Transport Layer: TCP 1

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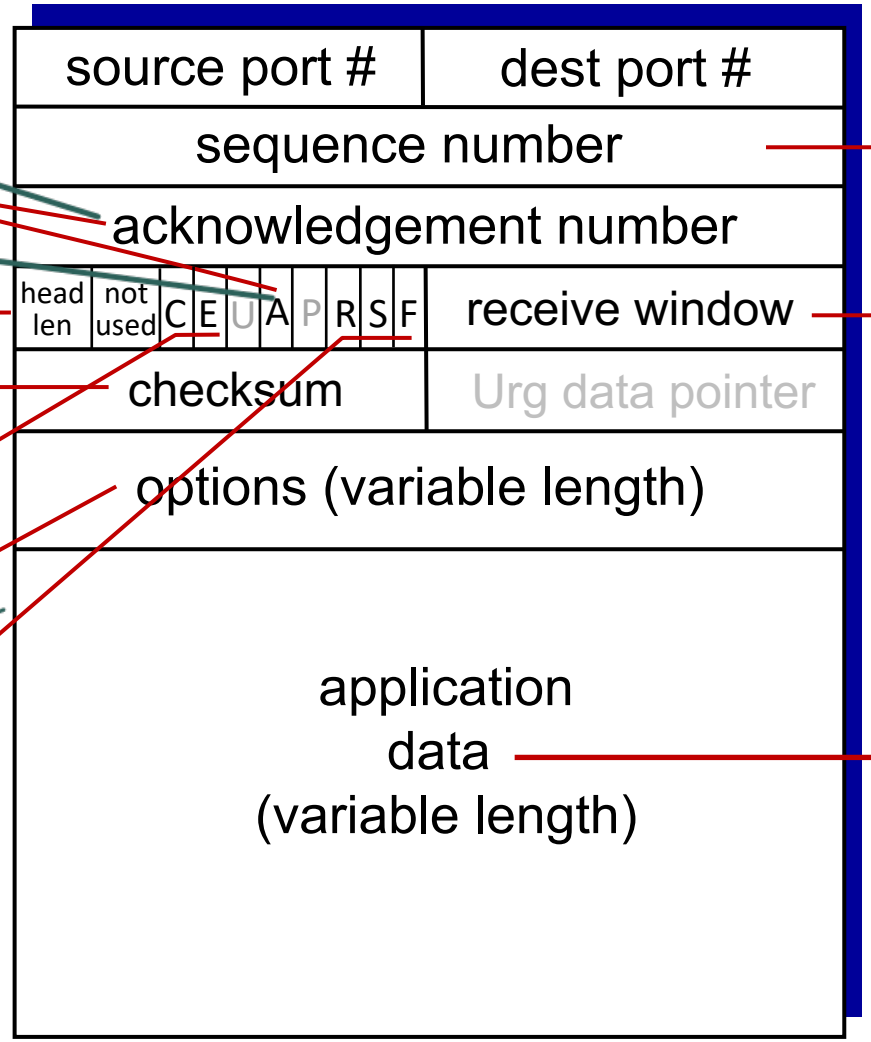
Chapter 3: roadmap

- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- Principles of reliable data transfer
- Connection-oriented transport: TCP
  - segment structure
  - reliable data transfer
  - flow control
  - connection management
- Principles of congestion control
- TCP congestion control

TCP: overview RFCs: 793,1122, 2018, 5681, 7323

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

TCP segment structure



ACK: seq # of next expected byte; A bit: this is an ACK

length (of TCP header)

Internet checksum

C, E: congestion notification

TCP options

RST, SYN, FIN: connection management

segment seq #: counting bytes of data into bytestream (not segments!)

flow control: # bytes receiver willing to accept

data sent by application into TCP socket

TCP sequence numbers, ACKs

Sequence numbers:

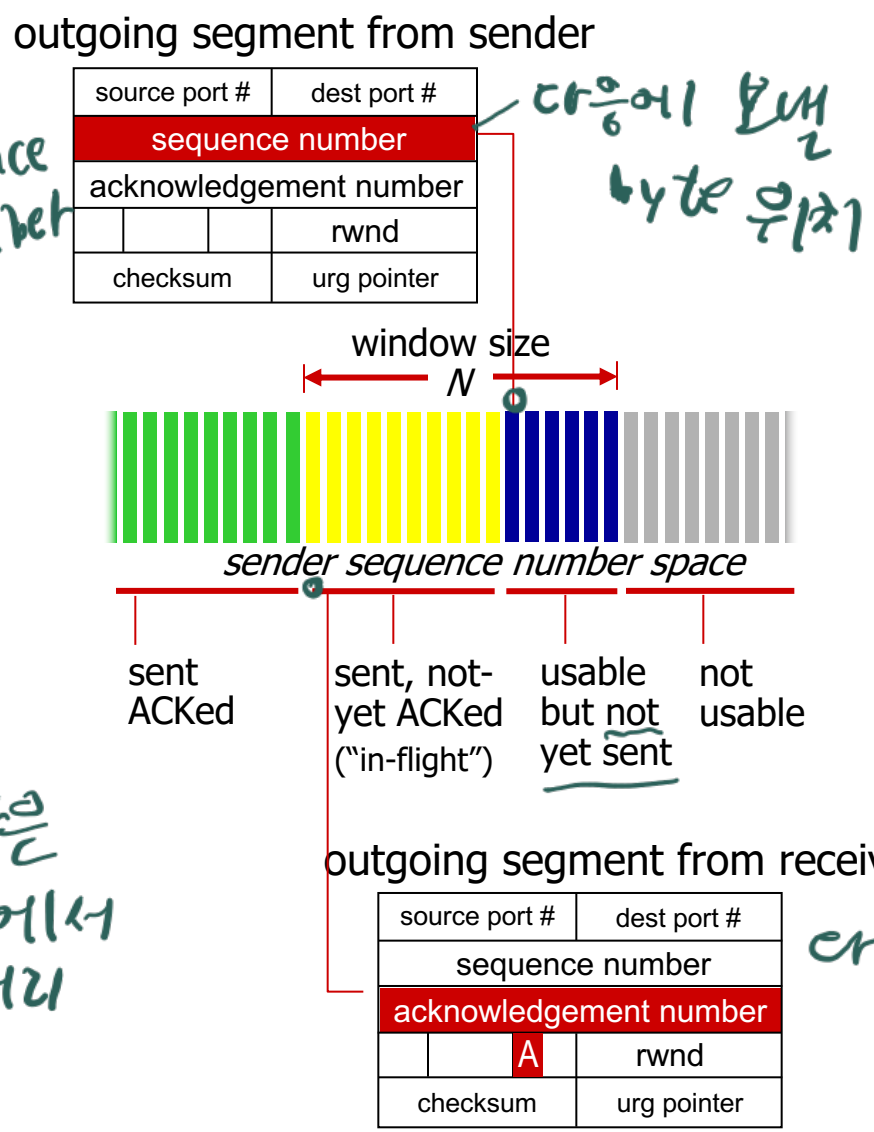
- byte stream "number" of first byte in segment's data

Acknowledgements:

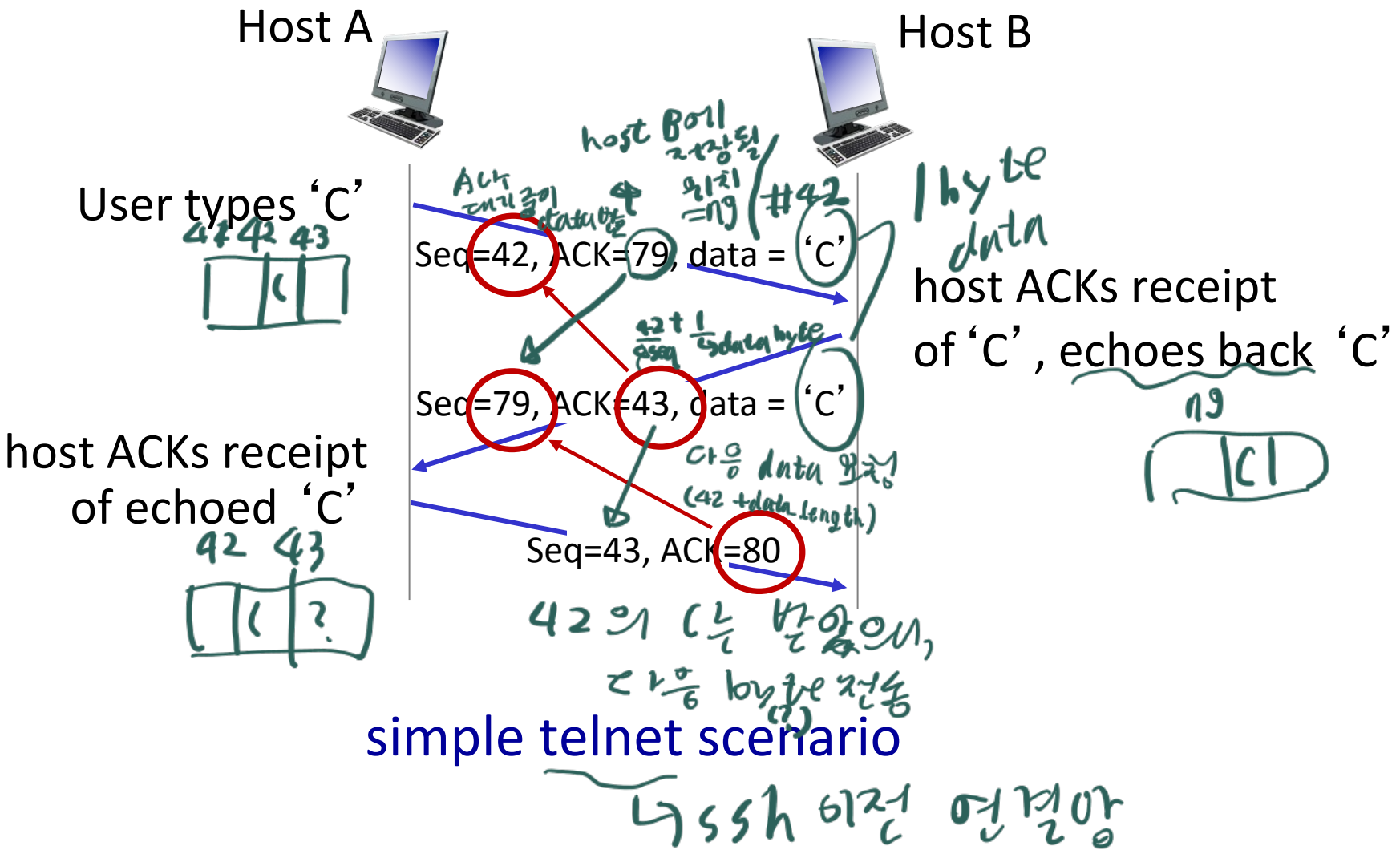
- seq # of next byte expected from other side
- cumulative ACK

Q: how receiver handles out-of-order segments

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



TCP sequence numbers, ACKs



TCP round trip time, timeout

적절한 timer 길이 - too short : multiple retransmit
                  - too long : time loss

