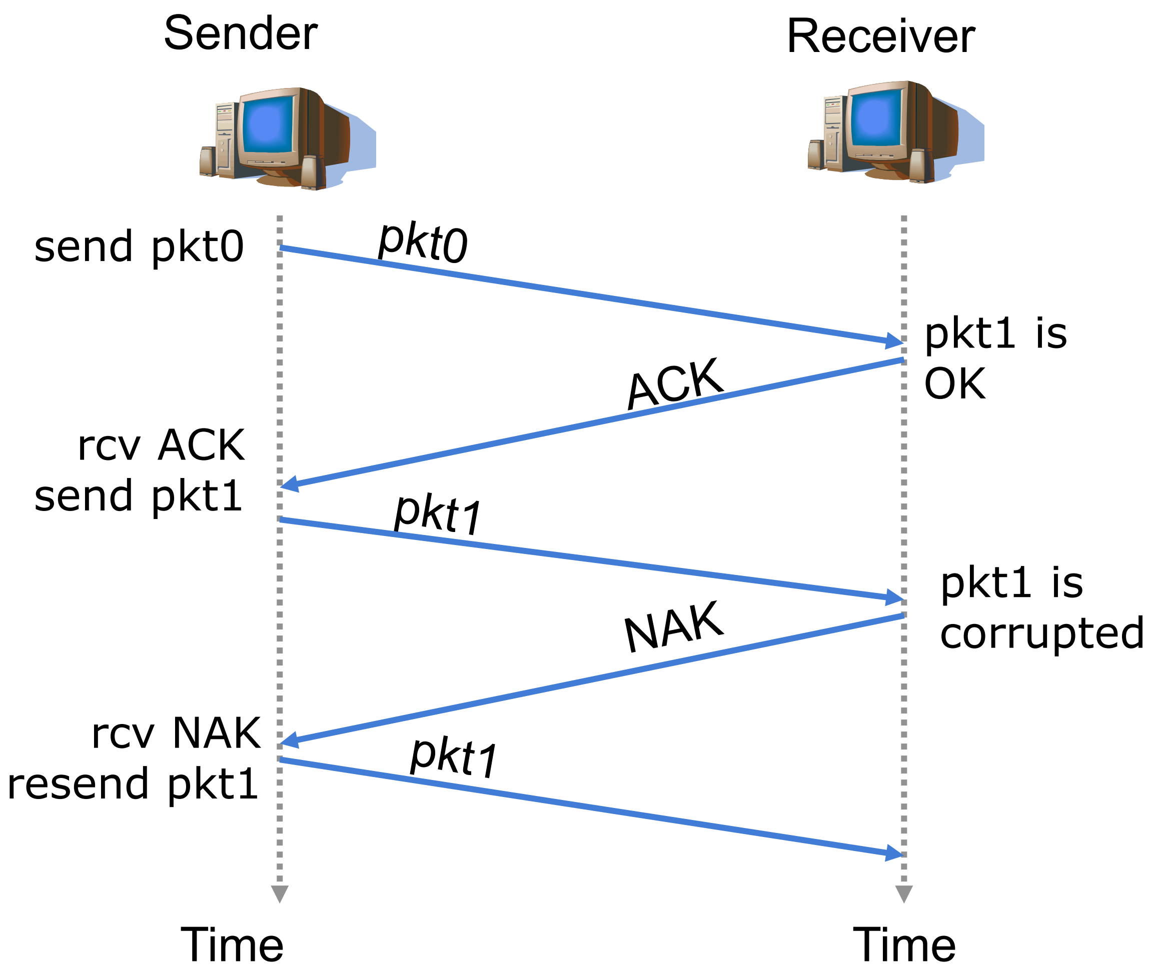
- No congestion control
  - Cause overload of the Internet
- No reliability
  - Applications have to implement themselves mechanisms to control errors

Error control

Error control

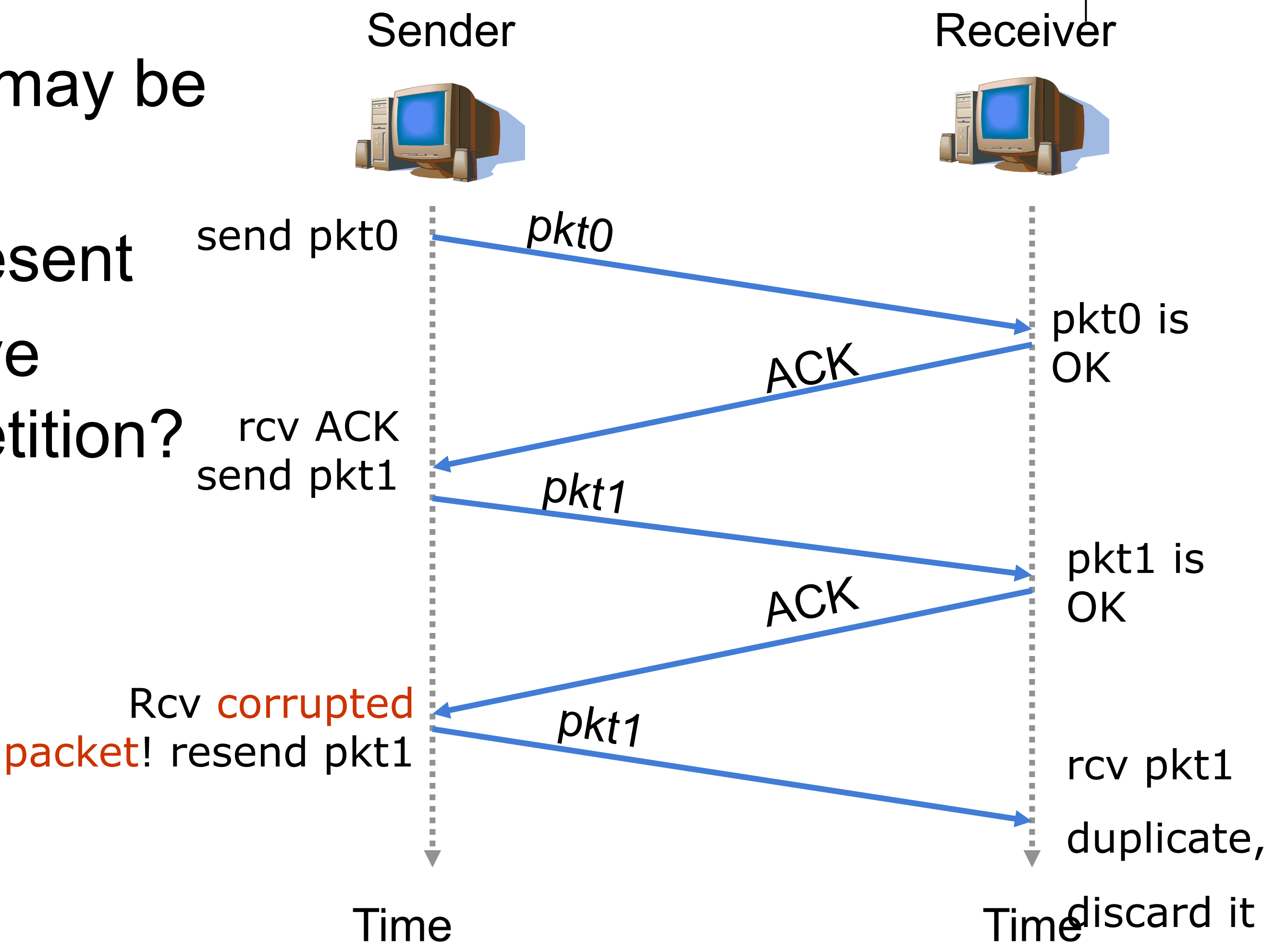
- How to detect error?
  - Checksum
- How to inform sender?
  - ACK (*acknowledgements*):
  - NAK (*negative acknowledgements*): tell sender that pkt has error
- Reaction of sender?
  - Retransmit the error packet once received NAK

