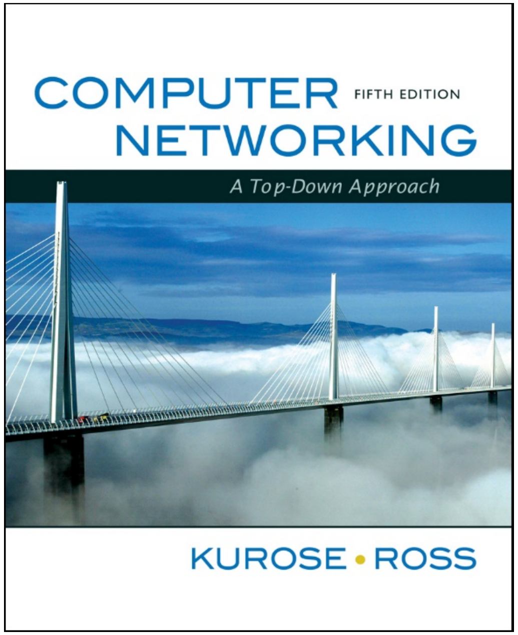
Chapter 3
Transport Layer



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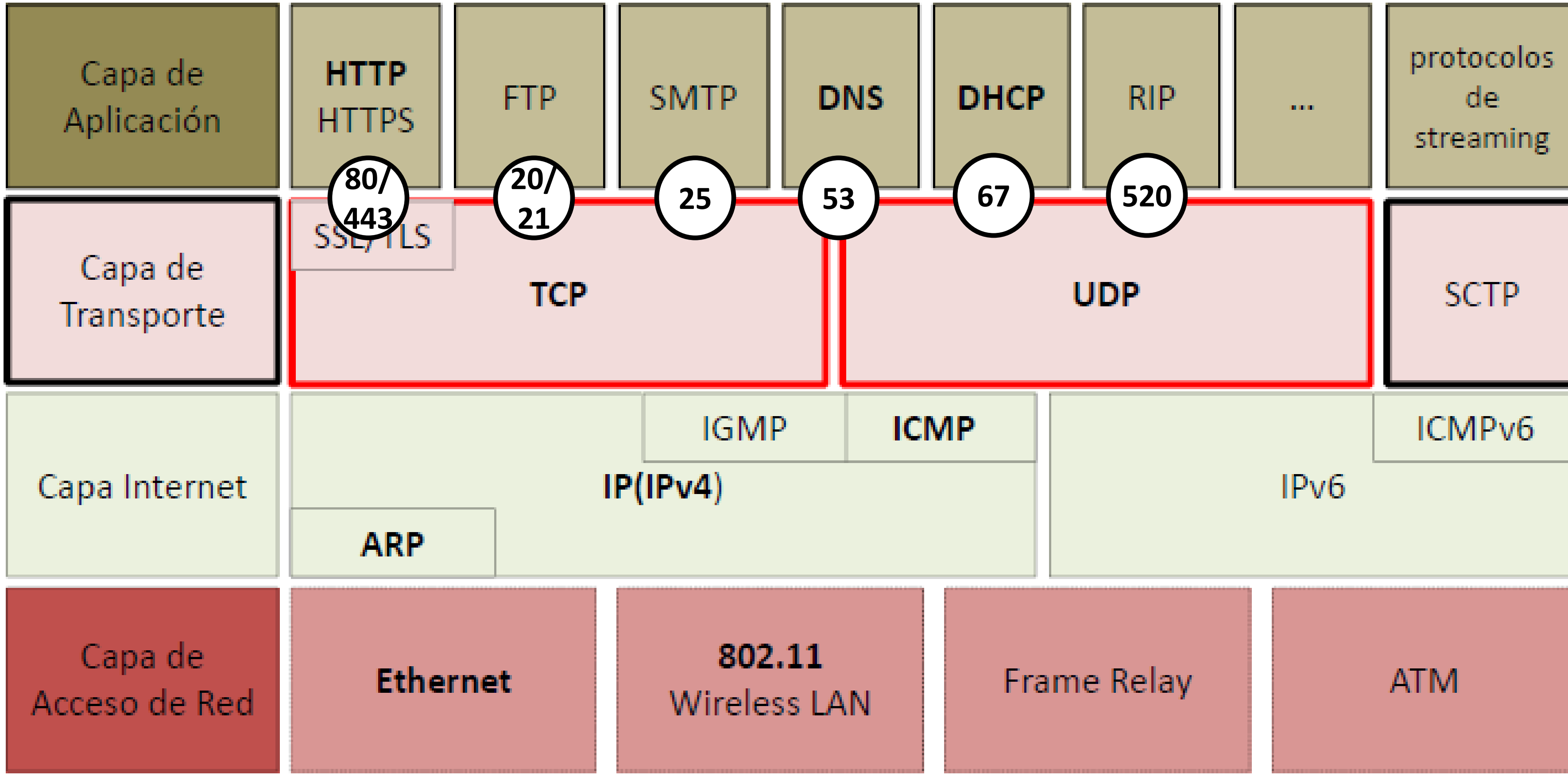
Thanks and enjoy! JFK/KWR

Computer Networking: A Top Down Approach
5th edition.
Jim Kurose, Keith Ross
Addison-Wesley, April 2009.

3 Gaia: Garraio geruza



3 Gaia: Garraio geruza

Helburua:

- Garraio geruzak ematen dituen zerbitzuak ulermena:
  - multiplexing/demultiplexing
  - Informazioaren garraio fidagarria
  - Fluxu kontrola
  - Pilaketen kontrola
- Garraio geruza Interneten:
  - UDP: konexiorik gabe
  - TCP: konexiora orientatuta
  - TCP pilaketen kontrola

3. Gaia:

- 3.1 Garraio geruzaren zerbitzuak
- 3.2 Multiplexing and demultiplexing
- 3.3 Konexiorik gabeko garraioa: UDP
- 3.4 Informazio garraio fidagarriaren oinarriak
- 3.5 Konexiorako bideratutako transportea: TCP
  - Segmentuen estruktura
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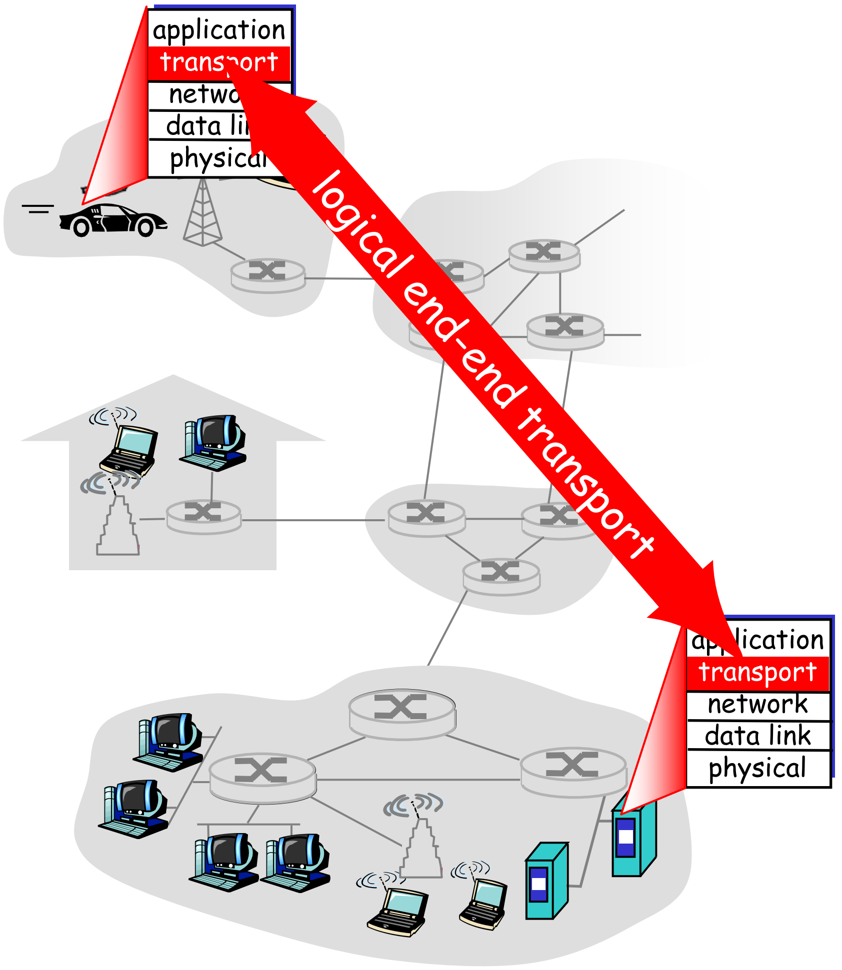
Garraio zerbitzu eta protokoloak

- *Komunikazio logikoa* gauzatzen du host desberdinetan dauden prozesuen artean
- Garraio protokoloak terminaletan lan egiten dute
  - Bidaltzaileak: Aplikazioaren mezuak segmentuetan zatikatzen ditu eta sare geruzara pasatzen ditu
  - Jasotzaileak: jasotako segmentuekin mezuak berrosatzen ditu eta aplikazio geruzara pasatzen ditu
- Protokolo desberdinak daude
  - Interneten: TCP eta UDP