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

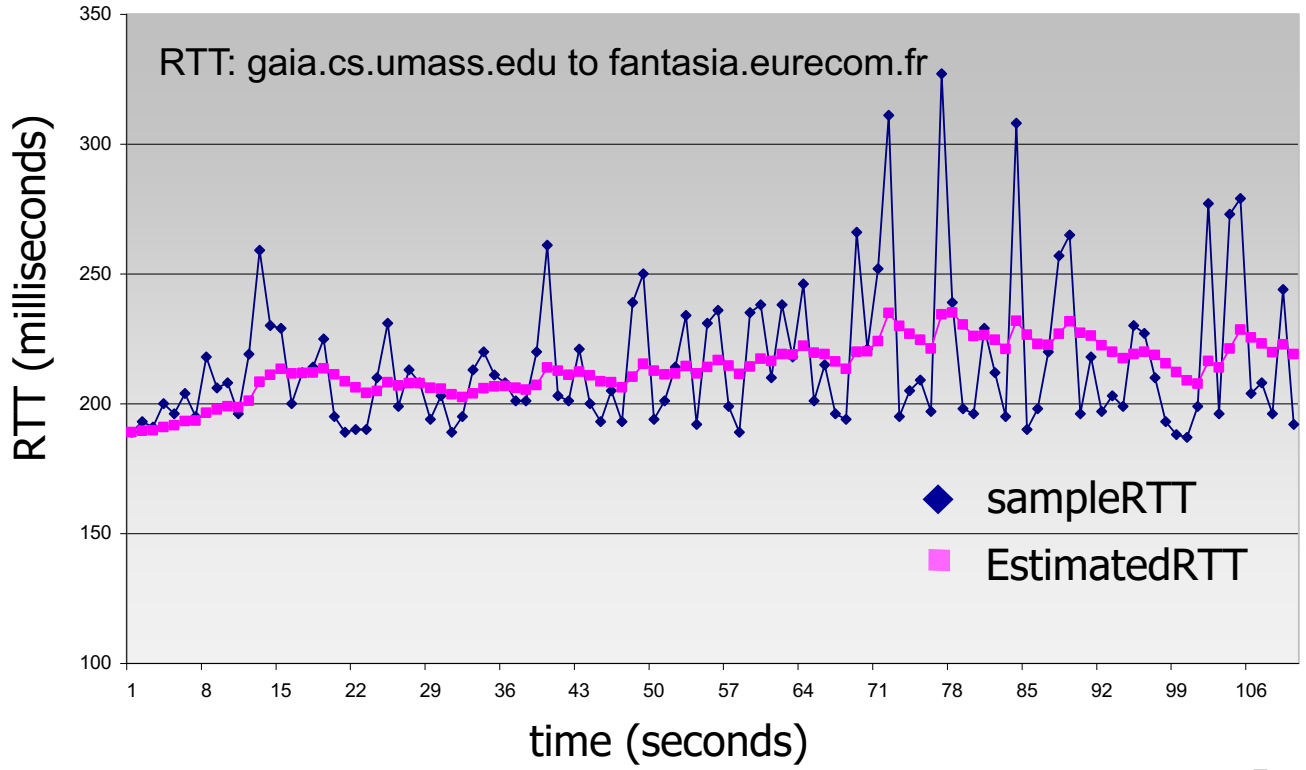
Q: how to estimate RTT?

- `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, timeout

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

- exponential weighted moving average (EWMA)
- 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**: want a larger safety margin

TimeoutInterval = EstimatedRTT + 4*DevRTT

estimated RTT

"safety margin"

편차

- **DevRTT**: EWMA of **SampleRTT** deviation from **EstimatedRTT**:

DevRTT = (1-β)*DevRTT + β*|SampleRTT-EstimatedRTT|

(typically, β = 0.25)

변동폭이 크면 편차↑ 이면 margin↑

TCP Sender (simplified)

event: data received from application

- 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`**

event: timeout

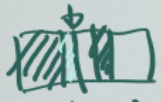
- retransmit segment that caused timeout
- restart timer

event: ACK received

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

TCP Receiver: ACK generation [RFC 5681]

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 starts at lower end of gap



전부 정상 수신

정상 & 이전 ACK 전송 대기중 (1번 event 발생)

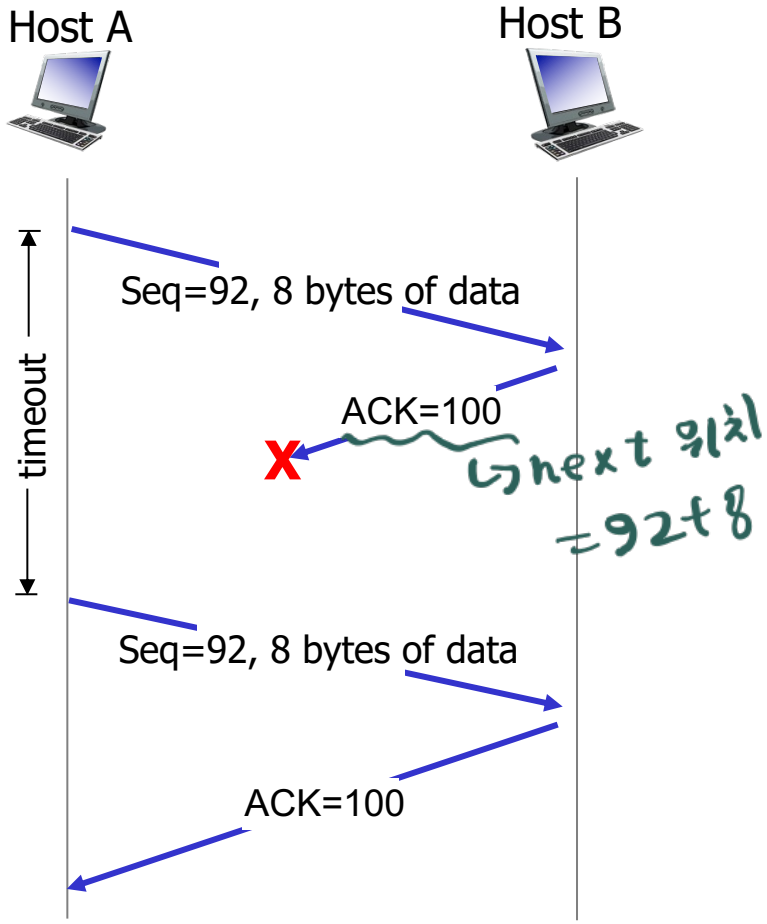
순서 X (seq보다 큰) ⇒ Gap발생

↳ 최소목표 채울라는 가정

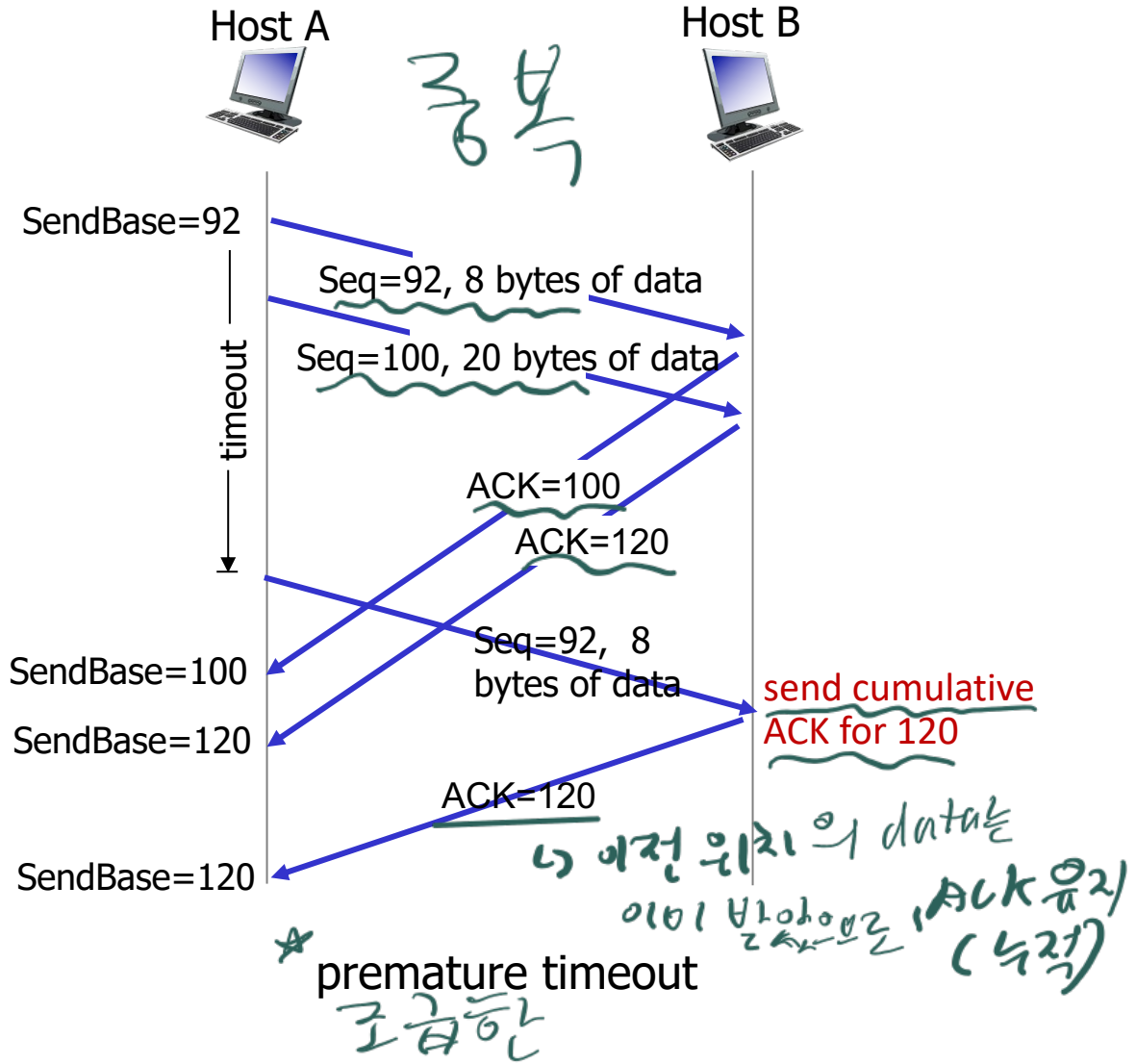
→ 다수의 packet을 묶어 하나의 packet으로 전송 (resource 절약)

↳ 빈 값에 대한 ACK 요청 전송 (이전 ACK 반복)