Error control

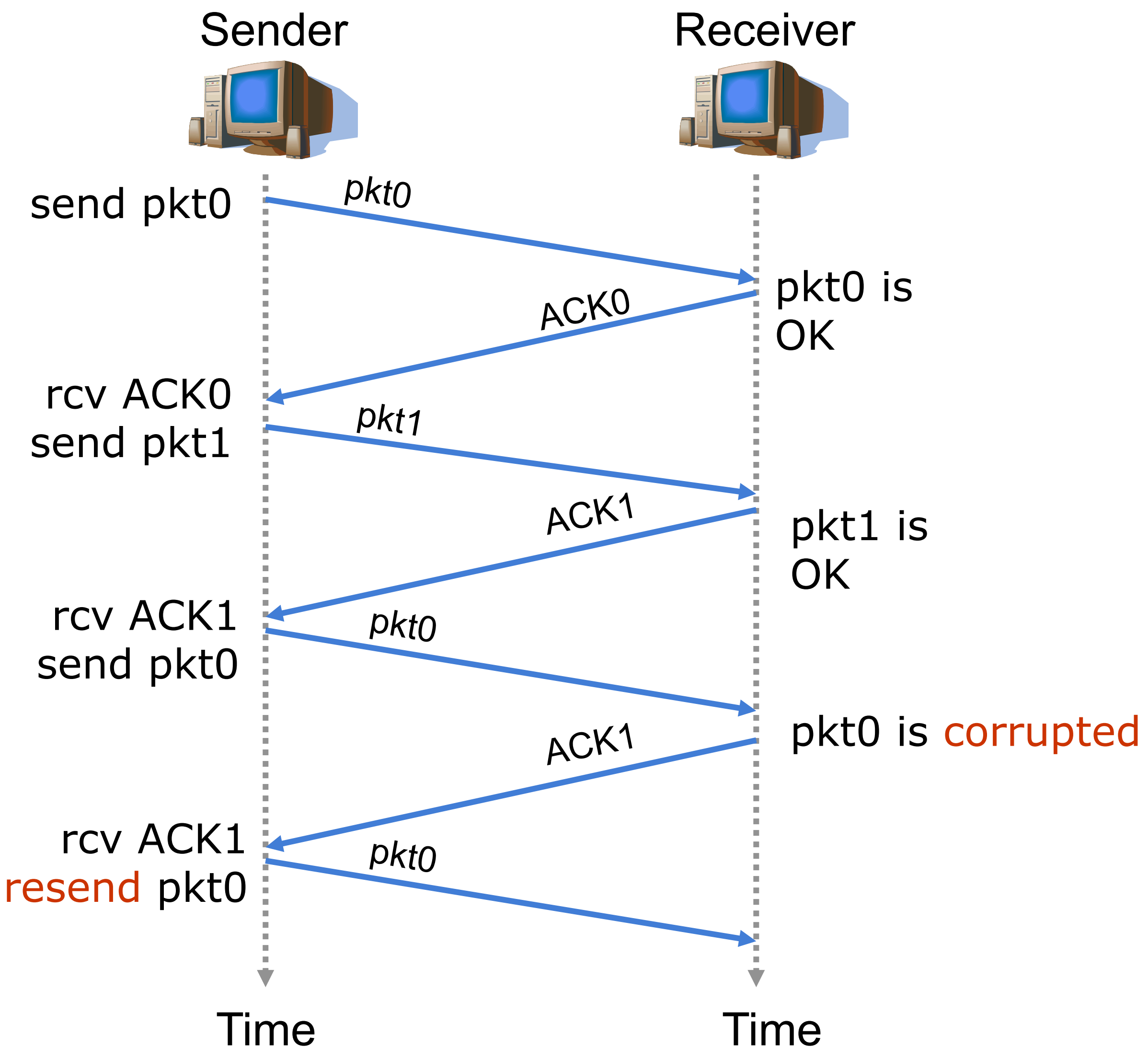


Error in ACK/NAK

- ACK/ NAK may be corrupted
- Packet is resent
- How to solve packet repetition?
- Use Seq.#



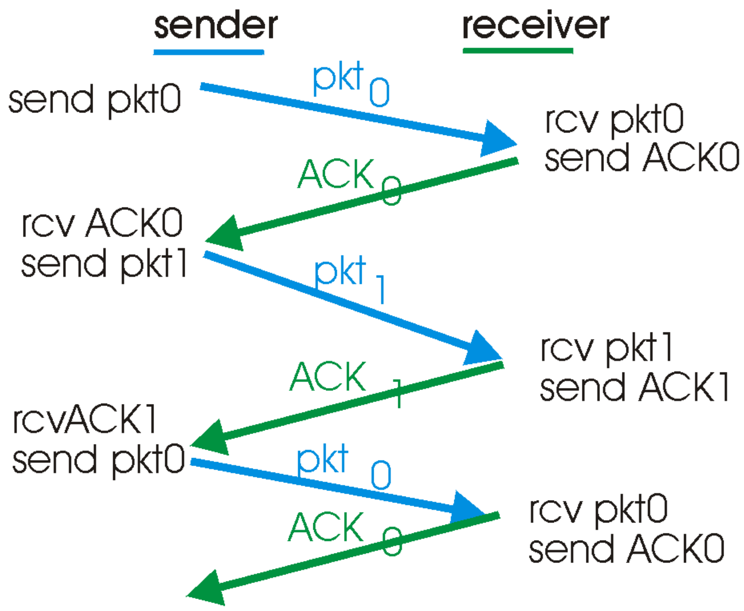
Error control without NAK



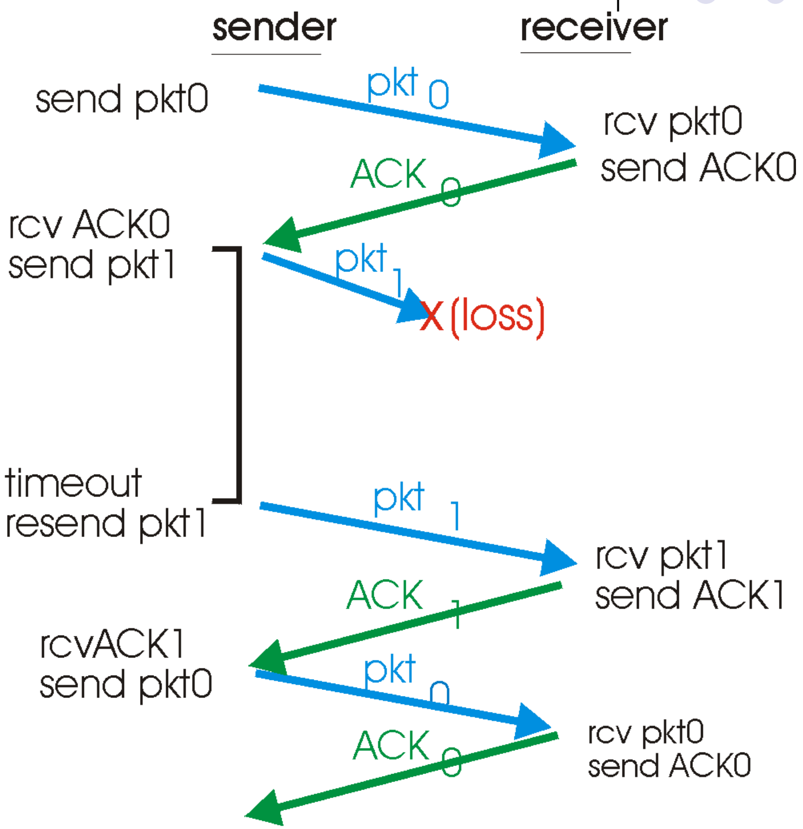
Chanel with error and packet lost

- Data and ACK can be lost
  - If no ACK is received?How sender knows and decides to resend data?
  - Sender should wait for ACK for a certain time. Timeout!
- How long should be timeout?
  - At least 1 RTT (Round Trip Time)
  - Need to start a timer each time sending a packet
- What if packet arrives and ACK is lost?
  - Packet should be numbered.