Garraio geruza vs sare geruza

- *Sare geruza:* Komunikazio logikoa host artean
- *Garraio geruza:* Komunikazio logikoa prozesuen artean
  - Sare geruzak dituen zerbitzuaz fido da hauek hobetzen

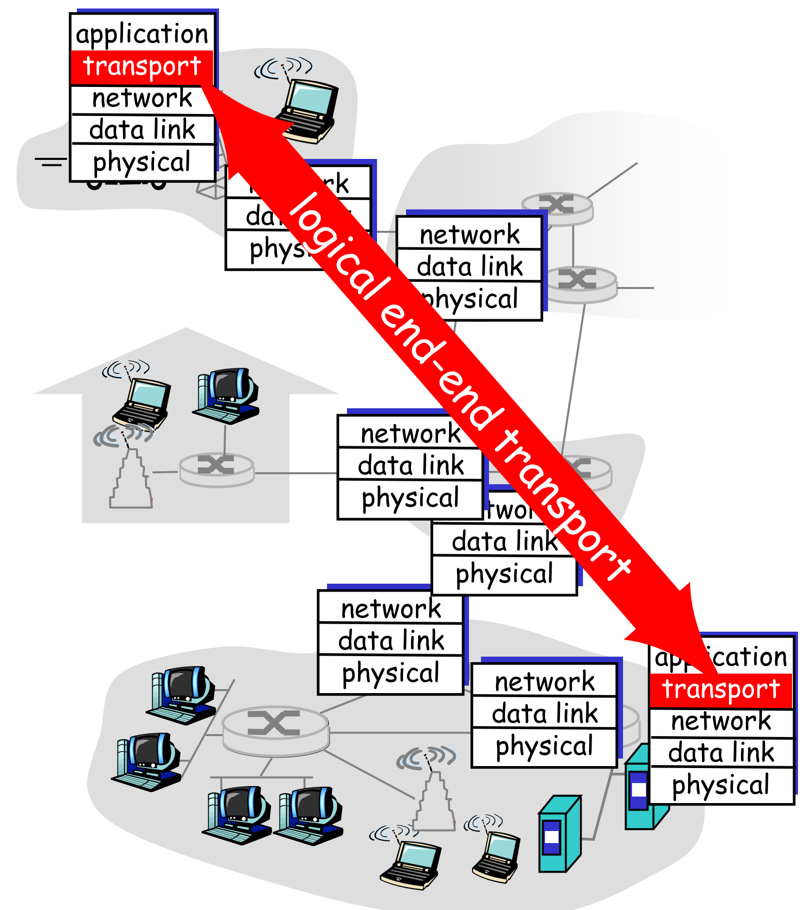
Household analogy (liburuan):

12 kids sending letters to 12 kids

- processes = kids
- app messages = letters in envelopes
- hosts = houses
- transport protocol = Ann and Bill
- network-layer protocol = postal service

Internetaren garraio geruzaren protokoloak

- Bideraketa ordenatua, fidagarria (TCP)
  - Pilaketen kontrola
  - Fluxuaren kontrola
  - Konexioa
- Bideraketa ez-ordenatua, ez-fidagarria : UDP
  - no-frills extension of “best-effort” IP



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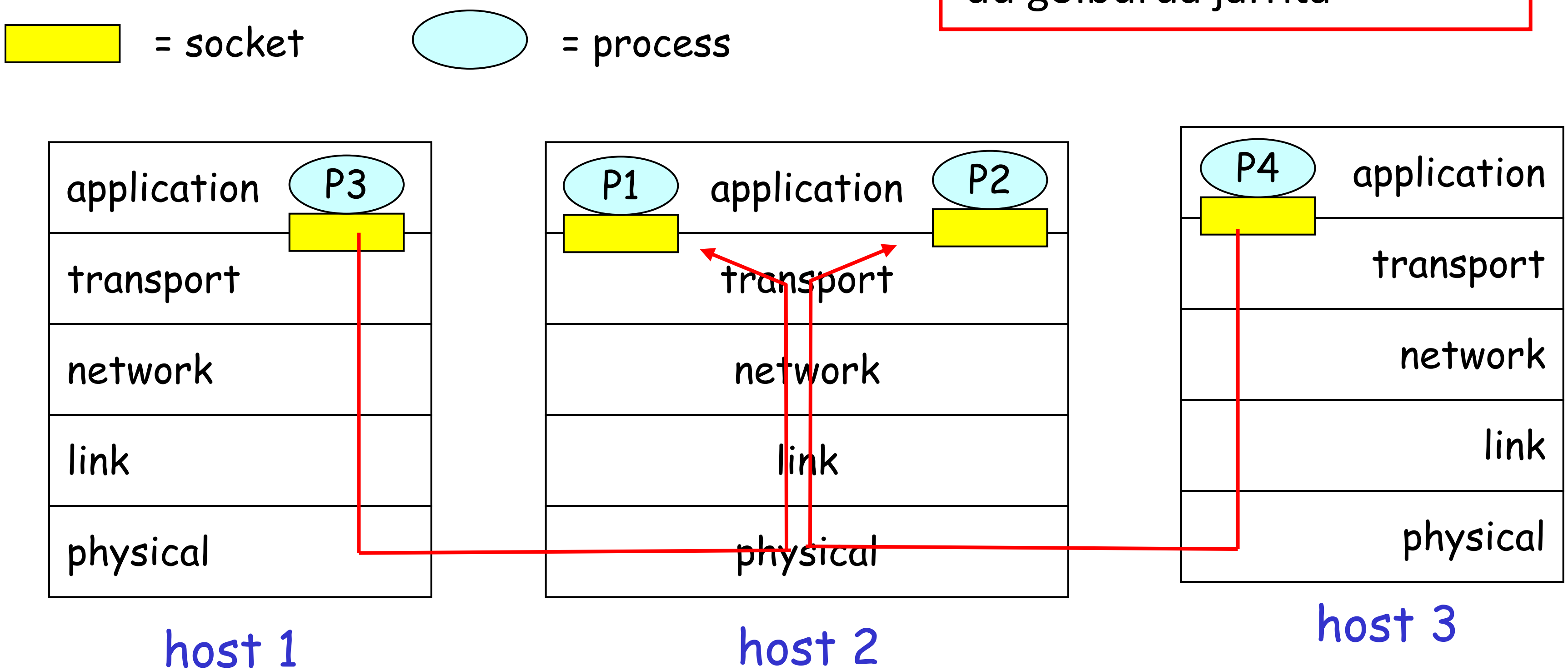
Multiplexing/demultiplexing

Demultiplexazioa jasotzailean

Dagokion socket-i bidaltzen dizkio jasotako segmentuak

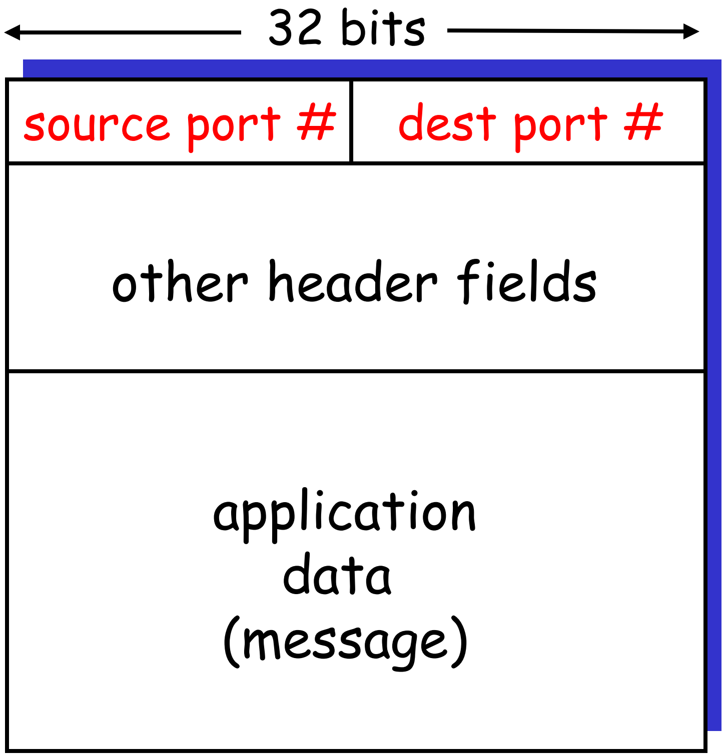
Multiplexazioa igorlean

Informazioa socket desberdinetatik jasotzen du eta segmentuetan banatzen du goiburua jarrita



Nola demultiplexatu

- Host-ak IP datagramak jasotzen ditu
  - Datagrama bakoitzak igorlearen IP helbidea dauka, baita jasotzailearen IP ere
  - Datagrama bakoitzak garraio geruzaren segmentu bat dakar
  - Segmentu bakoitzak igorle eta jasotzailearen portuen zenbakiak dakarzki
- Host-ak IP helbideak eta atakak erabiltzen ditu segmentuak dagokien socket-era bidaltzeko