TCP: retransmission scenarios



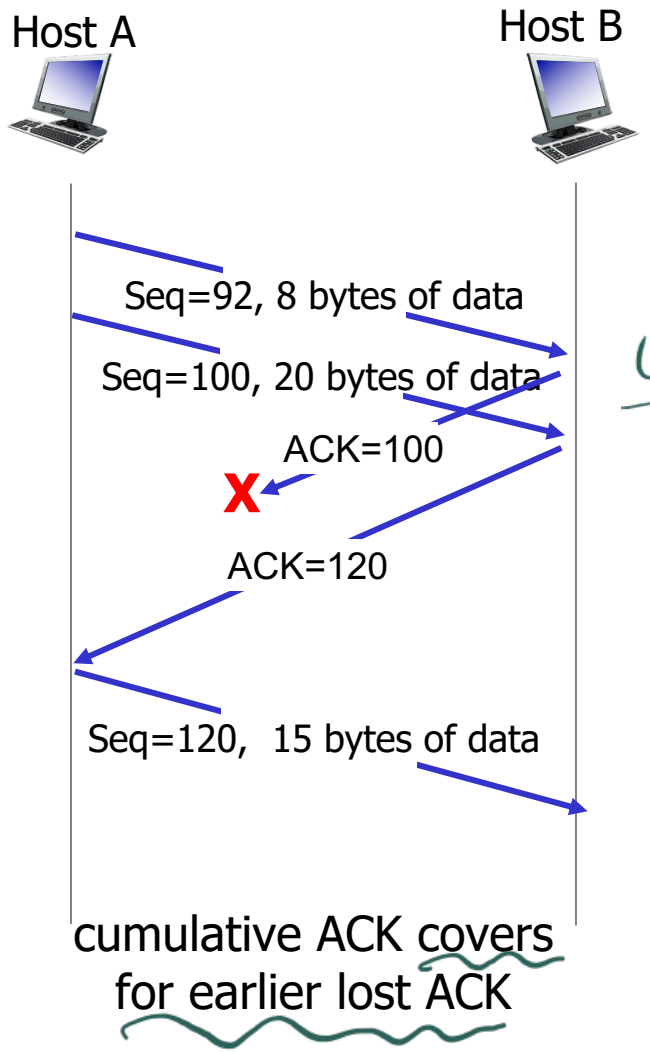
lost ACK scenario



premature timeout

조급한

TCP: retransmission scenarios



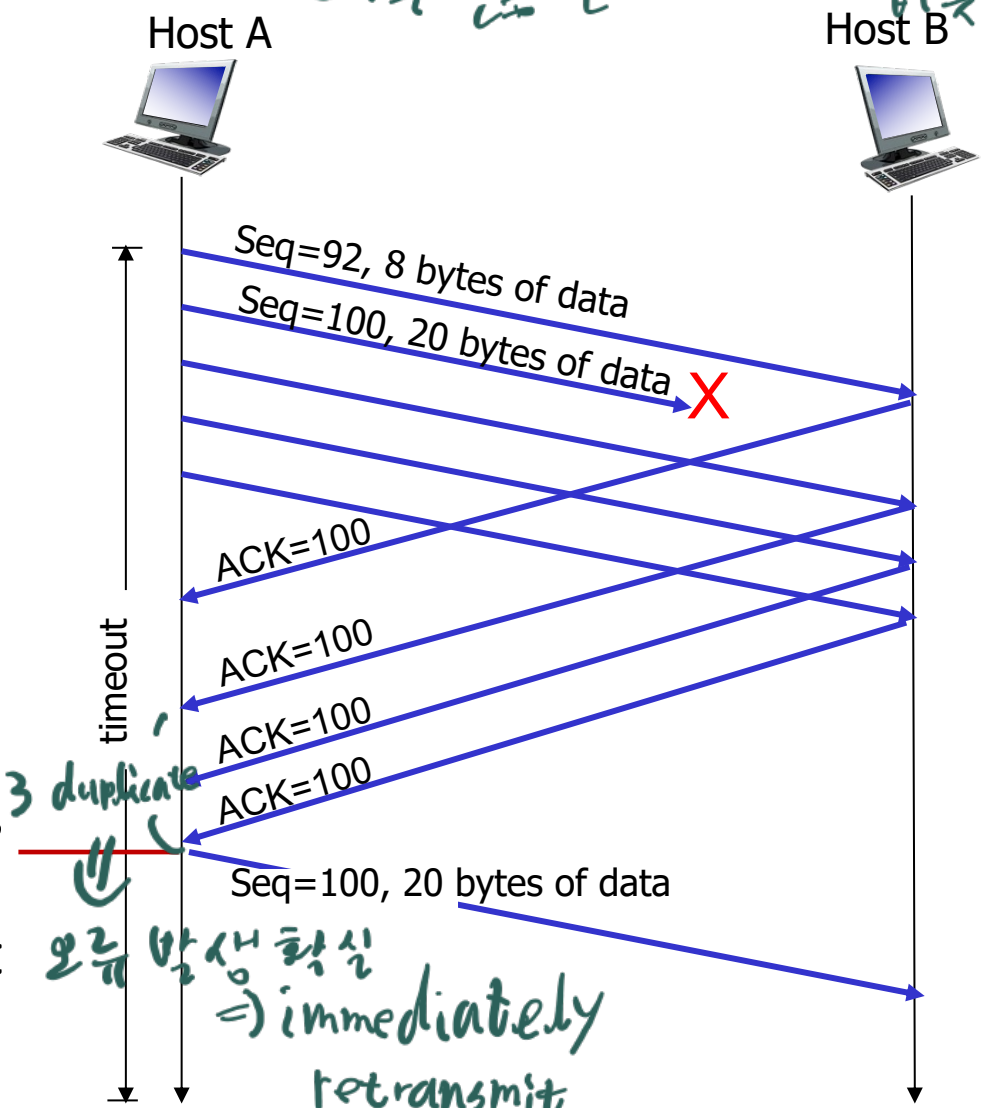
누적 ACK를
사용하므로,
이전 ACK에서
오류가 발생해도
cover 가능

TCP fast retransmit

TCP fast retransmit

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

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



Receipt of three duplicate ACKs indicates 3 segments received after a missing segment – lost segment is likely. So retransmit!

How to determine TCP window size?

경험적 평가?
RTT 비례?
packet loss 비례?

ACK 회신 텀이
└ 한 번에
보낼 수 있는
packet 수

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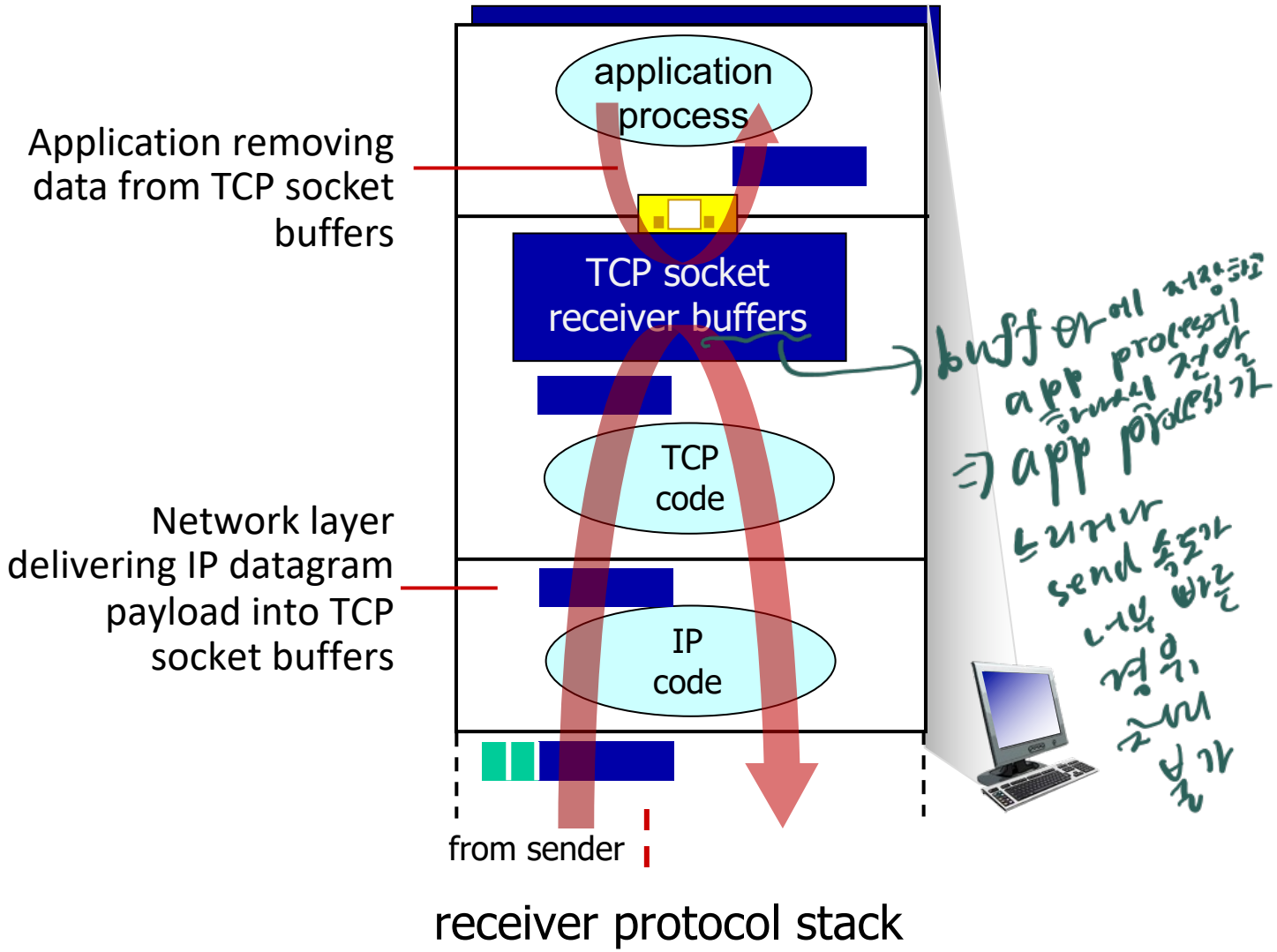
TCP flow control

Q: What happens if network layer delivers data faster than application layer removes data from socket buffers?

⇒ 공간이 부족 (overflow)



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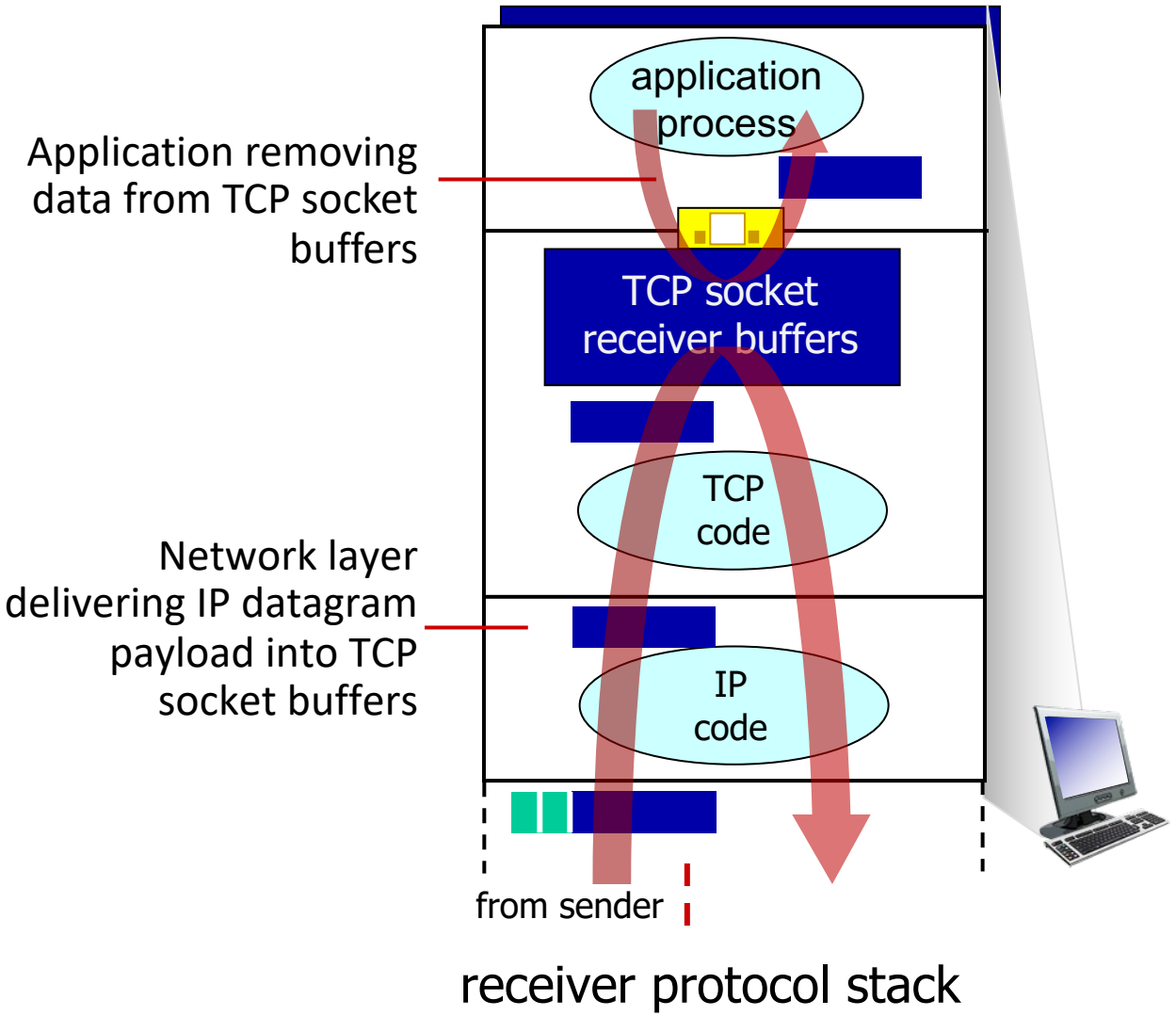