Illustration

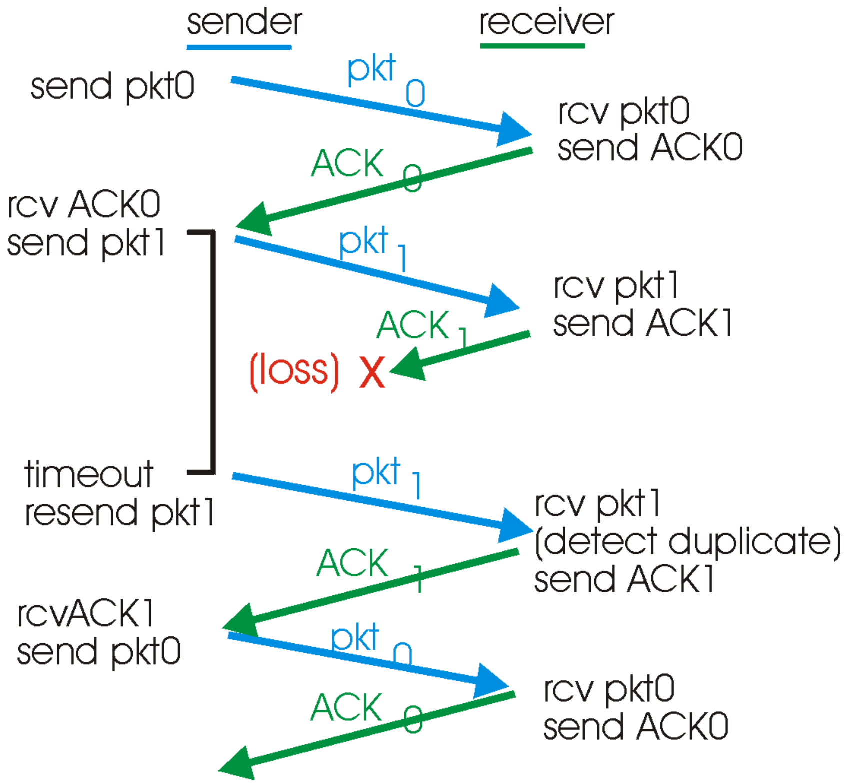


(a) operation with no loss



(b) lost packet

Illustration



sender

receiver

send pkt0

pkt 0

rcv pkt0
send ACK0

ACK 0

rcv ACK0
send pkt1

pkt 1

rcv pkt1
send ACK1

ACK 1

(loss) X

timeout
resend pkt1

pkt 1

rcv pkt1
(detect duplicate)
send ACK1

ACK 1

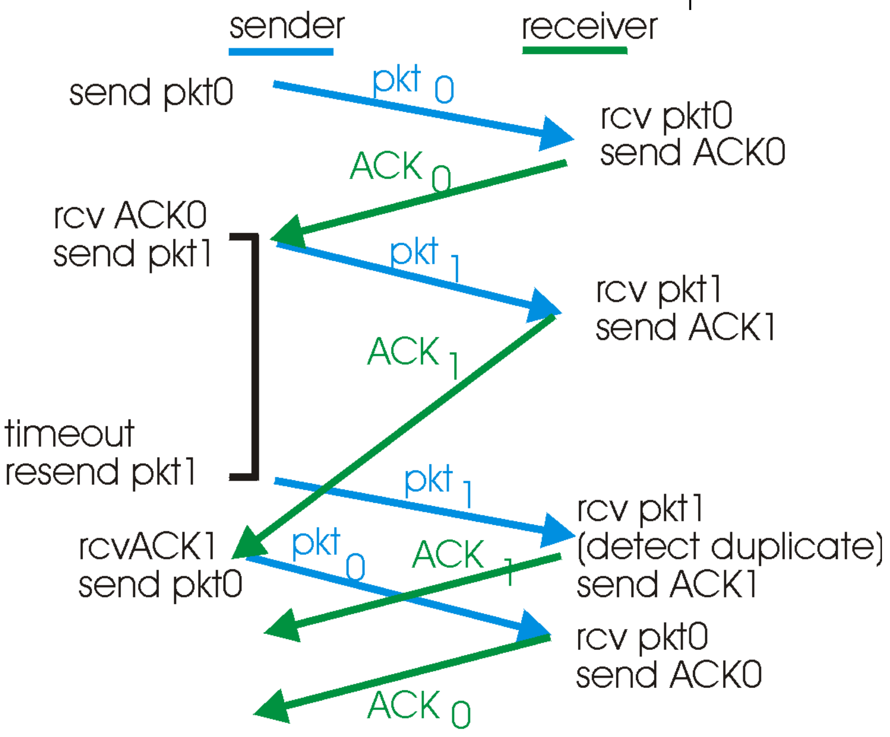
rcvACK1
send pkt0

pkt 0

rcv pkt0
send ACK0

ACK 0

(c) lost ACK



sender

receiver

send pkt0

pkt 0

rcv pkt0
send ACK0

ACK 0

rcv ACK0
send pkt1

pkt 1

rcv pkt1
send ACK1

ACK 1

timeout
resend pkt1

pkt 1

rcv pkt1
(detect duplicate)
send ACK1

rcvACK1
send pkt0

pkt 0

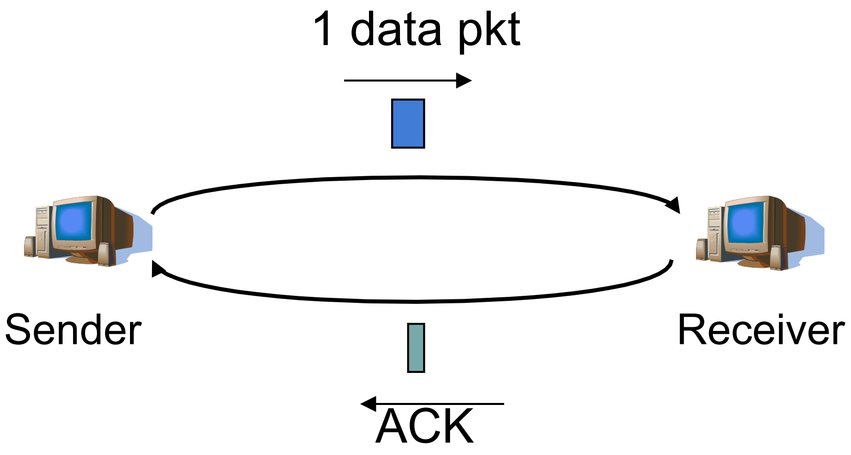
ACK 1

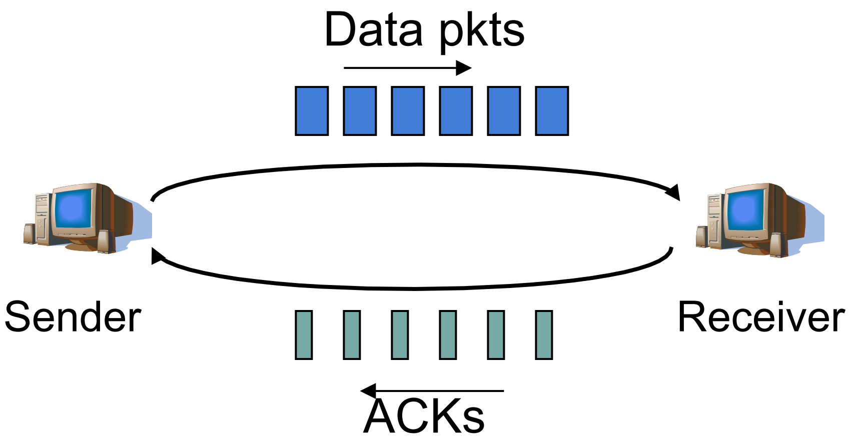
rcv pkt0
send ACK0

ACK 0

(d) premature timeout

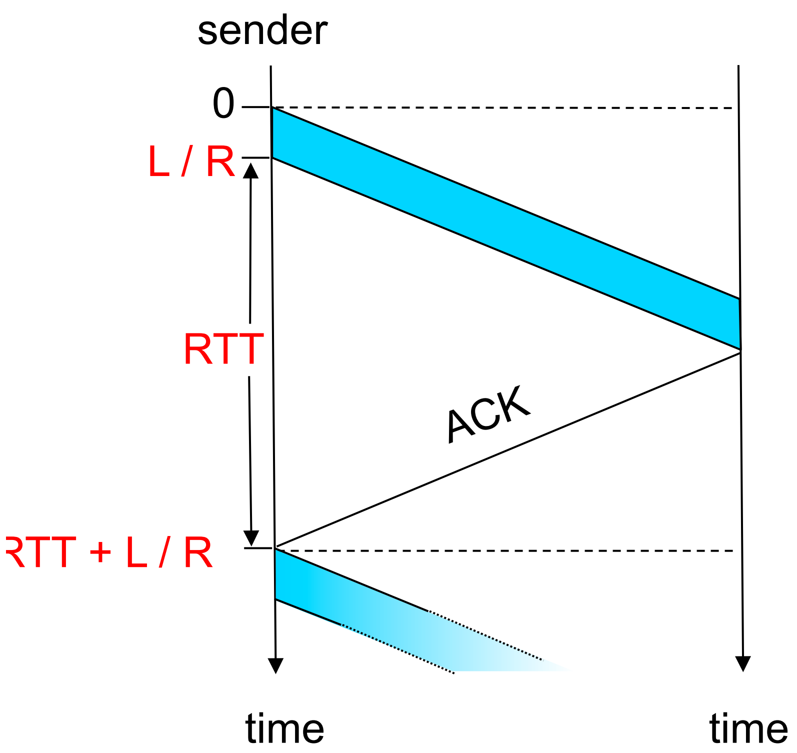
Transmission in pipeline



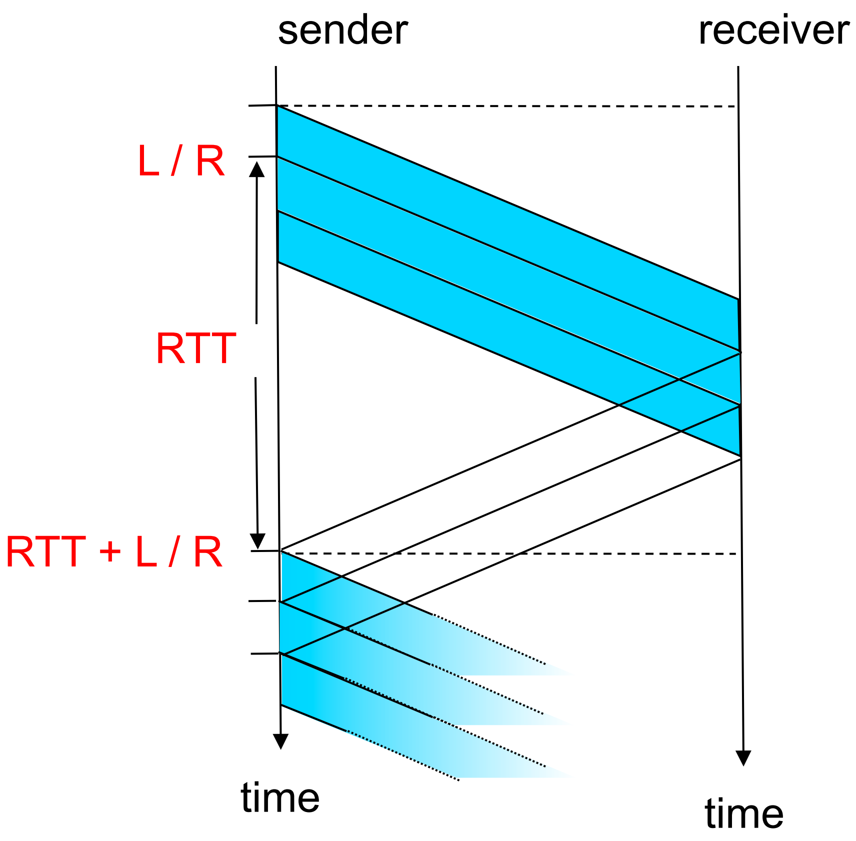


Comparison of efficiency

stop-and-wait



Pipeline



L: Size of data pkt
R: Link bandwidth
RTT: Round trip time

$$\text{Performance} = \frac{L / R}{RTT + L / R}$$

$$\text{Performance} = \frac{3 * L / R}{RTT + L / R}$$