TCP/UDP segment format

Konexiorik gabeko demultiplexazioa

- Socketak ataka zenbakiarekin sortzen dira:

```
DatagramSocket mySocket1 = new
   DatagramSocket(12534);
DatagramSocket mySocket2 = new
   DatagramSocket(12535);
```

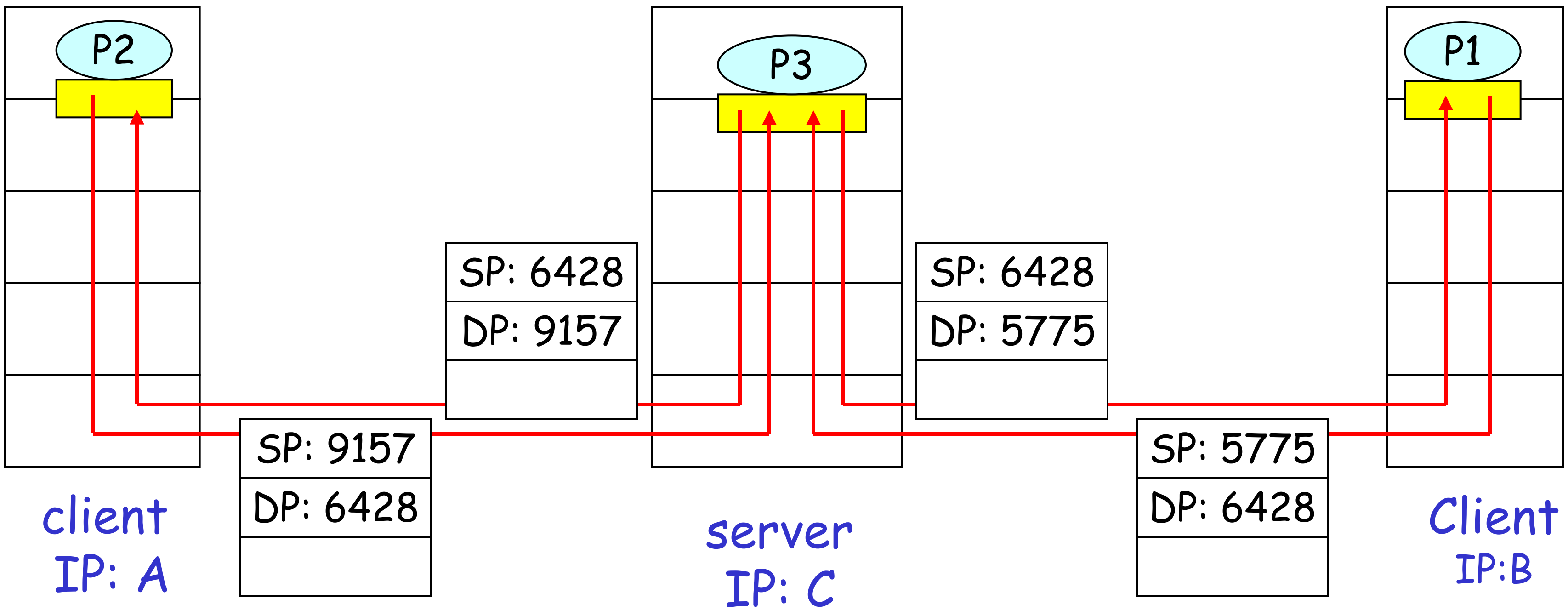
- UDP socketa tupla bikotzaz identifikatzen da:

  (dest IP address, dest port number)

- Hostak UDP segmentua jasotzen duenean:
  - Segmentuaren ataka zenbakia begiratzen du
  - UDP segmentua ataka hori duen socket-era bideratzen du
- IP datagrama igorle desberdinek (IP helbide/ataka desbedinek) bidalitako segmentuak socket berera bidera daitezke

Konexiorik gabeko demux (jarraipena)

```
DatagramSocket serverSocket = new DatagramSocket(6428);
```



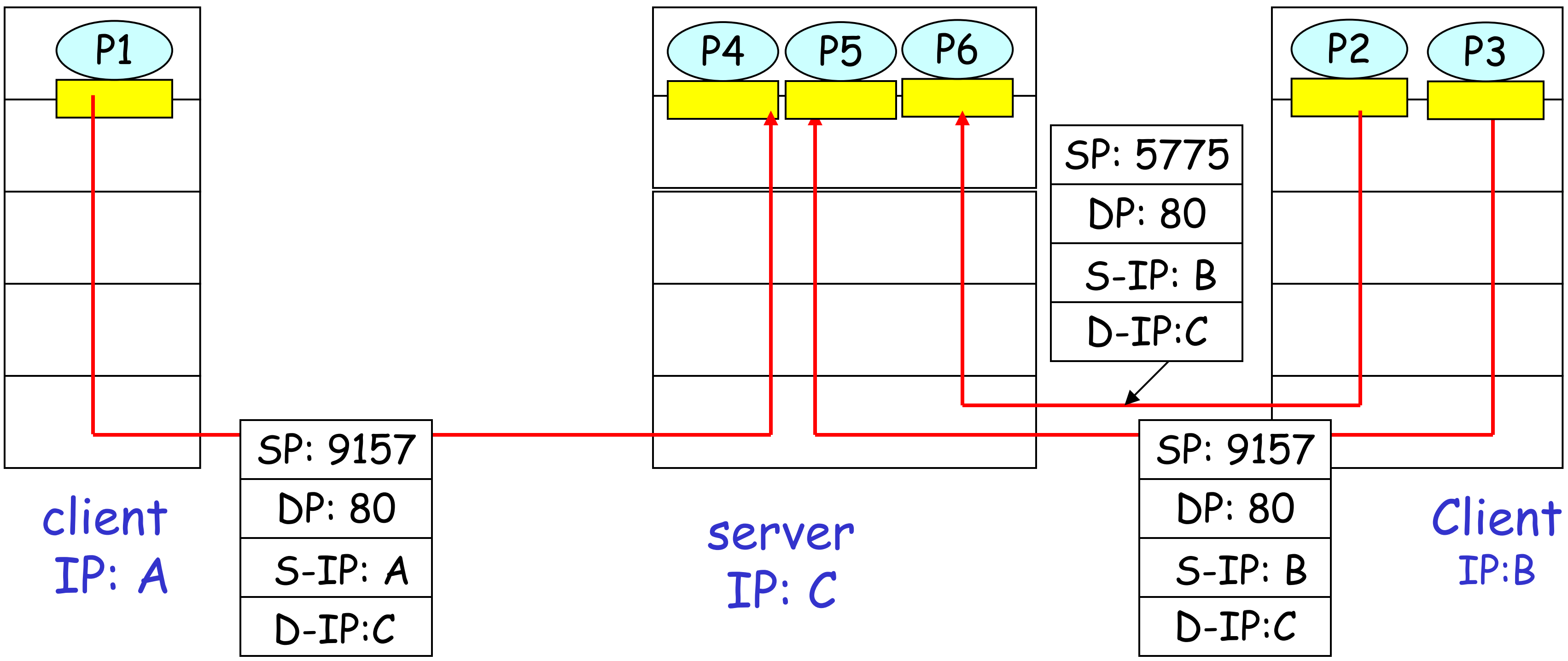
SP (Source Port) “return address” ematen du