receiver protocol stack

TCP flow control

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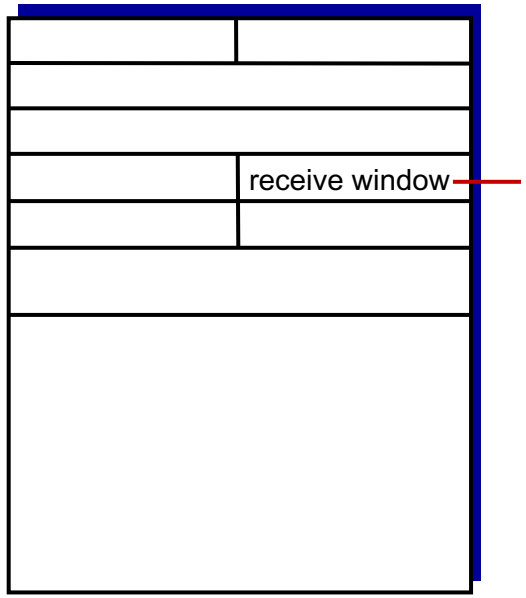
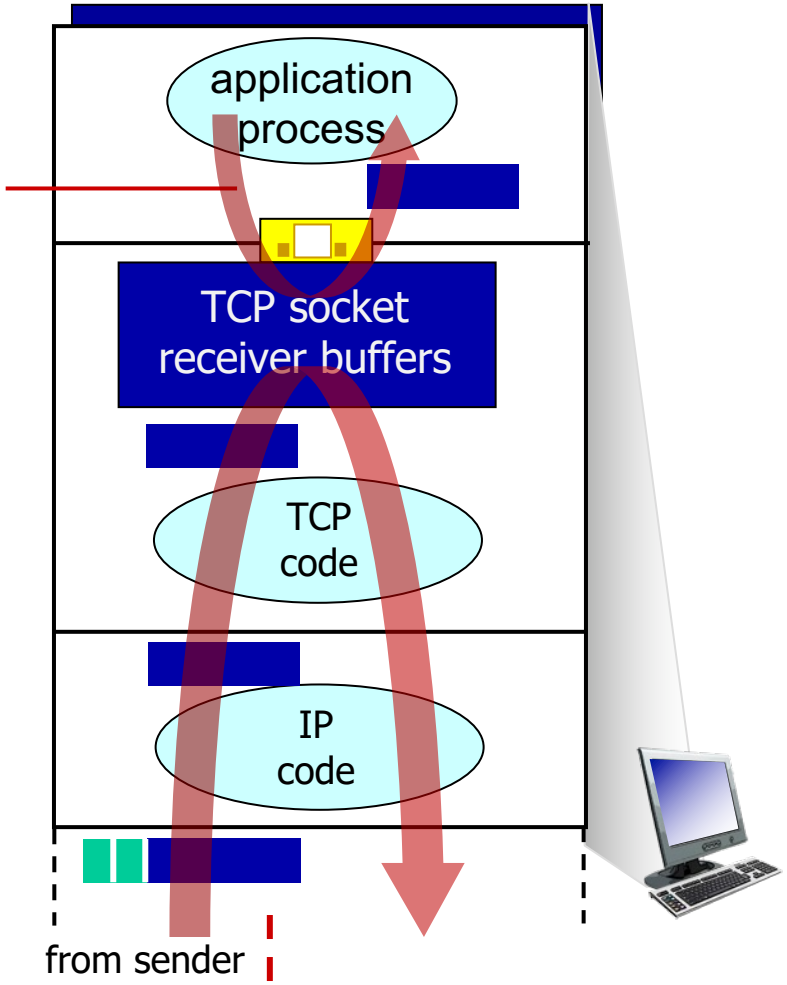


flow control: # bytes receiver willing to accept

Application removing data from TCP socket buffers



receiver protocol stack

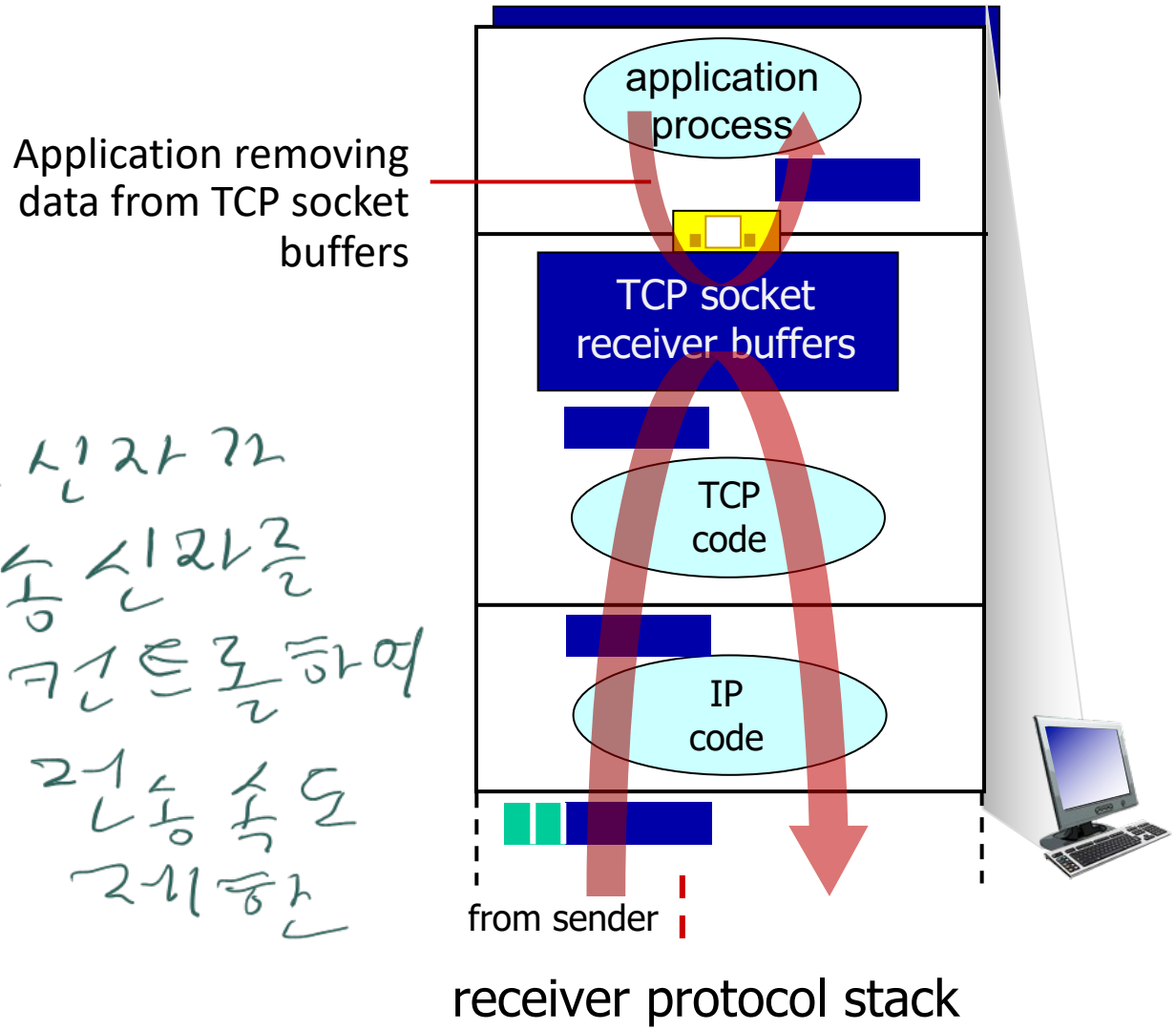
TCP flow control

Q: What happens if network layer delivers data faster than application layer removes data from socket buffers?

flow control

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

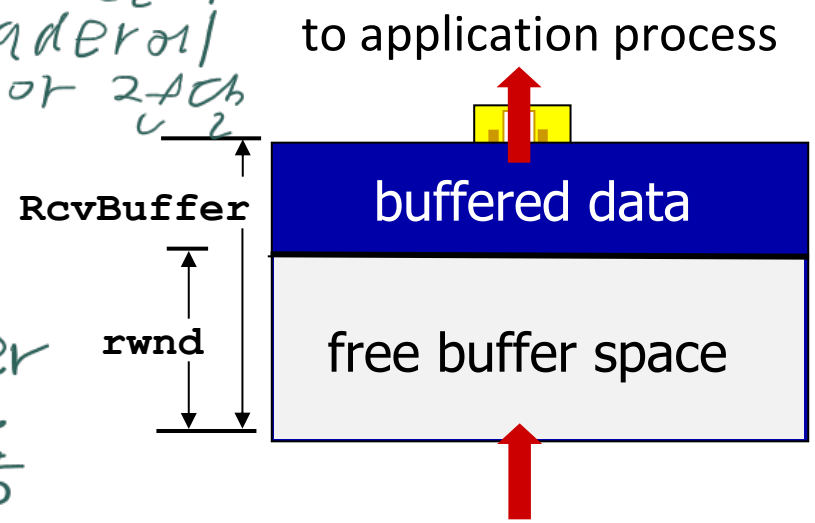
수신자가 송신자를 컨트롤하여 전송 속도 조절함



receiver protocol stack

TCP flow control

- TCP receiver "advertises" free buffer space in **`rwnd`** field in TCP header
  - **`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 received **`rwnd`**
- guarantees receive buffer will not overflow



TCP receiver-side buffering

TCP flow control

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  - **`RcvBuffer`** size set via socket options (typical default is 4096 bytes)
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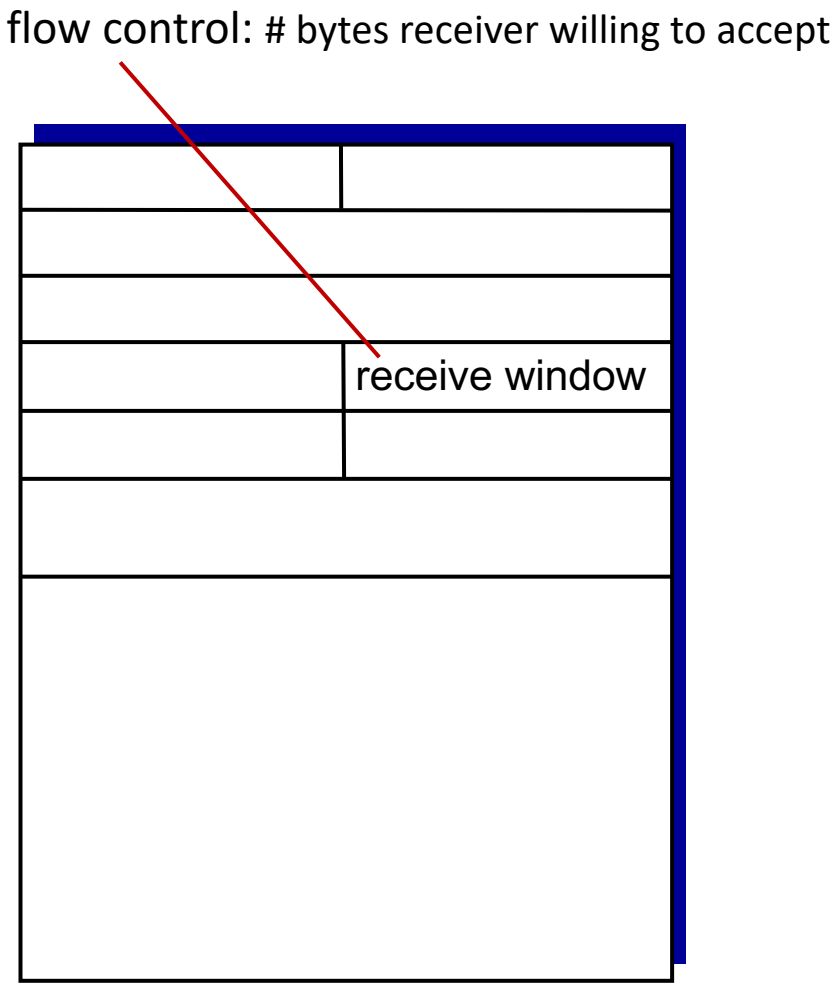
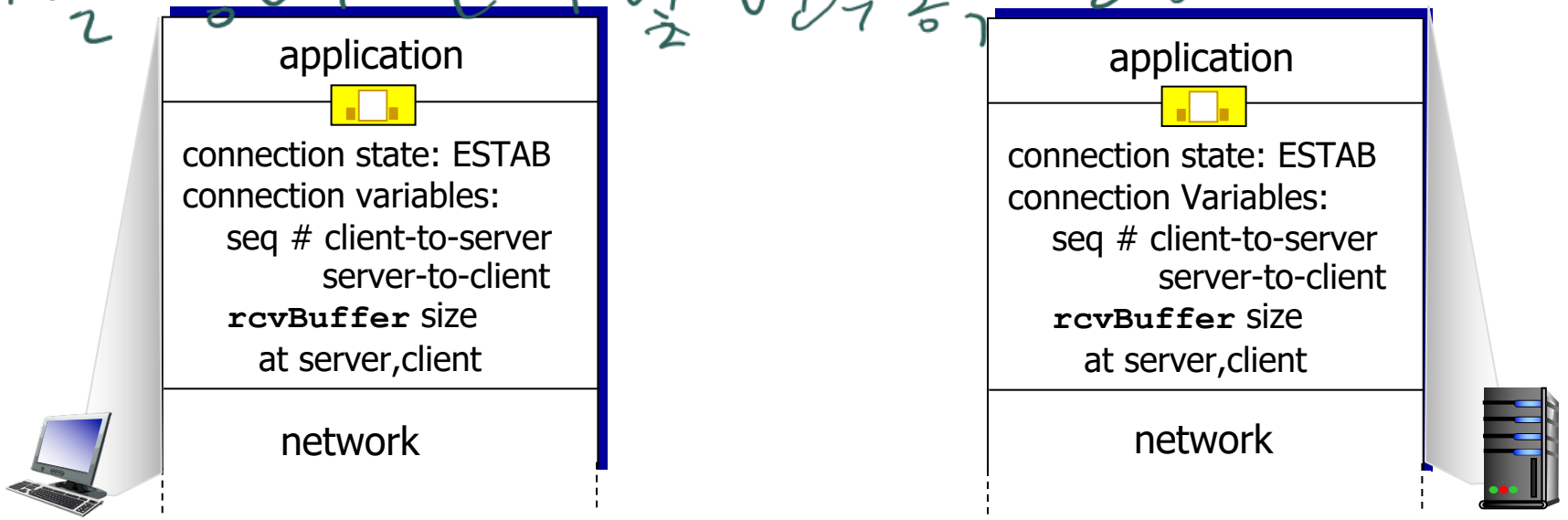