TCP
Transmission Control Protocol

TCP segment structure

Connection management

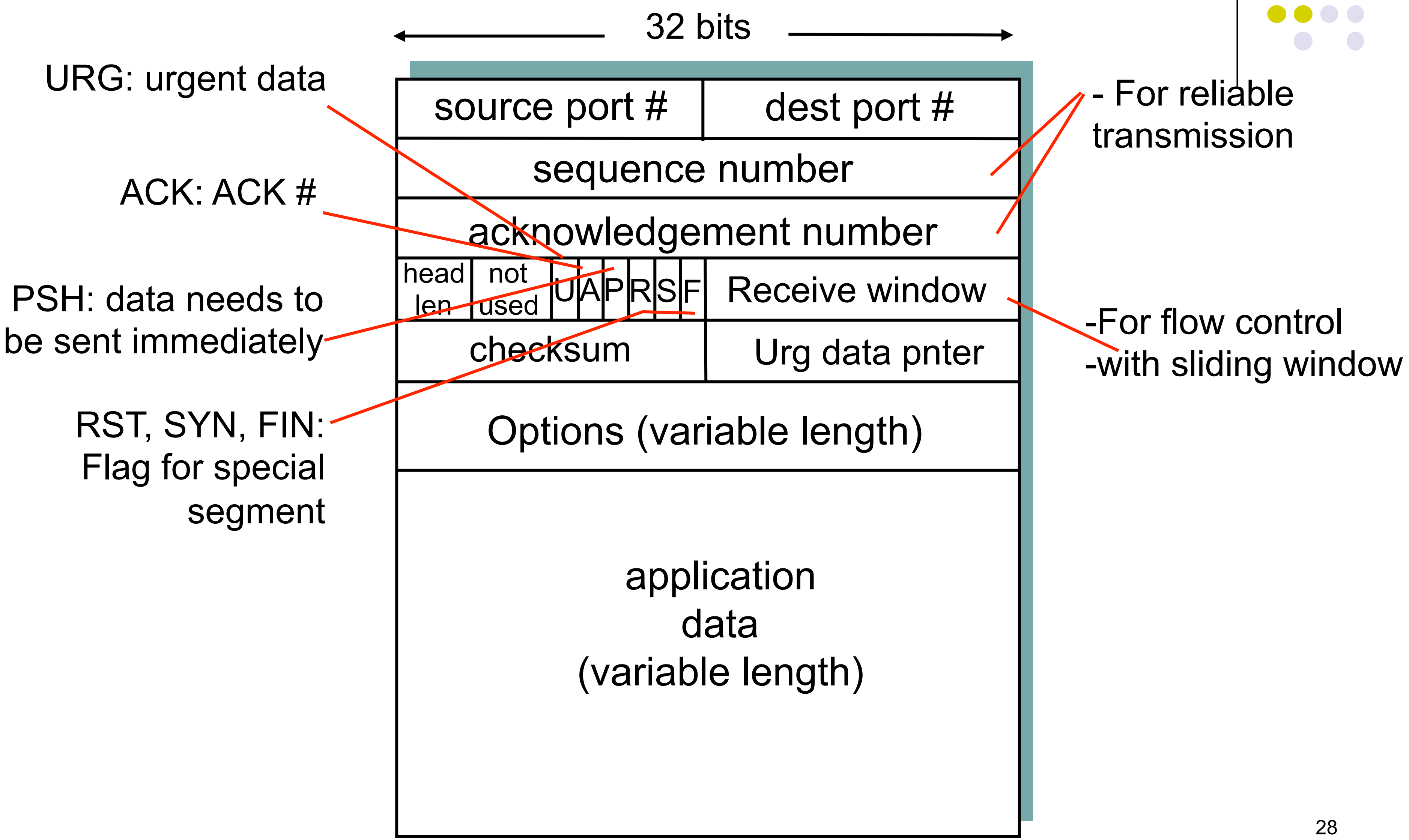
Flow control

Congestion control

Overview of TCP

- Connection oriented
  - 3 steps hand-shake
- Data transmission in stream of byte, reliable
  - Use buffer
- Transmit data in pipeline
  - Increase the performance
- Flow control
  - Sliding windows
- Congestion control
  - Detect congestion and solve

TCP segment



32 bits

source port # | dest port #

sequence number

acknowledgement number

head len | not used | U | A | P | R | S | F | Receive window

checksum | Urg data pnter

Options (variable length)

application data (variable length)

URG: urgent data

ACK: ACK #

PSH: data needs to be sent immediately

RST, SYN, FIN: Flag for special segment

- For reliable transmission

-For flow control
-with sliding window

How TCP provide reliable service?

- In order to assure if data arrives to destination:
  - Seq. #
  - Ack
- TCP cycle life:
  - Connection establishing
    - 3 steps
  - Data transmission
  - Close connection