Konexiora bideratutako demux

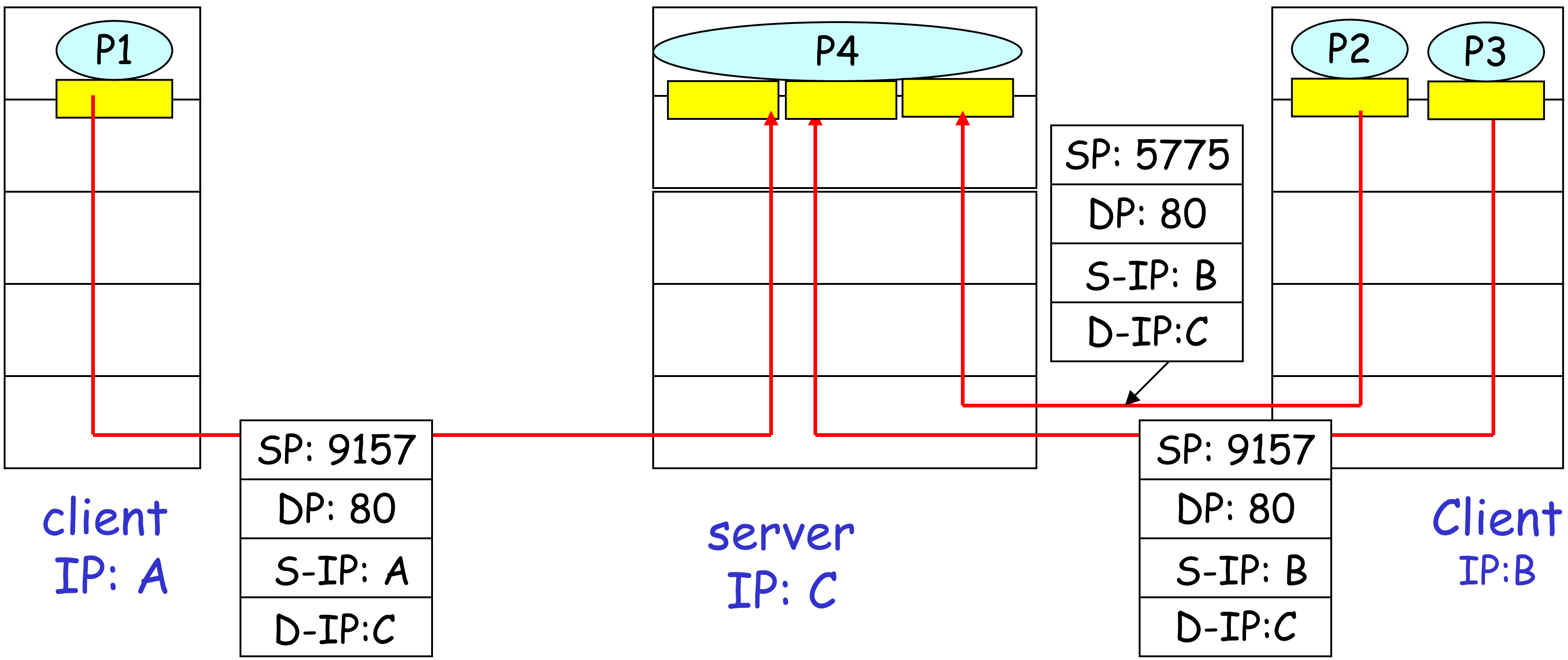
- TCP socketa 4-tupla bidez identifikatuta:
  - source IP address
  - source port number
  - dest IP address
  - dest port number
- Host jasotzaileak lau balioak erabiltzen ditu segmentua dagokion socket-ari bideratzeko

- Zerbitzariek, TCP socket desberdinak jaso dezakete aldi berean:
  - Socket bakoitza horren 4-tupla bidez identifikatzen da
- Web zerbitzariek socket desberdinak dituzte konektatzen den kliente bakoitzeko
  - non-persistent HTTP will have different socket for each request

Konexiora bideratutako demux (jarraipena)



Konexiora bideratutako demux : Threads erabiltzen dituen Web Server-a



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TCP eta UDP Protokoloak: Alderaketa

- Garraio geruzaren protokolorik garrantzitsuenak:
  - TCP: Transmision Control Protocol
  - UDP: User Datagram Protocol
- Aplikazio desberdinen komunikazioak kudeatzen dituzte
- Funtzio desberdinak inplementatzen dituzte:

TCP-ren betekizunak	UDP-ren betekizunak
• Aplikazioen multiplexazioa • Segmentazioa • Akatsen kontrola • Fluxu kontrola • Pilaketen kontrola • Galdutako datuen berbidalketa • Konexio/deskonexioa	• Aplikazioen multiplexazioa • Segmentazioa • Akatsen kontrola (aukeran)

UDP: User Datagram Protocol [RFC 768]

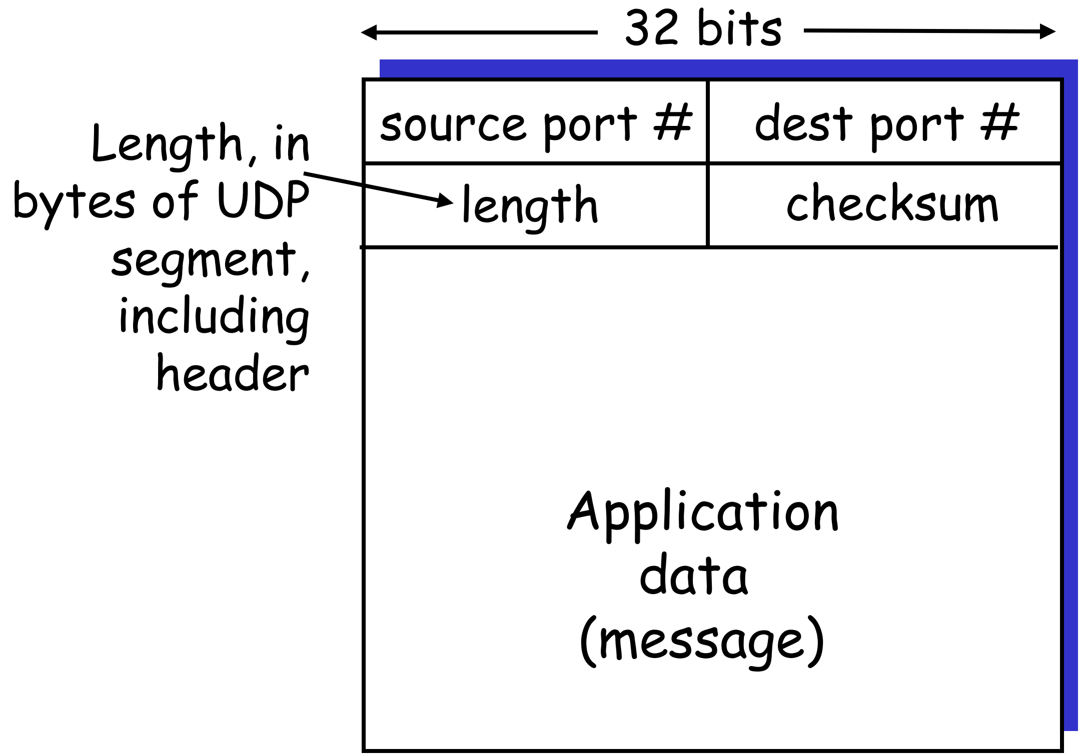
- Interneterako oinarrizko garraio protokoloa
- "best effort" service, UDP segmentuak:
  - Gal daitezke
  - Desordenatuta hel daitezke
- *Konexiorik gabe:*
  - Ez dago konexio protokolorik igorle eta jasotzaile artean
  - UDP segment bakoitzak indibidualki tratatzen da

Zergatik UDP?

- no connection establishment (which can add delay)
- simple: no connection state at sender, receiver
- small segment header
- no congestion control: UDP can blast away as fast as desired

UDP: gehiago

- Streaming aplikazioetan erabilita
  - loss tolerant
  - rate sensitive
- Beste erabilerak
  - DNS
  - SNMP
- Garraio fidagarria UDP gainean: fidagarritasuna aplikazio mailan gehitzen zaio
  - Aplikazio bakoitzak beraren errore kudeaketa!



UDP segment format

bitak	0 – 7	8 – 15	16 – 23	24 – 31
0	Jatorriko helbidea			
32	Helburuko helbidea			
64	Zeroak	Protokoloa	UDP luzera	
96	Jatorriko portua		Helburuko portua	
128	Luzera		Checksum	
160+	Data			

Checksum egiteko kontutan hartzen den informazioa IPv4an

UDP checksum: RFC 768-etan

Helburua: Akatzen datekzioa (e.g., flipped bits) igorritako segmentuetan

Igorleak:

- Segmentuaren informazioa 16-bit-eko integer bezala tratatzen ditu
- checksum: addition (1's complement sum) of segment contents
- Igorleak checksumaren balioa jartzen du UDP segmentuare checksum esparruan

Jasotzaileak:

- Jasotako segmentuen checksum egiten du
- Alderatzen du segmentuak dakarren checksum-arekin:
  - NO - error detected
  - YES - no error detected. *But maybe errors nonetheless?* More later ....

Internet Checksum Example

- Note
  - When adding numbers, a carryout from the most significant bit needs to be added to the result
- Example: add two 16-bit integers

		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
wraparound	1	1	0	1	1	1	0	1	1	1	0	1	1	1	0	1	1
sum		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