TCP segment format

TCP 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., starting seq #s)



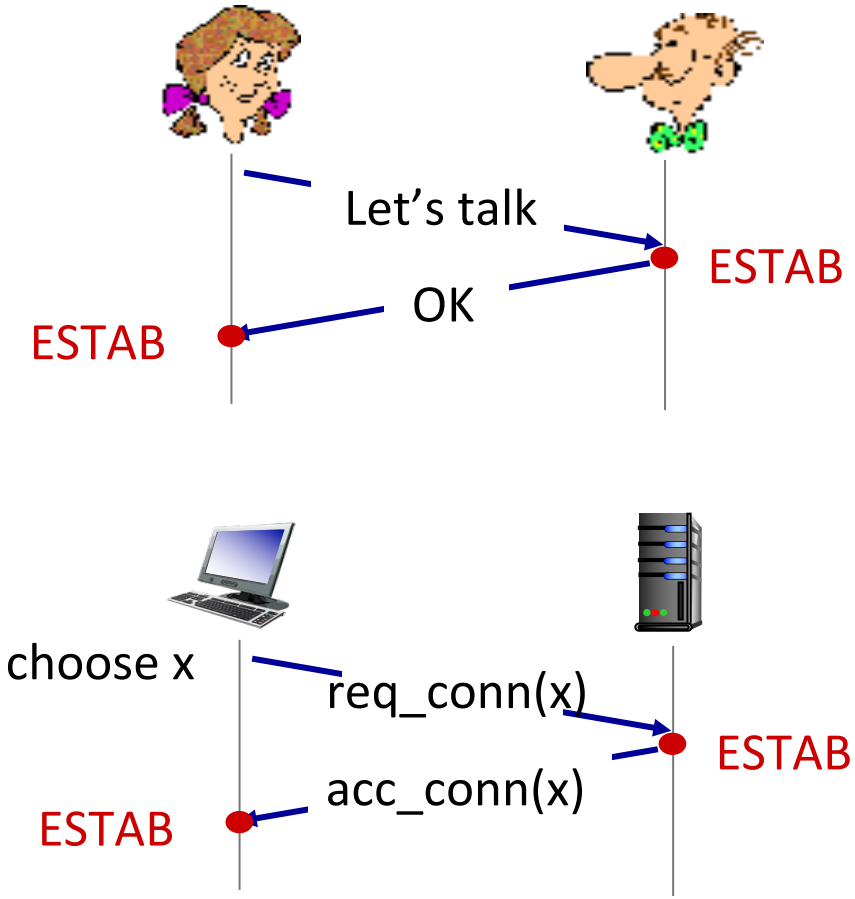

```
Socket clientSocket =
  newSocket("hostname","port number");
```

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

Agreeing to establish a connection

On Lec 10

2-way handshake:



Q: will 2-way handshake always work in network?

- variable delays
- retransmitted messages (e.g. req_conn(x)) due to message loss
- message reordering
- can't "see" other side

TCP 3-way handshake

Lec 10

Client state

```
clientSocket = socket(AF_INET, SOCK_STREAM)
```

LISTEN

```
clientSocket.connect((serverName,serverPort))
```

choose init seq num, x
send TCP SYN msg

SYNSENT

SYNbit=1, Seq=x

choose init seq num, y
send TCP SYNACK
msg, acking SYN

SYNbit=1, Seq=y
ACKbit=1; ACKnum=x+1

received SYNACK(x)
indicates server is live;
send ACK for SYNACK;
this segment may contain
client-to-server data

ESTAB

ACKbit=1, ACKnum=y+1

received ACK(y)
indicates client is live

Server state

```
serverSocket = socket(AF_INET,SOCK_STREAM)
serverSocket.bind(('',serverPort))
serverSocket.listen(1)
connectionSocket, addr = serverSocket.accept()
```

LISTEN

SYN RCVD

ESTAB

A human 3-way handshake protocol

Closing a TCP connection

- client, server each close their side of connection
  - send TCP segment with FIN bit = 1
- respond to received FIN with ACK
  - on receiving FIN, ACK can be combined with own FIN
- simultaneous FIN exchanges can be handled

Chapter 3: roadmap

- Transport-layer services
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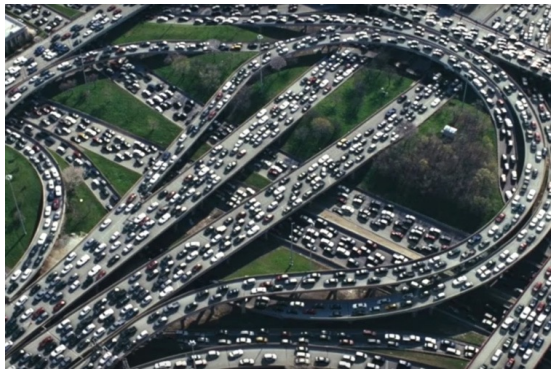
- Connectionless transport: UDP
- Principles of reliable data transfer
- Connection-oriented transport: TCP
- Principles of congestion control
- TCP congestion control
- Evolution of transport-layer functionality

Principles of congestion control

Congestion:

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



congestion control:
too many senders,
sending too fast

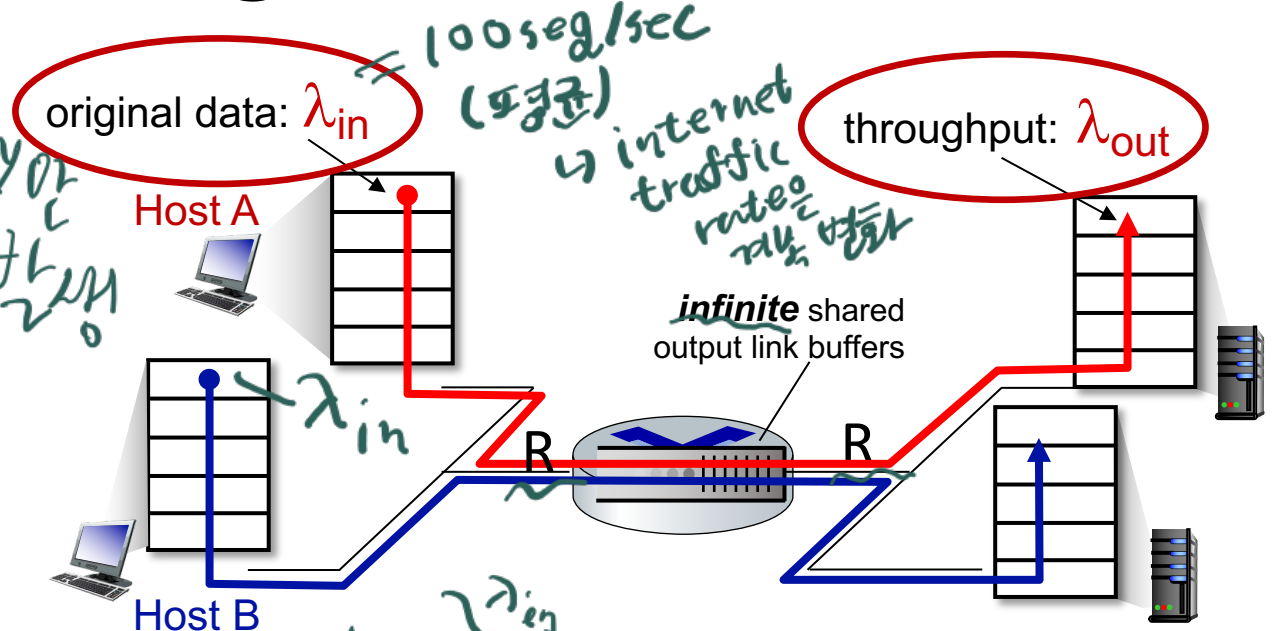
flow control: one sender
too fast for one receiver

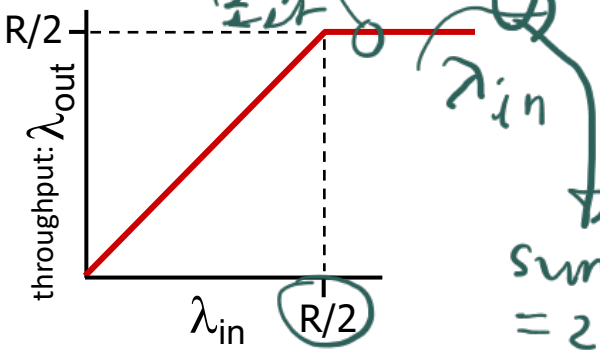
Causes/costs of congestion: scenario 1

Simplest scenario:

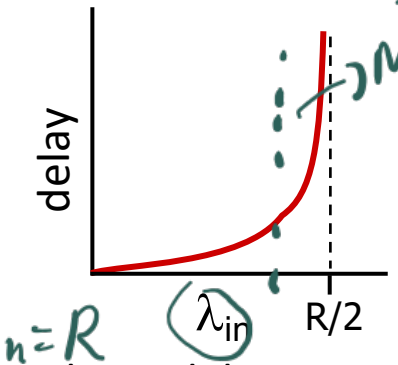
- one router, infinite buffers
- input, output link capacity: R
- two flows
- no retransmissions needed

Q: What happens as arrival rate λ_{in} approaches R/2?