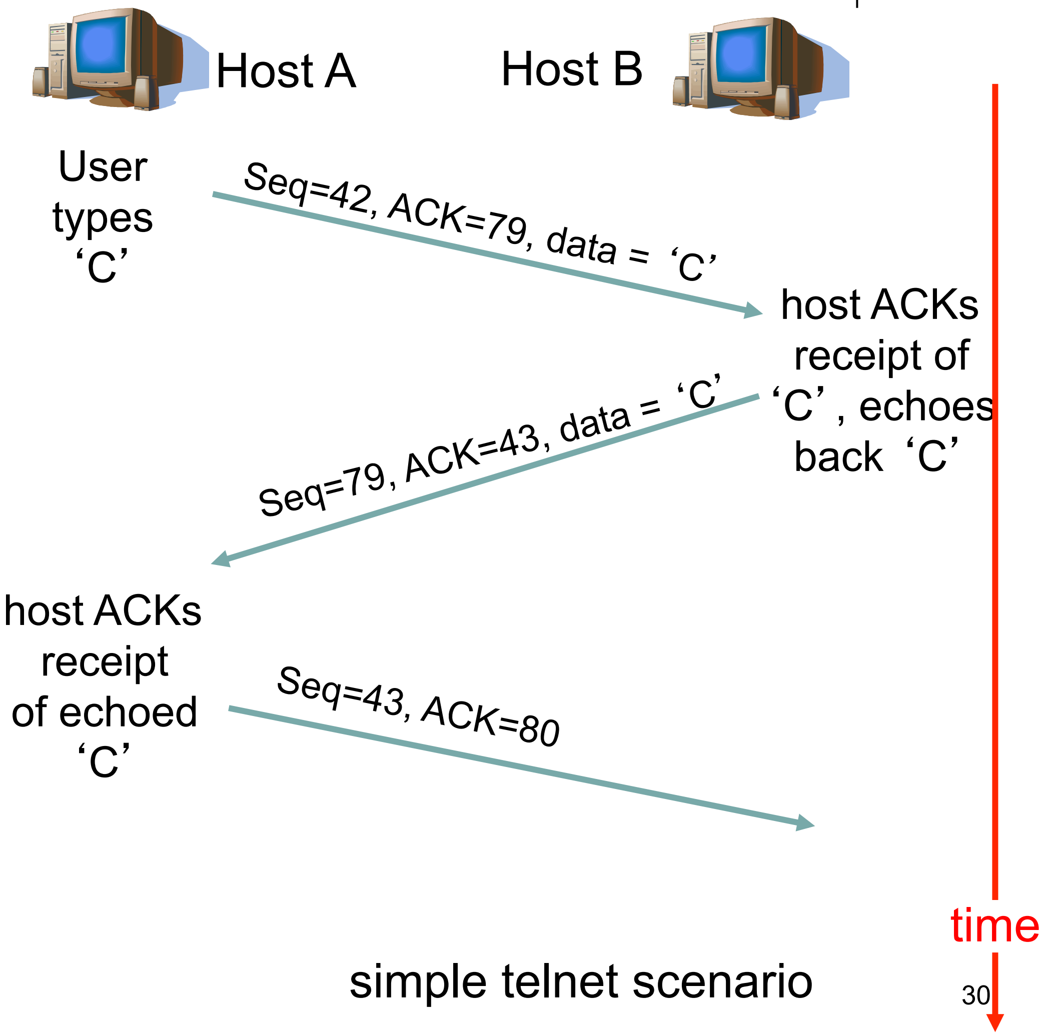
Acknowledgement in TCP

Seq. #:

- Index of the first byte of the segment in the data stream

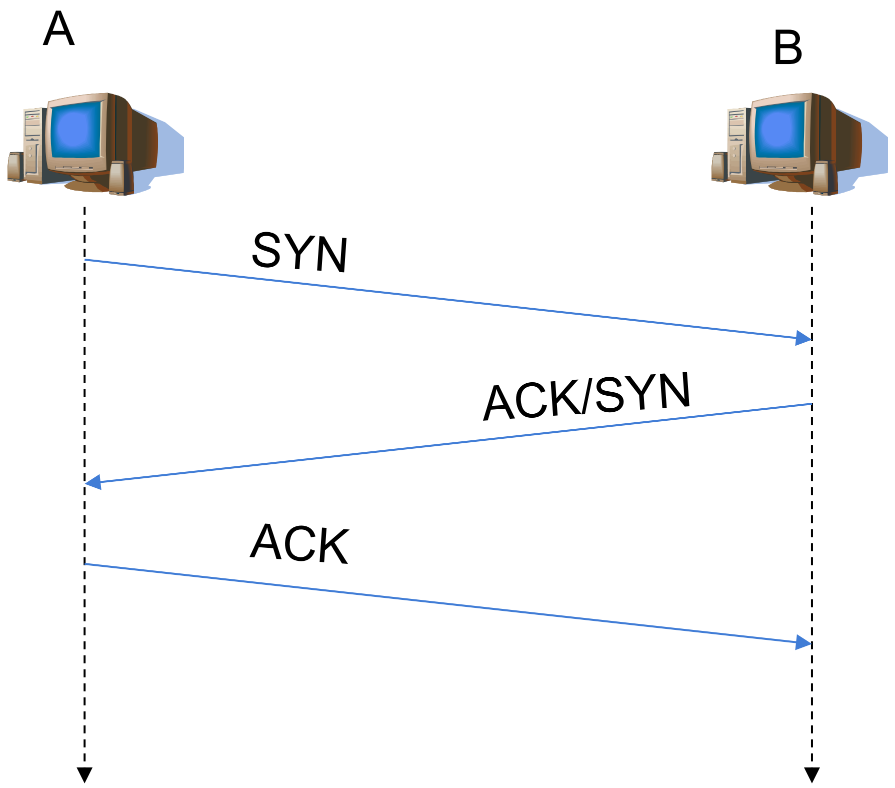
ACK:

- The index of the first byte expected to receive from the other-side
- Implicitly to confirm that the ACK senders have received well previous bytes


Host A
Host B
User types 'C'
Seq=42, ACK=79, data = 'C'
host ACKs receipt of 'C', echoes back 'C'
Seq=79, ACK=43, data = 'C'
host ACKs receipt of echoed 'C'
Seq=43, ACK=80
time

simple telnet scenario

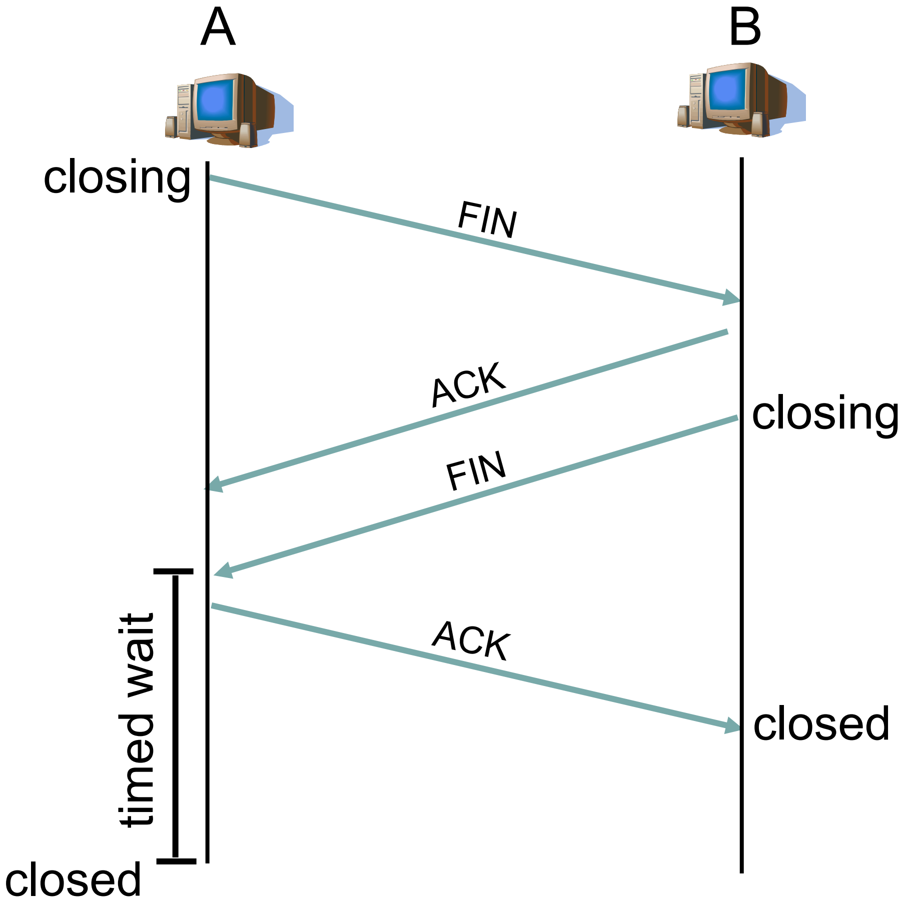
Connection establishing in TCP : 3 steps



- Bước 1: A sends SYN to B
  - Indicate initial value of seq # of A
  - No data
- Bước 2: B receives SYN, replies by SYNACK
  - B initiates the buffer on its side
  - Indicate initial value of seq. # of B
- Bước 3: A receives SYNACK, replies ACK, maybe with data.