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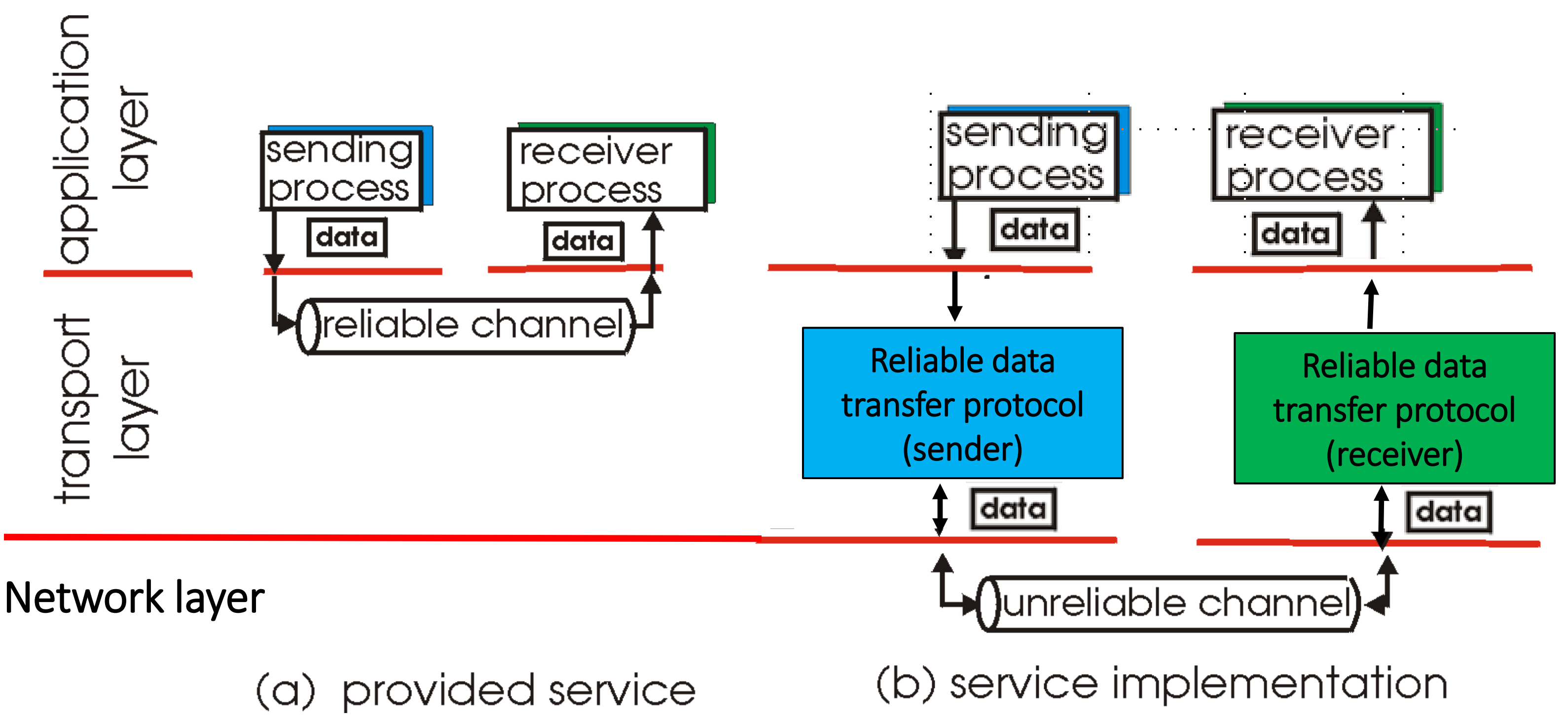
Informazio garraio fidagarriaren oinarriak



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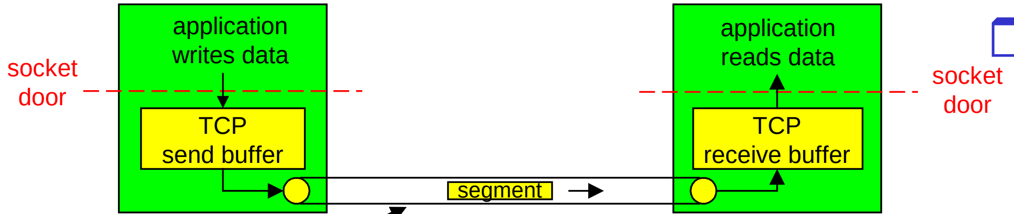
TCP: Overview

RFCs: 793, 1122, 1323, 2018, 2581

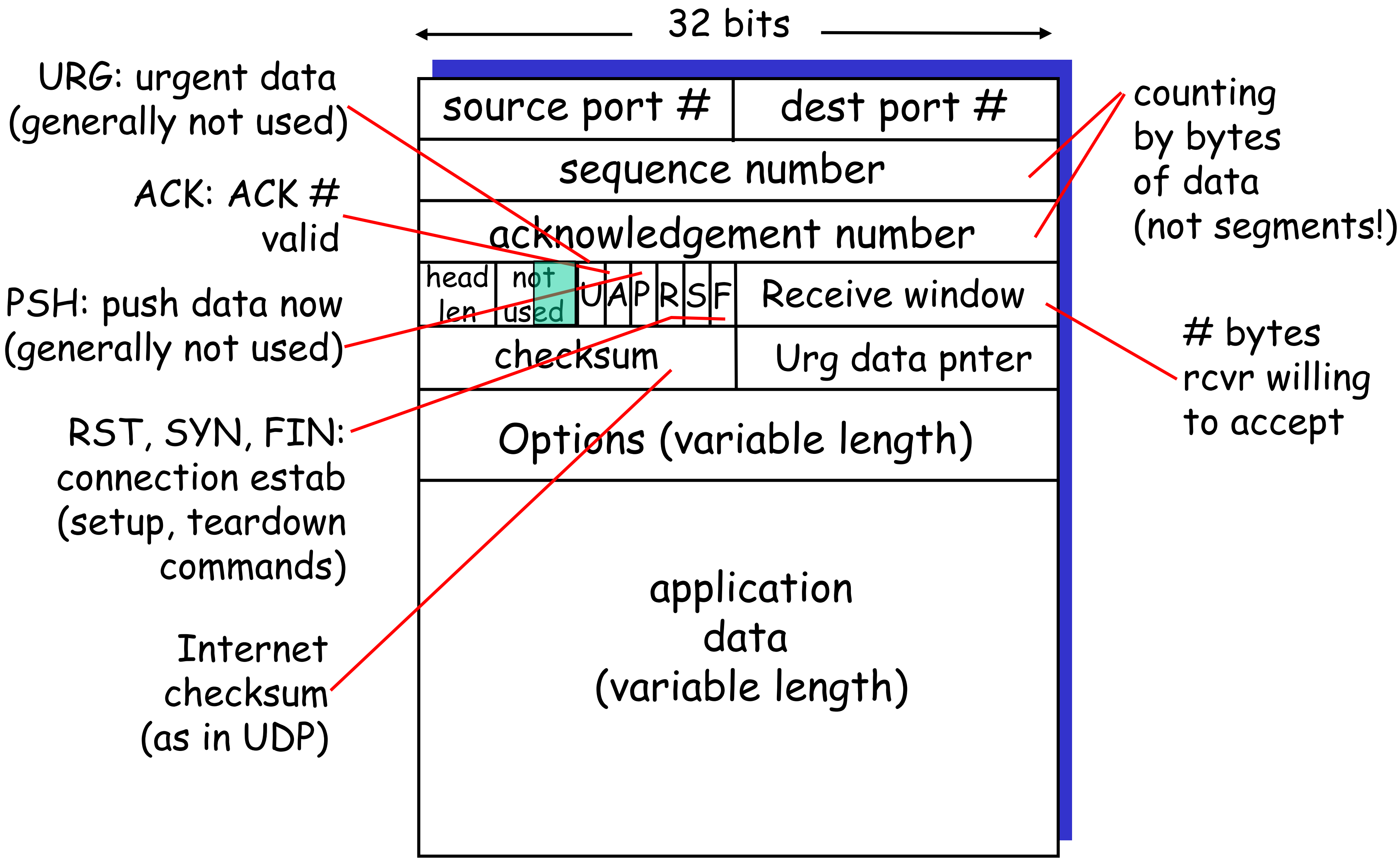
- point-to-point:
  - Igorle bat, jasotzaile bat
- fidagarria, ordenatutako *byte steam:*
- Pipelined, bideratuta:
  - TCPren pilaketa eta fluxu kontrolak lehioaren tamaina finkatzen du
- *igorle & jasotzailearen buffers*