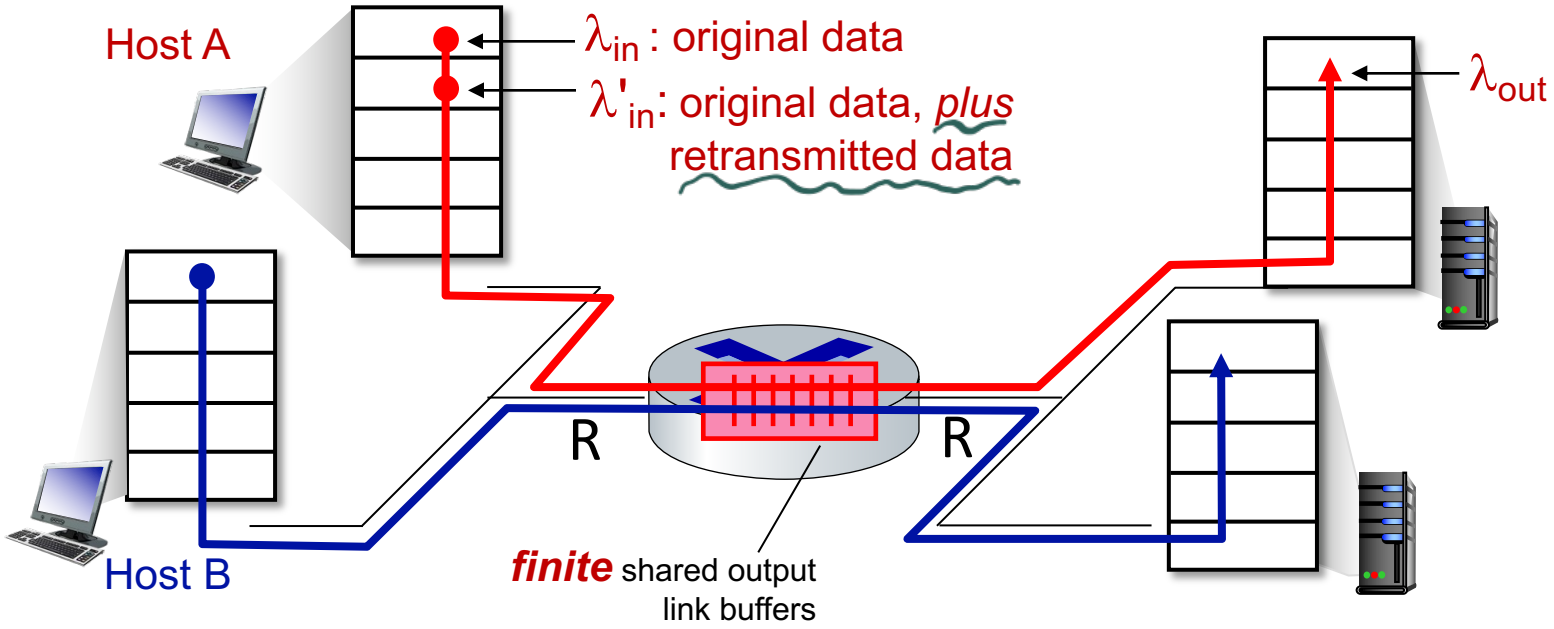
maximum per-connection throughput: R/2



large delays as arrival rate λ_{in} approaches capacity

Causes/costs of congestion: scenario 2

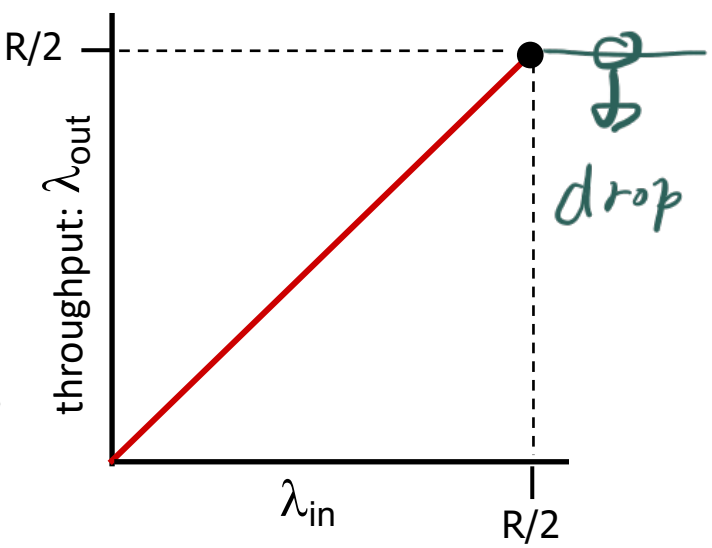
- one router, *finite* buffers
- sender retransmits lost, timed-out packet
  - application-layer input = application-layer output: $\lambda_{in} = \lambda_{out}$
  - transport-layer input includes *retransmissions* : $\lambda'_{in} \geq \lambda_{in}$

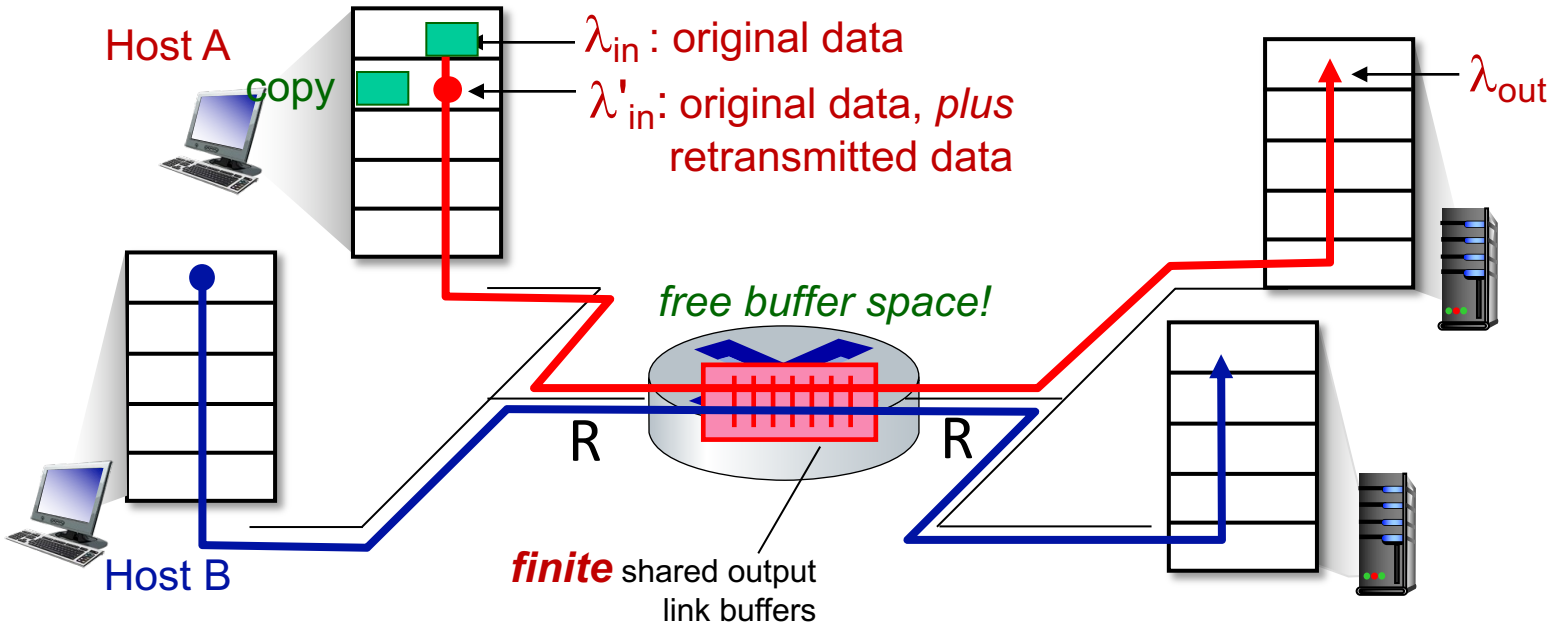


Causes/costs of congestion: scenario 2

Idealization: perfect knowledge → loss 발생 시점을 인지 가능 ⇒ drop 된 것을 알 수 있음

- sender sends only when router buffers available

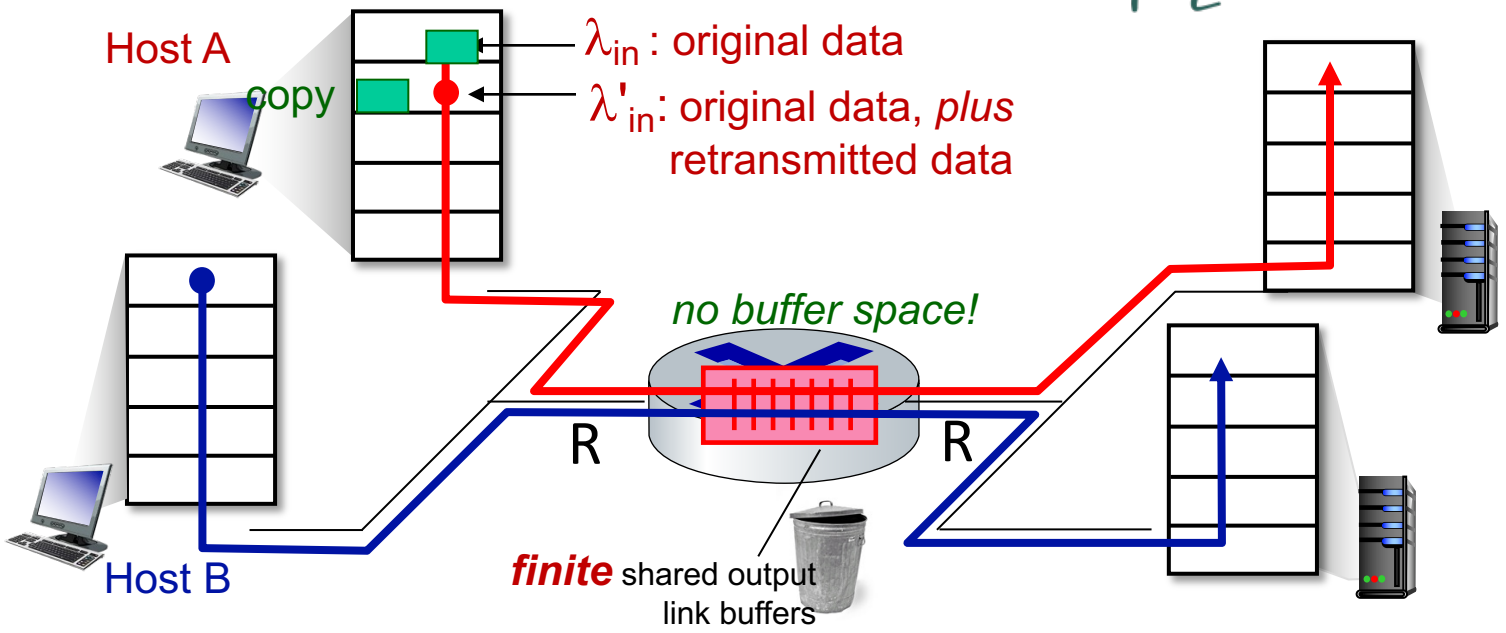




Causes/costs of congestion: scenario 2

Idealization: *some* perfect knowledge

- packets can be lost (dropped at router) due to full buffers
- sender knows when packet has been dropped: only resends if packet *known* to be lost