Close connection

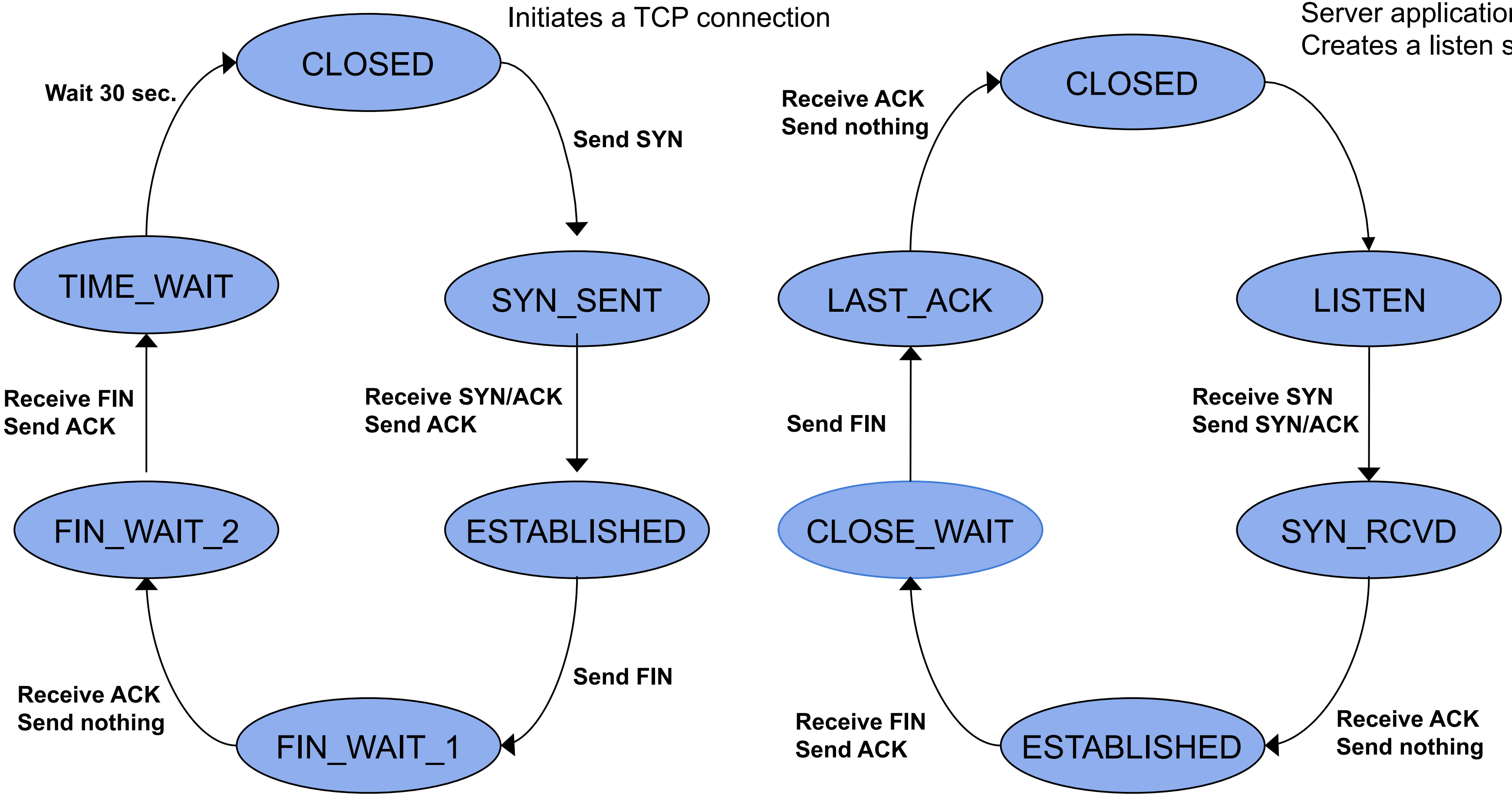
- Step 1: Send FIN to B
- Step 2: B receives FIN, replies ACK, closes the connection and sends FIN.
- Step 3: A receives FIN, replies ACK, go to “waiting”.
- Bước 4: B receives ACK. close connection



Symplified life cycle of TCP

Client application
Initiates a TCP connection

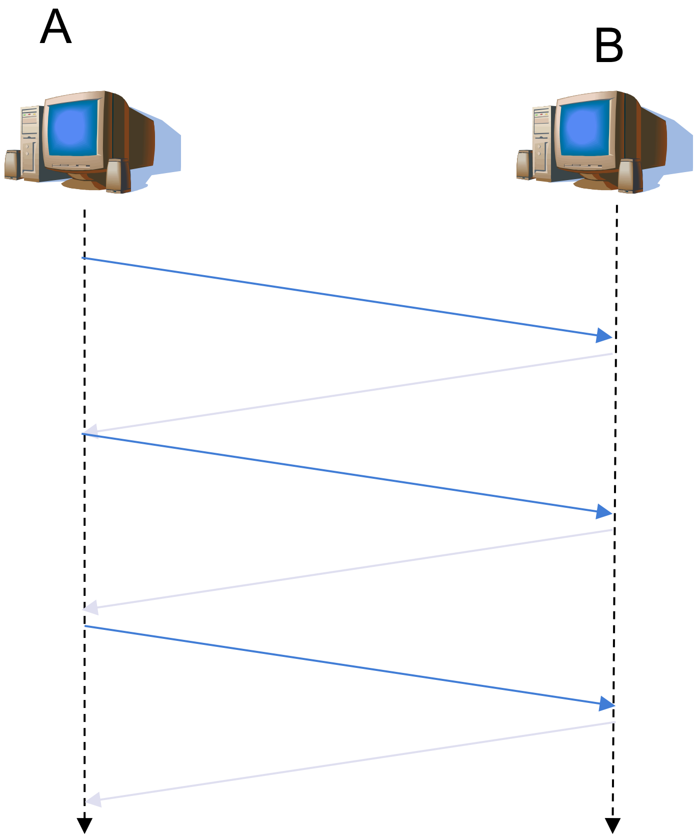
Server application
Creates a listen socket



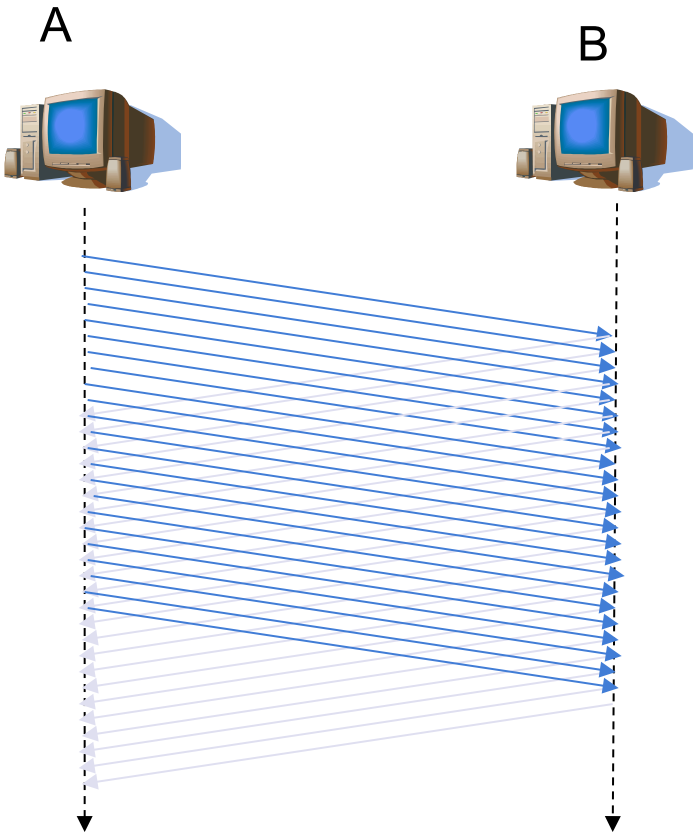
Client application
Initiates close connection

Flow control

Flow control(1)



Slow

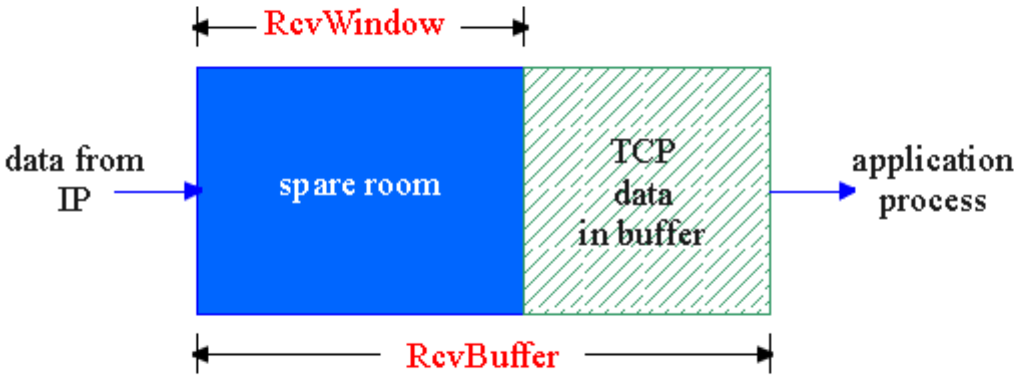


Overload

Flow control (2)

- Control the amount of data to be sent
  - Assure the best efficiency
  - Avoid overloading the receiver.
- Two windows
  - Rwnd: Receive window on receiver side
  - CWnd: Congestion window on sender side
- The maximum amount of data to be sent should be min(Rwnd, Cwnd)