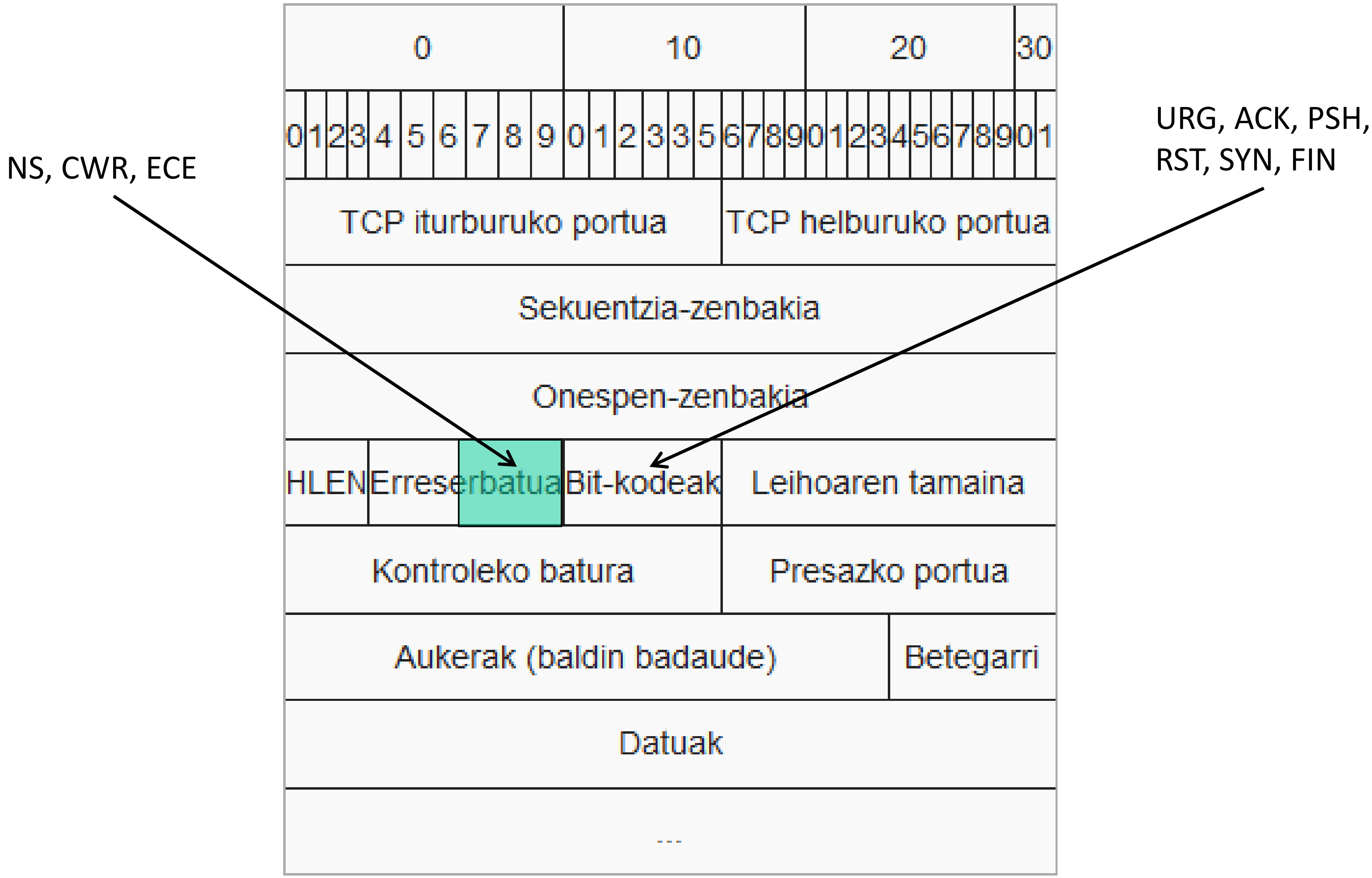
Baina sareko elementuetan ez du ezer inplementatzen

- full duplex data:
  - Informazioaren fluxu bi-directional konexio berean
  - MSS: maximum segment size
- Konexiora orientatuta:
  - **handshaking** (kontrol mezuen trukaketa) igorle eta jasotzailearen egoerarekin informazio transferentzia aurretik
- Fluxuaren kontrola:
  - Igorleak ez du hartzailea "itoko"

TCP segment structure



TCP segment structure (Wikipedia)



TCP segment structure (Wikipedia)

- **TCP iturburuko portua** (16 bit). Iturburuko makinaren portu-zenbakia.
- **TCP helburuko portua** (16 bit). Helburuko makinaren portu-zenbakia.
- **Sekuentzia-zenbakia** (32 bit). **TCP** informazioaren korrontearen barruan uneko datuen posizioa adierazten du.
- **Onespen-zenbakia** (32 bit). Eremu honekin aurretik jasotako byteak zuzen jaso direla adierazteko erabiltzen du helburuko makinak.
- **HLEN** (4 bit). Goiburuko luzera adierazten du. 32 bitetako multiploetan adierazita, segmentuaren tamaina. Gutxienekoa 5 (1001 bitarrean) litzateke, 20 bytetako datu gabeko segmentuari dagokiona.
- **Erreserbatua** (6 bit). Etorkizun batean erabiltzeko.
- **Bit-kodeak** (6 bit). Segmentuaren asmoa zehazten dute.
    - **NS.** Hautazkoa, jasotzaileak erabiltzen du datuak jasotzen dituela jakinarazteko
    - **CWR.** ECE jaso dela jakinarazten du. Pilaketen kontrako tresna erabili du
    - **ECE.** Pilaketa gertatu dela jakinarazten du
    - **URG**. Presazko erakuslearen datuak bilatu eta prozesatu behar dira.
    - **ACK**. Onespen-segmentua da, beraz onespen-zenbakia kontuan hartzen da. SYN eta FIN kodeekin konexioaren ezarpena eta askapena adierazten dute. Segmentu berberak noranzko baten datuak bidal ditzake eta komunikazioaren beste noranzkoaren onespenak.
    - **PSH**. Buffer osoa betetzen den arte segmentua buferretan ez gordetzeko erakuslea, honela iritsi ahala aplikazioari bidaliko zaio. Push eragiketa.
    - **RST**. Uneko konexioa etetea.
    - **SYN**. Konexio eskaera. Sekuentzia-zenbakia nondik hasiko den adierazten du. ACK kodearekin batera (ACK=1;SYN=1) konexio-eskaerari onespena ematen zaio.
    - **FIN**. Konexio amaitu nahi dela adierazteko, ez baitaude datu gehiagorik bidaltzeko.
- **Leihoaren tamaina** (16 bit). Konexio batentzako helburuko makinak onartuko duen buferren tamaina zehazten du.
- **Kontroleko batura** (24 bit). Uneko segmentuaren erroren kontroleko batura. Iturburuko eta helburuko IPren helbideak ere sartzen dira (sasigoiburua).
- **Presazko erakuslea** (8 bit). Presazko datuak non hasten diren zehazten du.
- **Aukerak** (aldakorrak). Aukera hau zehaztuta badago, onartuko den segmenturen gehienezko tamaina zehazten du.
- **Betegarri**. Eremu honekin segmentua 32 bitetako multiplokoa izatea behartuko litzateke.
- **Datuak**. Aplikazioari bidaltzen zaion informazioa.

TCP seq. #'s and ACKs

Seq. #'s:

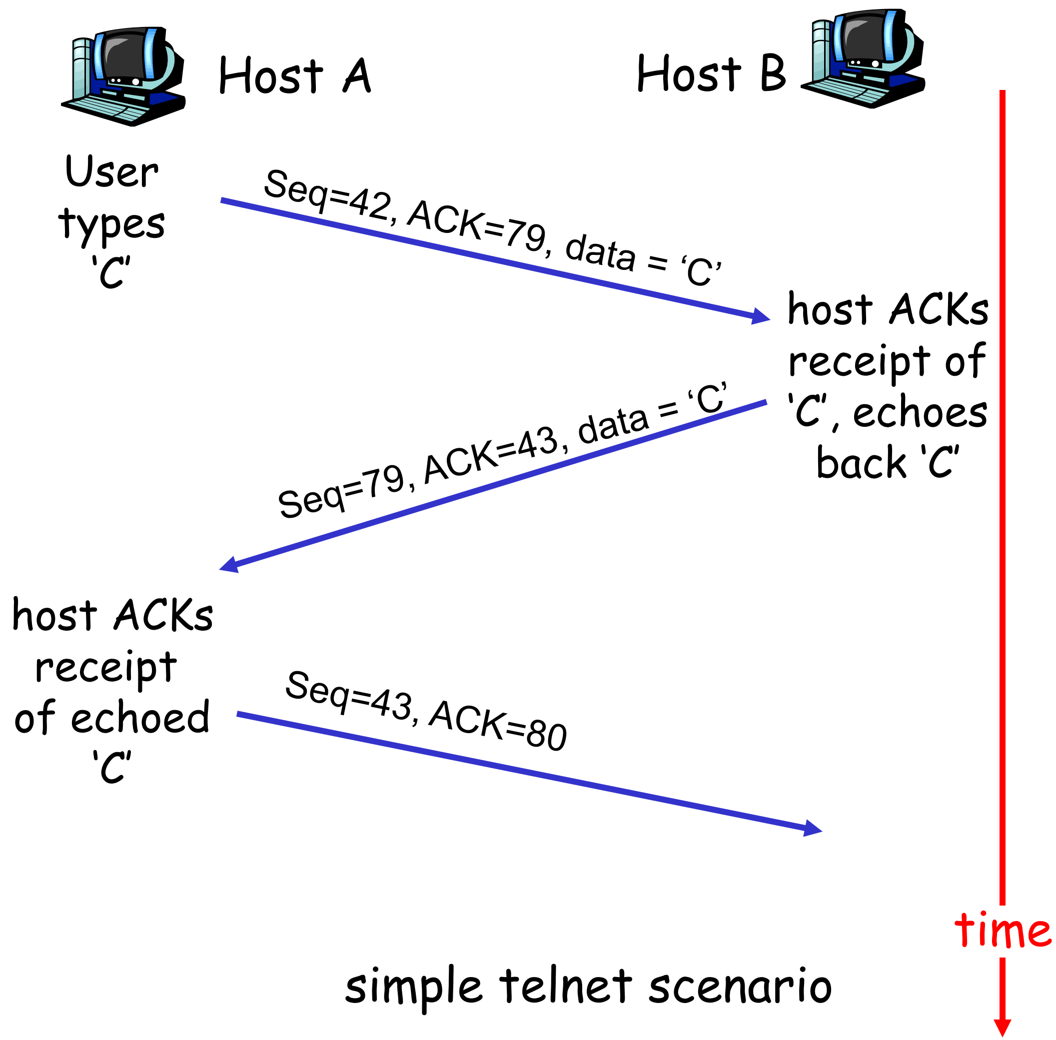
- byte stream "number" of first byte in segment's data

ACKs:

- Beste aldetik espero den hurrengo byte-aren **Seq #**
- cumulative ACK

Q: Nola kudeatzen dira ordenatu gabeko segmentuak?