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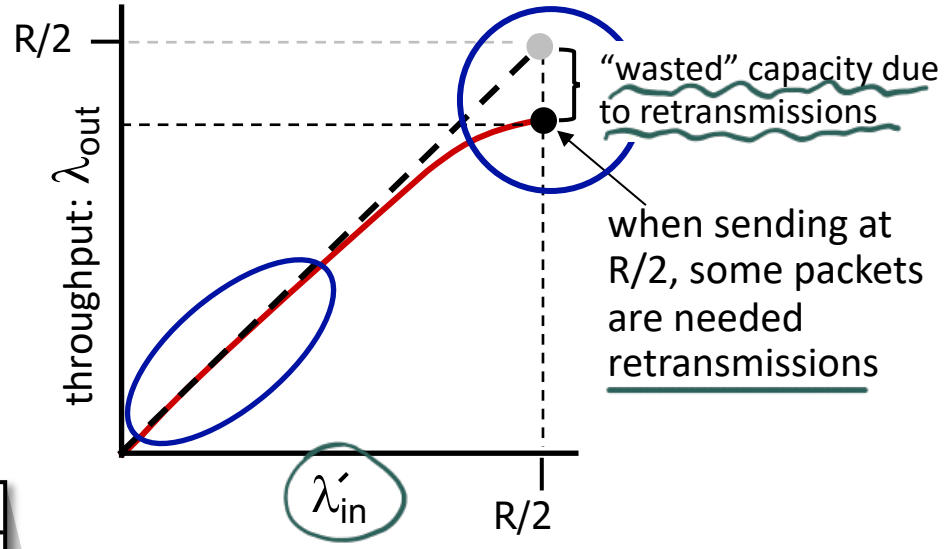
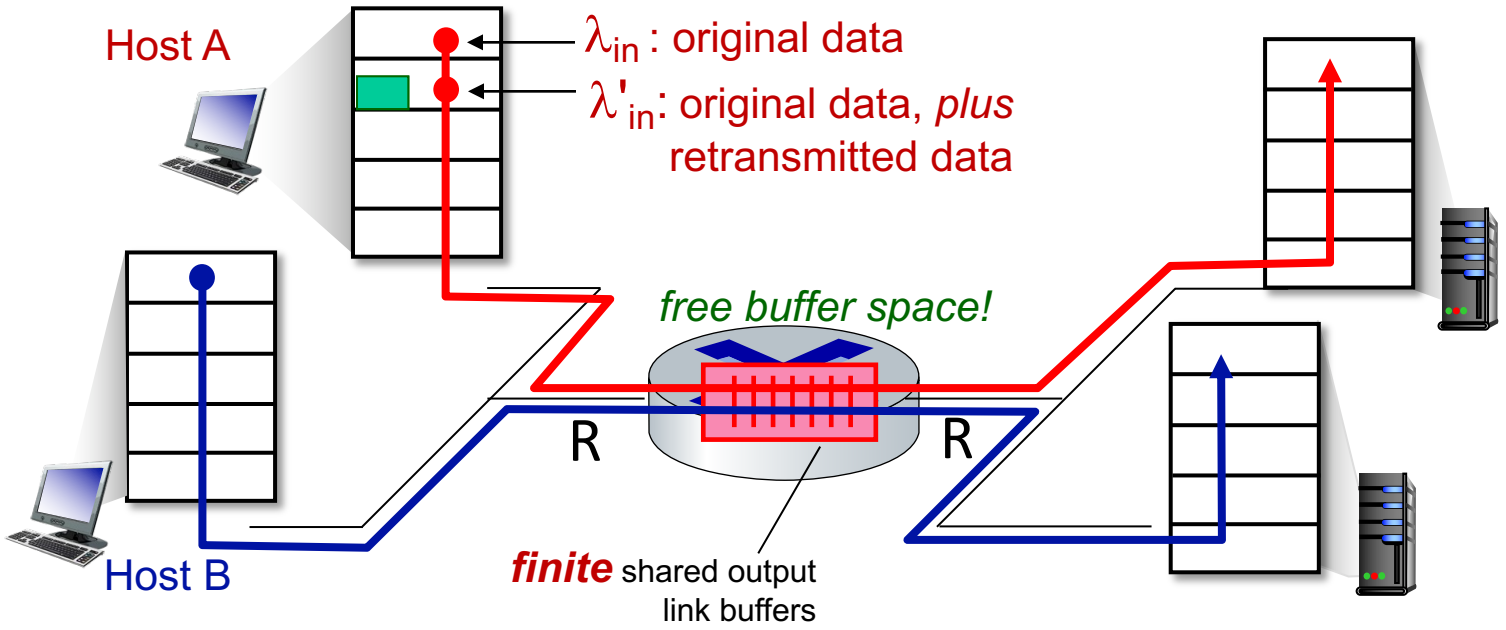
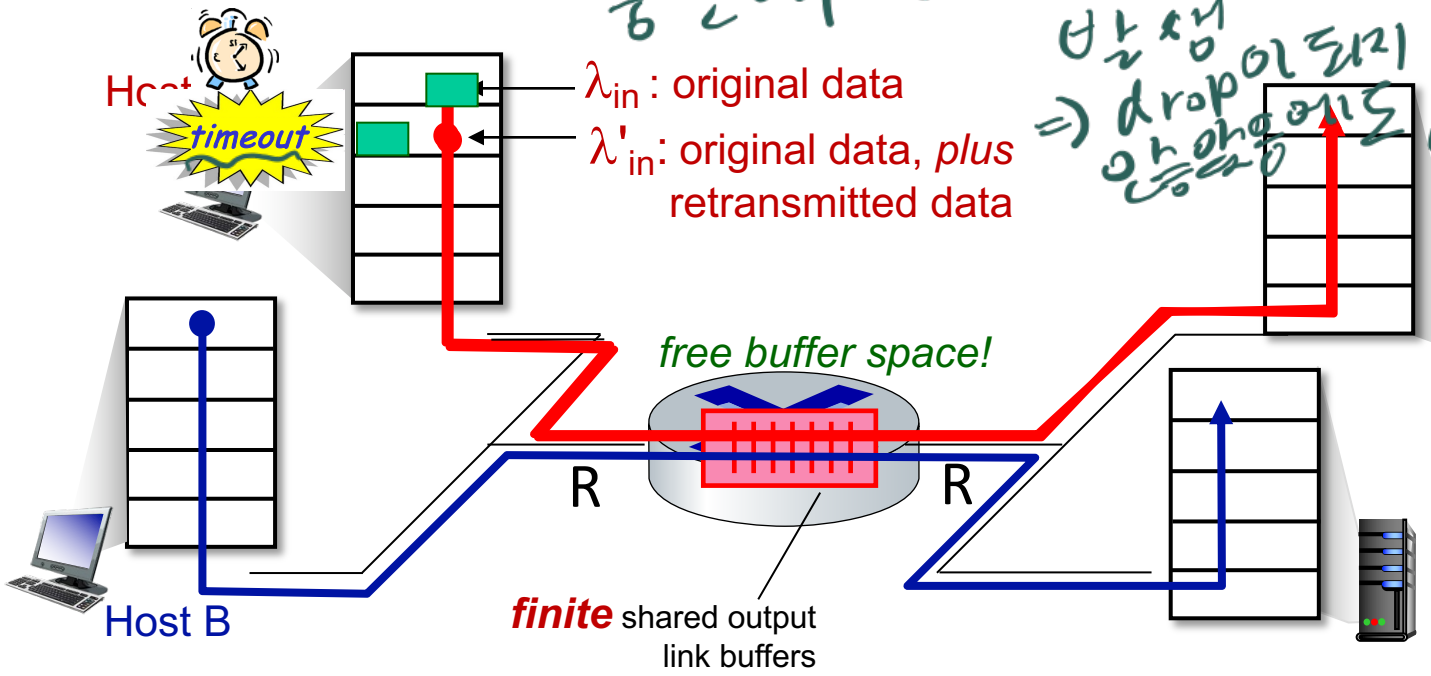


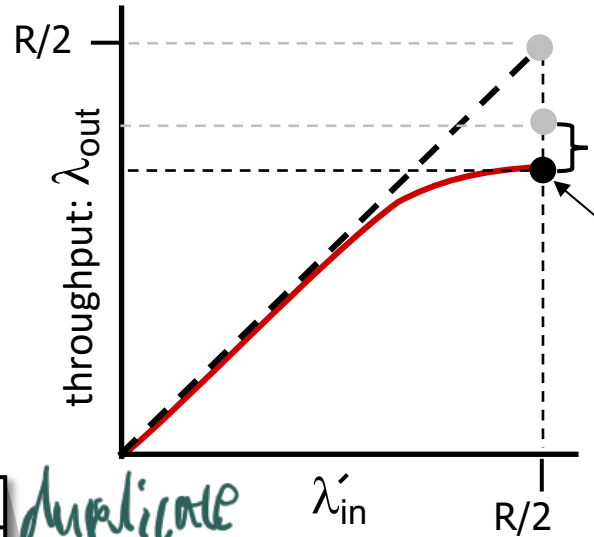


Causes/costs of congestion: scenario 2

Realistic scenario: *un-needed duplicates*

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





"wasted" capacity due to un-needed retransmissions

when sending at R/2, some packets are retransmissions, including needed and *un-needed* duplicates, that are delivered!

Causes/costs of congestion: scenario 2

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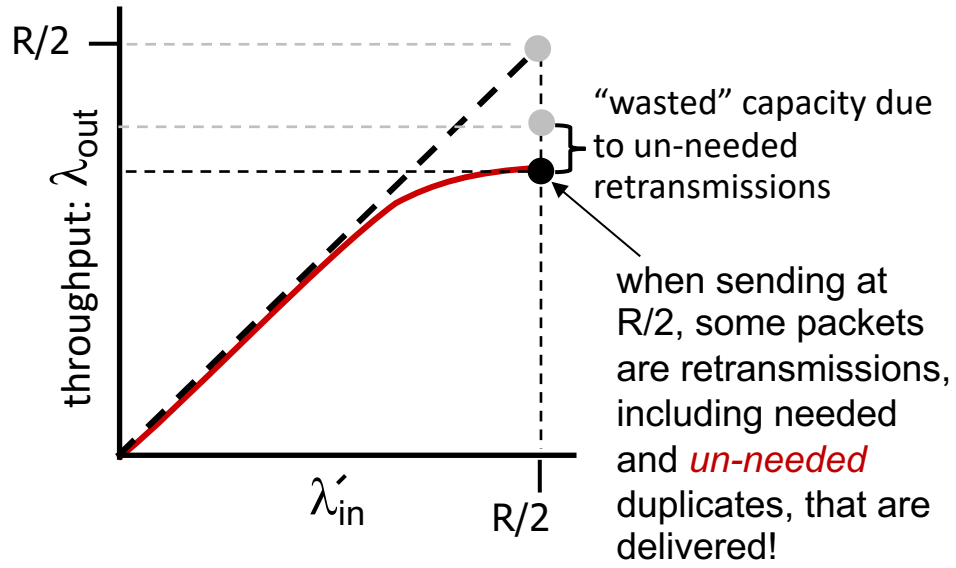


"costs" of congestion:

- more work (retransmission) for given receiver throughput
- unneeded retransmissions: link carries multiple copies of a packet
  - decreasing maximum achievable throughput