Flow control TCP

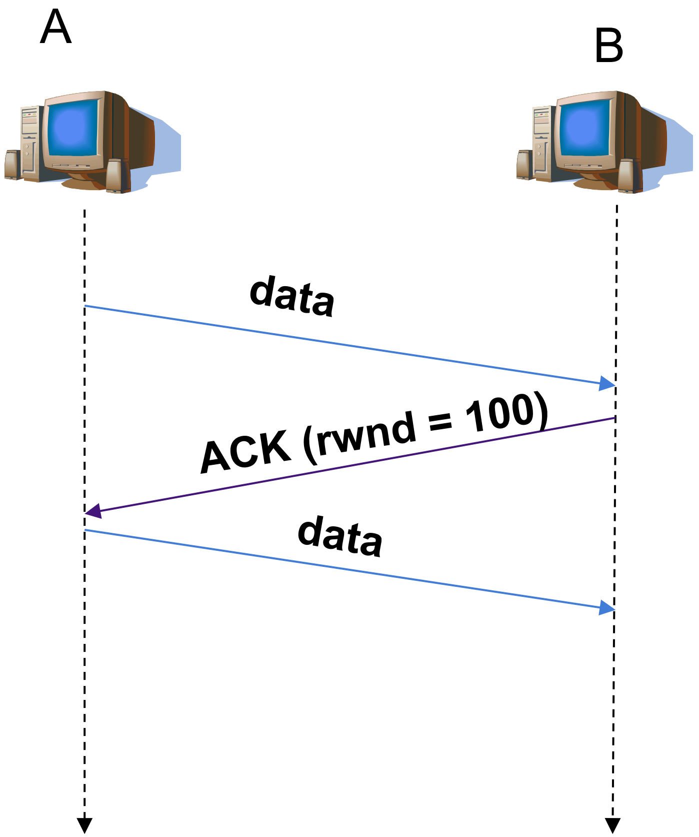


- Size of free buffer

= Rwnd

= `RcvBuffer-[LastByteRcvd - LastByteRead]`

Information exchanged on Rwnd

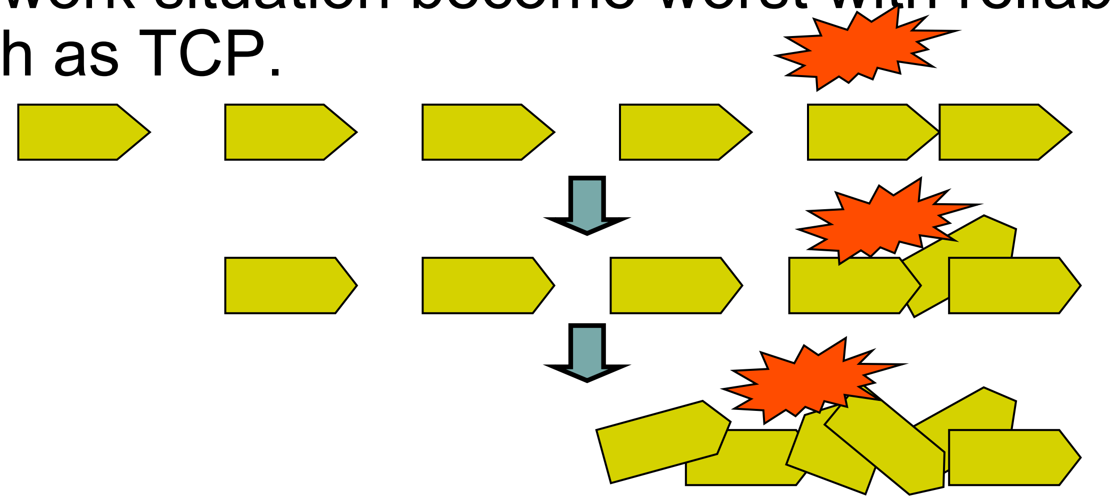


- Receiver inform regularly to senders the value of `Rwnd` in acknowledgment segments

Congestion control in TCP

Overview of Congestion control

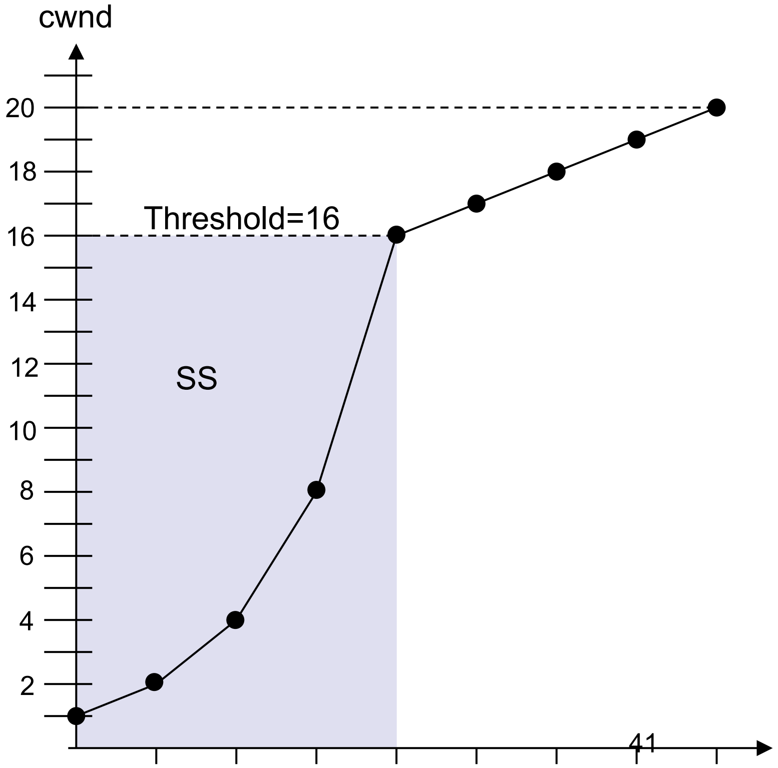
- When congestion happens?
  - Too many pairs of senders-receivers in the network
  - High traffic
- Consequence of congestion
  - Packet loss
  - Reduce of throughput, increase of delay
  - Network situation become worst with reliable protocol such as TCP.



Congestion occur

Principles of congestion control

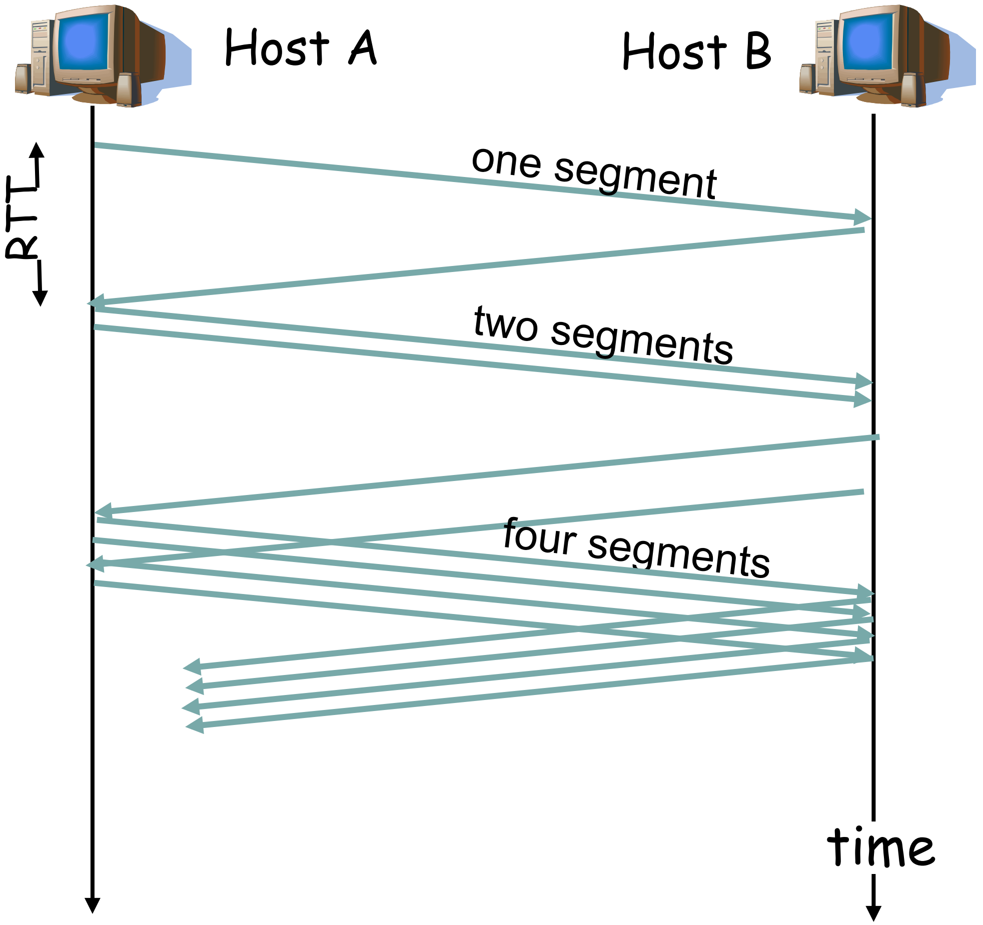
- Slow-start
  - Increases the transmission speed in exponential order
  - Increase until a threshold
- Congestion avoidance
  - Increase the transmission speed in linear order until congestion is detected
- How to detect the congestion?
  - By packets lost?