- A: TCP espezifikazioek ez dute azaltzen, - inplementatzaileak definitu behar



simple telnet scenario

Joan-etorri denbora

TCP Round Trip Time and Timeout

Q: Nola ezarri Timeout balioa?

- RTT baino luzeago
  - baina RTT aldakorra da
- txikiegia: premature timeout
  - Behar ez diren birbidalketak
- luzeegia: segmentuen galeren aurreko erantzun geldoa

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**

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Informazio transferentzia fidagarria

- TCP zerbitzu fidagarria sortzen du IP sareko zerbitzu ez fidagarriaren gainean
- Pipelined segmentuak
- Cumulative acks
- TCP-k erretransmisio tenporizadore sinplea erabiltzen du
- Retransmissions are triggered by:
  - timeout events
  - duplicate acks
- Hasteko, demagun TCP sinplifikatu bat:
  - Bikoiztutako ACK-ak arbuiatzen dira
  - Fluxu eta pilaketa kontrolak arbuiatzen dira

TCP igorlearen gertakariak:

Aplikaziotik jasotako informazioa:

- Segmentua sortu sekuentzia zenbakiarekin **seq #**
- **seq #**: Bidaltzen den segmentuari dagokion informazio fluxuaren lehen byte zenbakia
- Abiarazi erlojua (timer) martxan ez badago (think of timer as for oldest unacked segment)
- expiration interval: `TimeOutInterval`

timeout:

- Berbidali timeout sortarazi duen segmentua
- Erlojua berabiatu

Jasotako ACK:

- Oraindik onartu ez den segmentu baten ACK jasotzen bada
  - Eguneratu onartu (ACK) behar direnen zerrenda
  - Erlojua berabiazi falta diren segmentuentzat

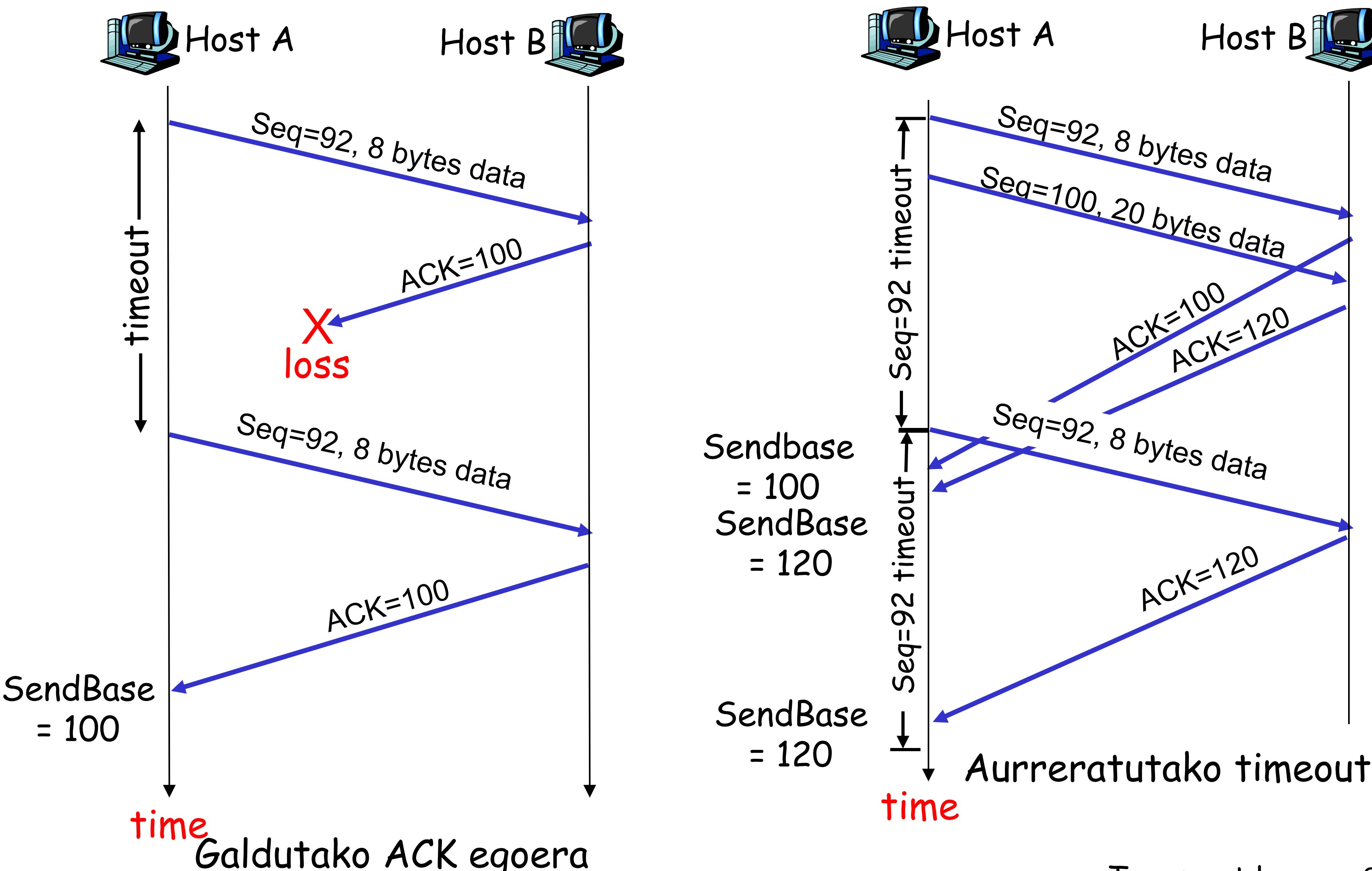
TCP igorle (sinplifikatuta)

```
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 */
```

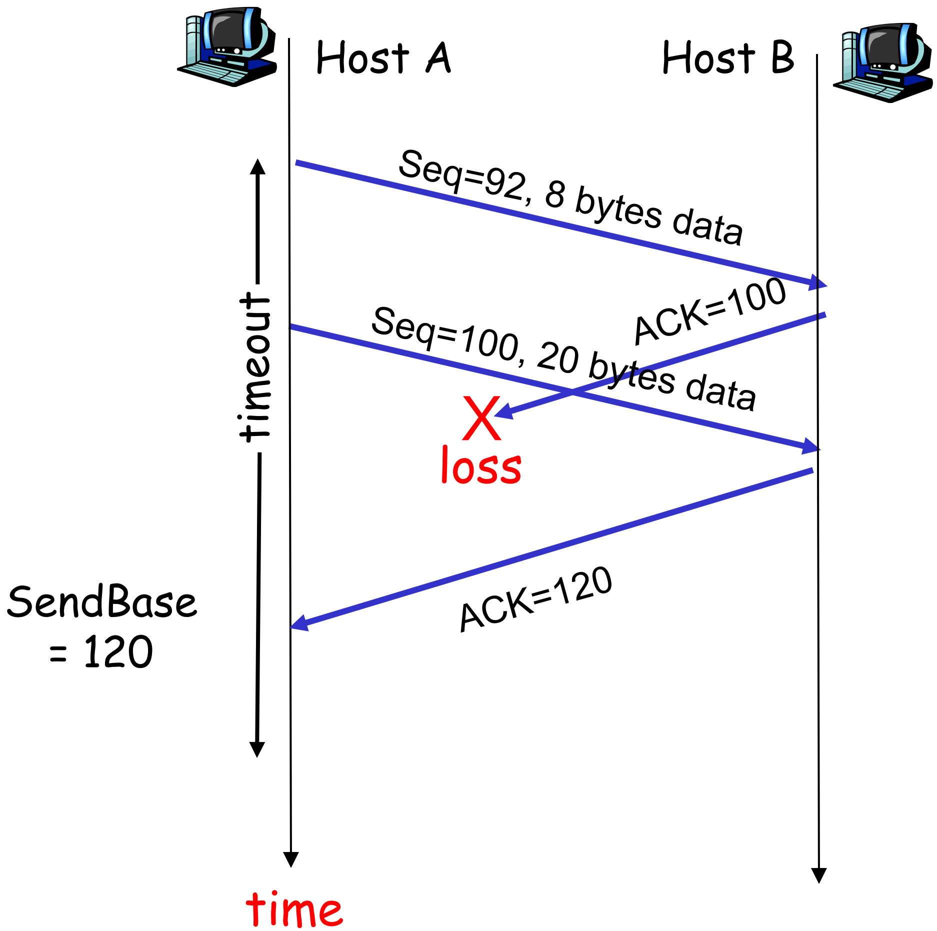
Comment:
- SendBase-1: last cumulatively ack'ed byte