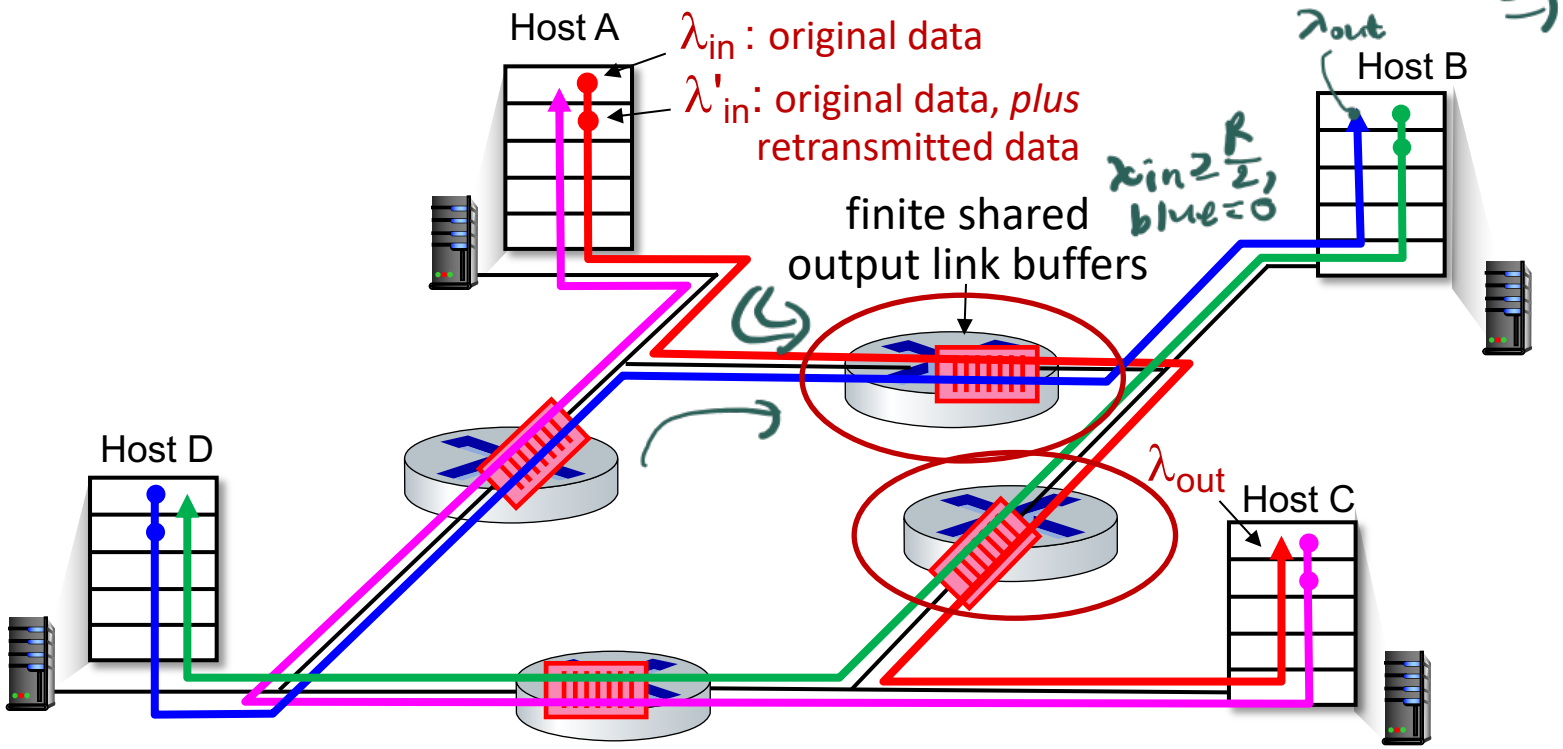
↳ 불필요한 재전송으로 처리량 ↓

Causes/costs of congestion: scenario 3

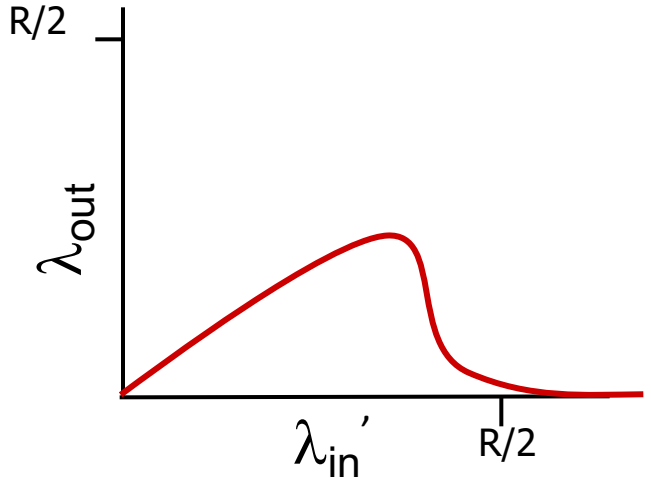
- *four* senders
- *multi-hop* paths
- timeout/retransmit

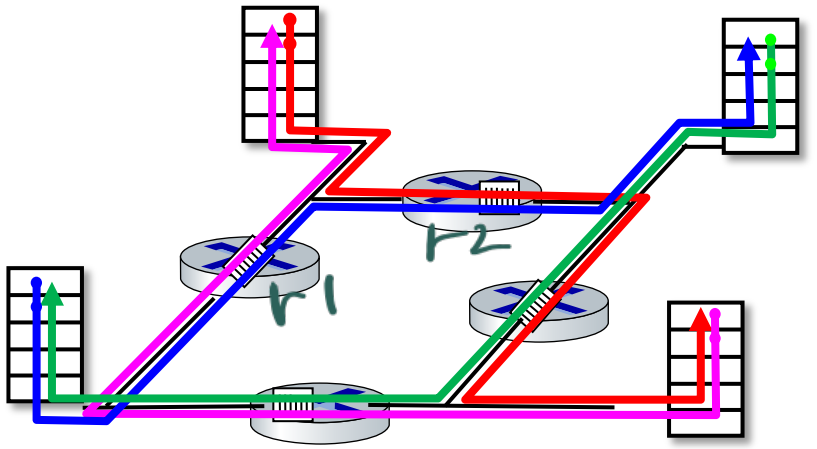
Q: what happens as λ_{in} and λ_{in}' increase ?

A: as red λ_{in}' increases, all arriving blue pkts at upper queue are dropped, blue throughput → 0



Causes/costs of congestion: scenario 3





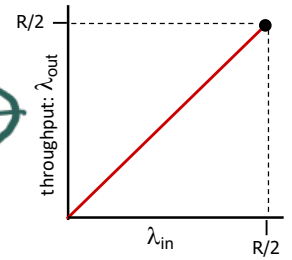
blue는 R1의 자원을,
but R2에서 전부 drop되며
R1 자원도 낭비

another "cost" of congestion:

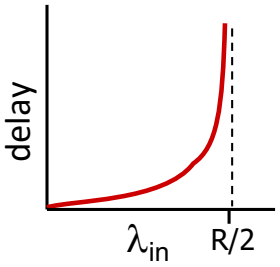
- when packet dropped, any upstream transmission capacity and buffering used for that packet was wasted!

Causes/costs of congestion: insights

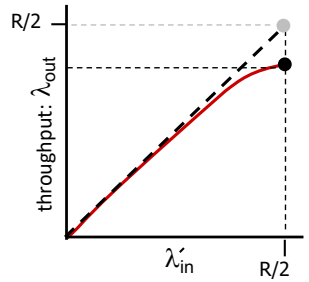
- throughput can never exceed capacity



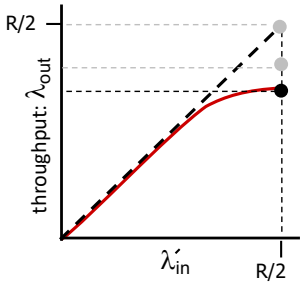
- delay increases as capacity approached



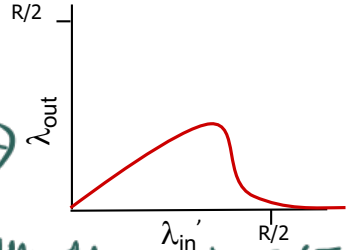
- loss/retransmission decreases effective throughput



- un-needed duplicates further decreases effective throughput



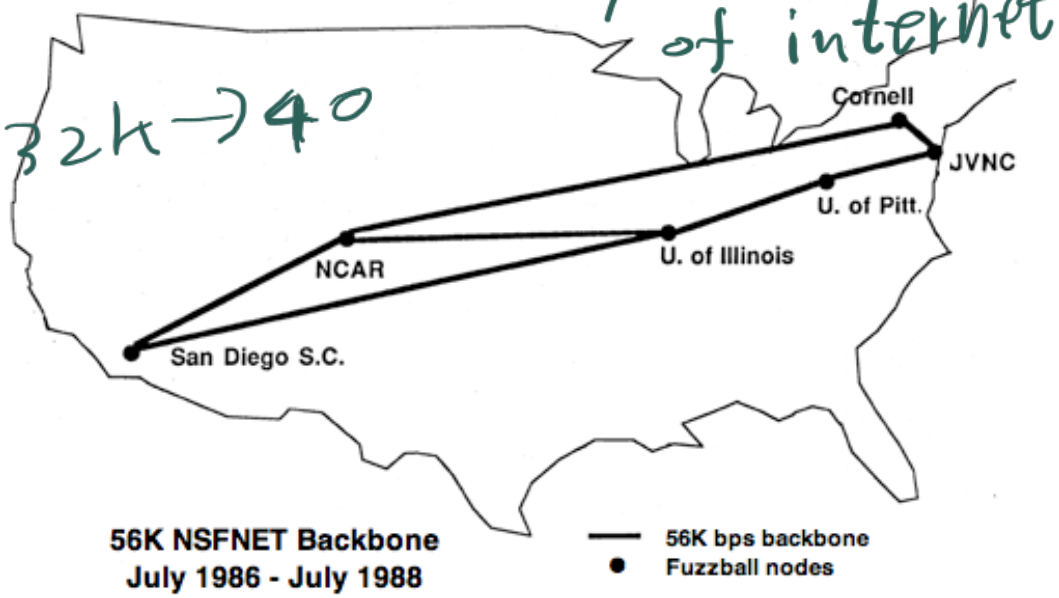
- upstream transmission capacity / buffering wasted for packets lost downstream



“Congestion Collapse”

- In October 1986, NSFNET backbone capacity dropped from 32 kbps to 40 bps
- TCP congestion control invented between 1987-1988 by Van Jacobson and Sally Floyd

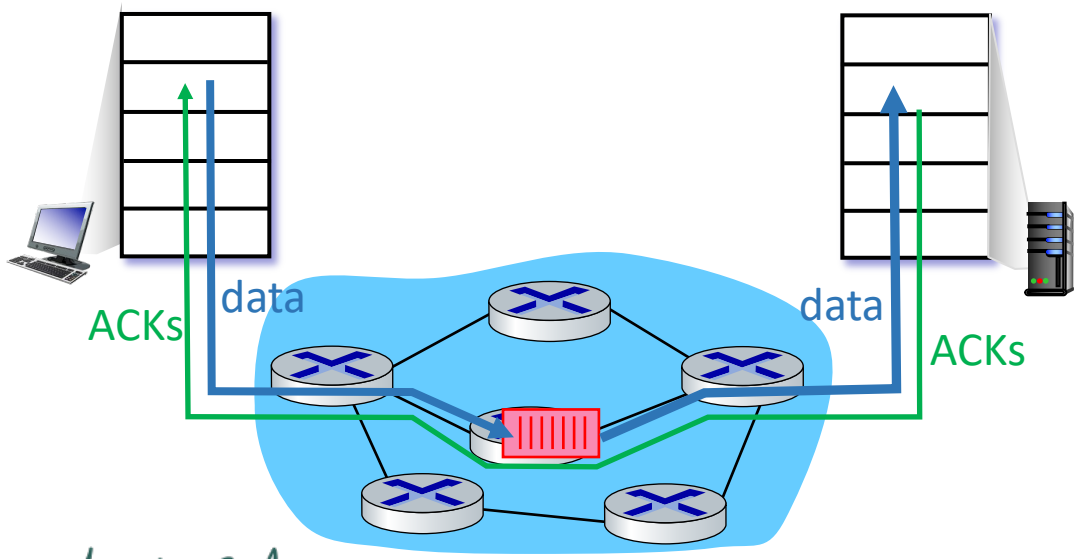
(no congestion control)
early version of internet
32k → 40



Approaches towards congestion control

End-end congestion control:

- no explicit feedback from network
- congestion *inferred* from observed loss, delay
  - approach taken by TCP

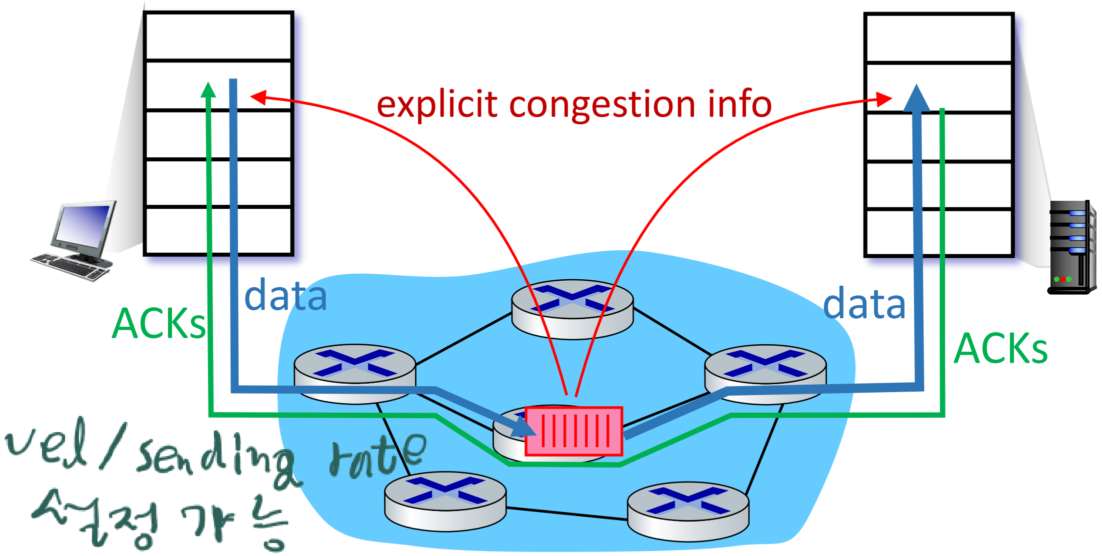


⇒ congestion 여부를 명시적으로 알려주지 않고 loss, delay를 통해 추론

Approaches towards congestion control

Network-assisted congestion control:

- routers provide *direct* feedback to sending/receiving hosts with flows passing through congested router 지시한다 → congestion level/sending rate 설정 가능
- may indicate congestion level or explicitly set sending rate



- TCP ECN, ATM, DECbit protocols

라우터가 host에 congestion 여부 명시적으로 전달

Next…

- *Chapter 3.7 TCP Congestion Control*