TCP Slow Start (1)

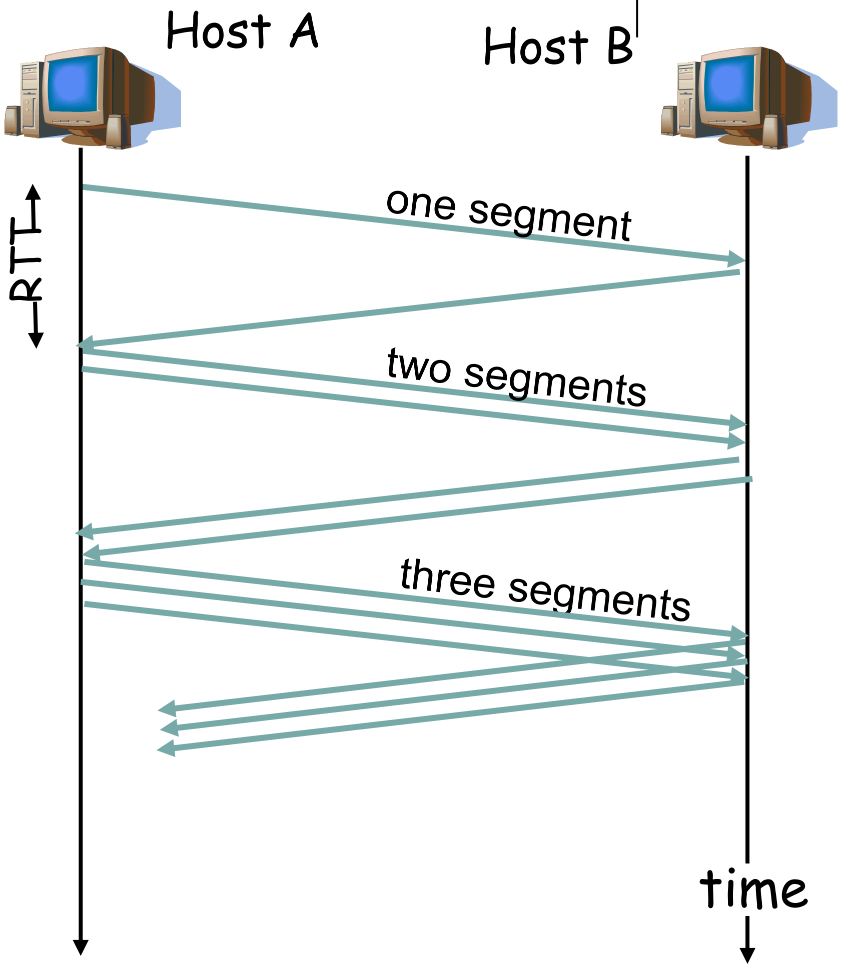
- Main idea
  - Initiate cwnd =1 MSS (Maximum segment size)
  - Increase cwnd =+1 MSS after each reception of a ACK packet from the receiver.
  - Increase slowly but the speed increase in exponential order
- Increase until a threshold: ssthresh
- After that TCP move to congestion avoidance period

TCP Slow Start (2)



Congestion avoidance

- Main idea
  - Increase cwnd in additional order until cwnd reaches to ssthresh
- After each RTT, cwnd =cwnd + 1 MSS



TCP reaction in congestion situation (1)

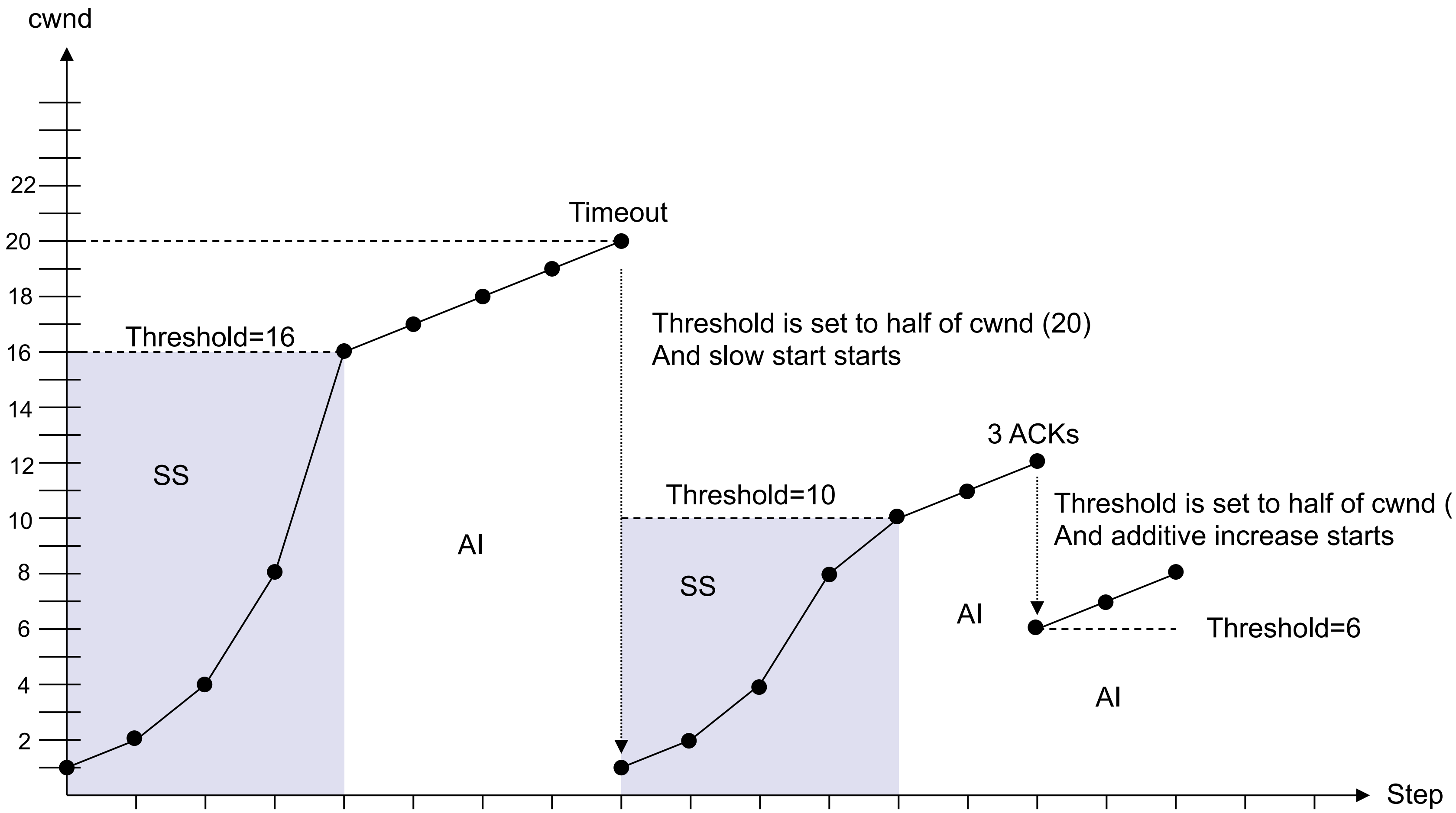
- Reduce the transmission speed
- How to detect the congestion?
  - If there are some re-transmits ➔ There might be congestion
- When the source node need to re-transmit data?
  - Timeout!
  - When it receives multiple ACK for the same segment

TCP reaction in congestion situation(2)

- When sender reach timeout but still does not receive ACK for a segment
  - TCP sets ssthresh = ½ current cwnd
  - TCP sets cwnd =1 MSS
  - TCP move to slow start phase

- If sender receives 3 identical ACK
  - TCP sets ssthresh = ½ current cwnd
  - TCP sets cwnd = ssthresh
  - TCP move to “congestion avoidance”

Congestion control – illustration



Exercise

- Assume that we need transmit 1 file
  - File size *O* =100KB over TCP connection
  - S is the size of each TCP segment, *S* = 536 byte
  - *RTT* = 100 *ms*.
- Assume that the congestion window size of TCP is fixed with value W.

  What is the minimum transmission time? If the transmission speed is

  - R = 10 Mbit/s;
  - R= 100 Mbits/s.

Solution (cont.)

- T transmit (W packet) = W * S/R
- Transmit without waiting:
- => (W-1)*S/R >= RTT
- => W >= RTT*R/S +1
- Time to transmit all data L = L/R + RTT
- R=100 Mbps
  - W>= 100ms * 100 Mbps/ (536*8) + 1

Exercise

- Assume that we need transmit 1 file
  - File size *O* =100KB over TCP connection
  - S is the size of each TCP segment, *S* = 536 byte
  - *RTT* = 100 *ms*.
- Assume that the congestion window of TCP works according to slow-start mechanism.
- What is the size of the congestion window when the whole file is transmited.
- How much of time is required for transmitting the file? If R = 10 Mbit/s; R= 100 Mbits/s.