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

TCP: erretransmisio egoerak



TCP: erretransmisio egoerak (gehiago)



Cumulative ACK scenario

TCP: ACK sorrera [RFC 1122, RFC 2581]

Gertaerak jasotzailean	TCP jasotzailearen ekintzak
Segmentuak ordenean heltzen dira espero den **seq** #. Aurreko informazio guztiaren ACK-k bidalita	Atzeratutako ACK. Itzaron beste segmentua 500ms arte. Beste segmenturik ez badago ACK bidali
Segmentuak ordenatuta heltzen dira espero den **seq** #. Beste segmentu Batzuen ACK bidali gabe	Berehala bidali ACK metagarria Honekin errekonozitzen dira horraino helduta ko segmentu guztiak
Heltzen den segmentua desodenatuta Espero den baino **seq** # handiago. Zuloa	Berehala bidali *duplicate ACK*, espero den hurrengo byte-aren **seq** # adieraziz
Zuloa erabat edo zatika betetzen duen segmentu baten iristera	Berehala bidali ACK, espero den segmentuaren **seq** # zuloaren goimuga da

Erretransmisio azkarra

- Time-out denbora, nahiko luzea sarritan:
  - Itzaron-denbora handia segmentua birbidali arte
- Errepikatutako ACK-en bidez galdutako segmentuak detektatu
  - Igorleak batzutan segmentuak bidalttzen ditu bata bestearen atzean
  - Segmentu bat galtzen bada, ACK errepikatuak egon daitezke

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**

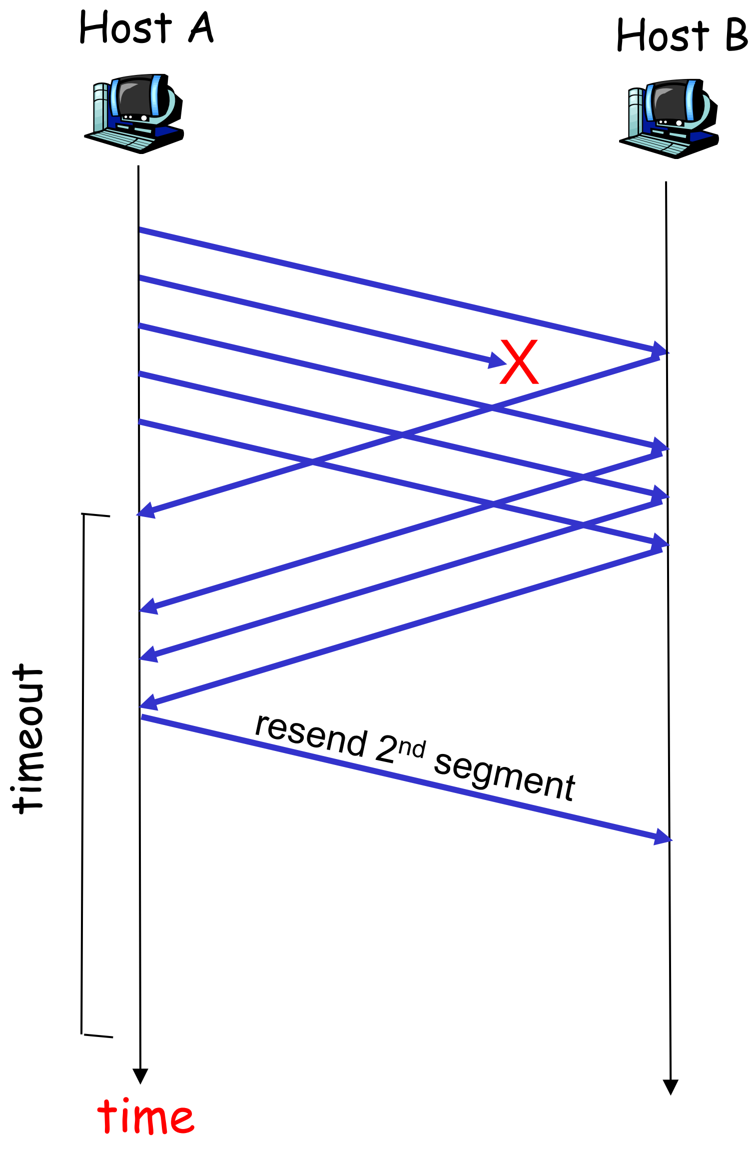


Figure 3.37 Resending a segment after triple duplicate ACK

3. Gaia:

- 3.1 Garraio geruzaren zerbitzuak
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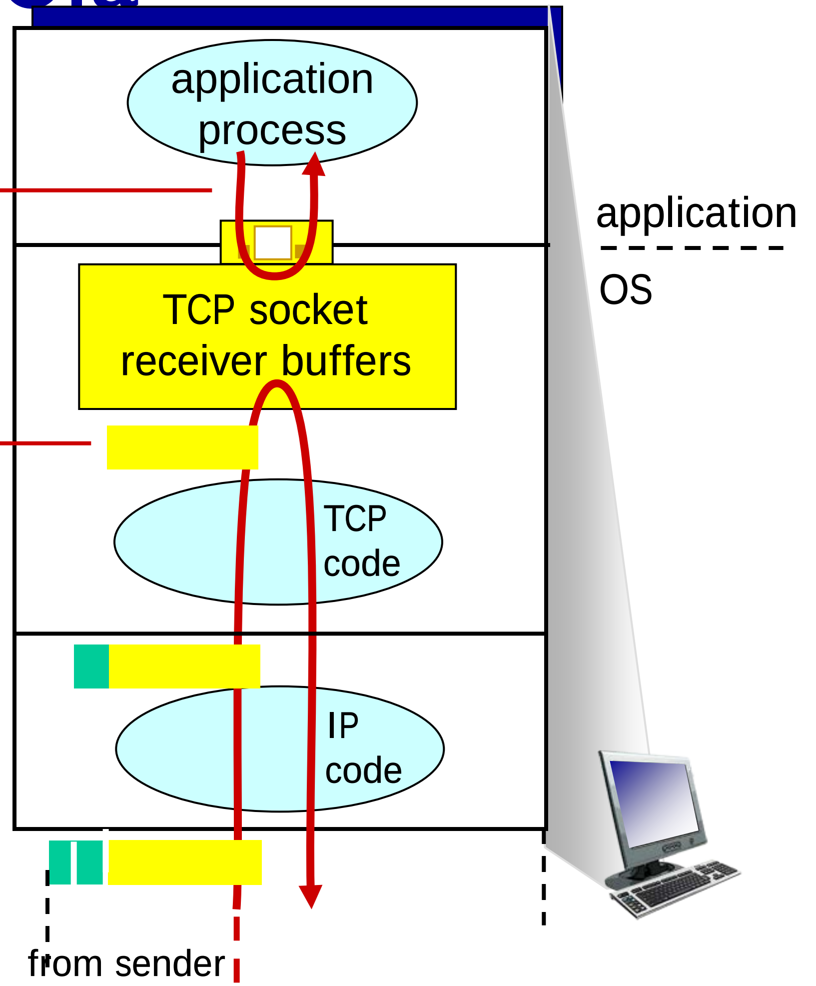
- 3.7 TCP-ren pilaketen kontrola

TCP Fluxuaren kontrola

application may
remove data from
TCP socket buffers ….

… slower than TCP
receiver is delivering
(sender is sending)



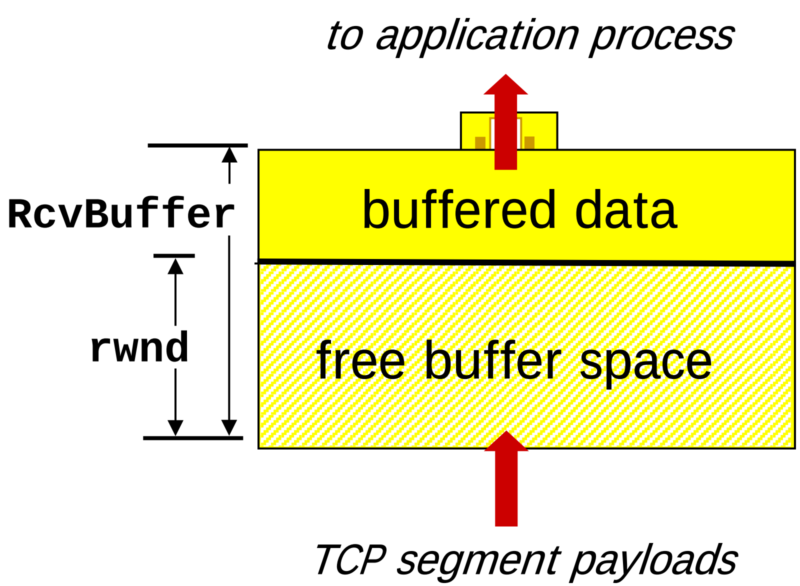
receiver protocol stack

flow control

Jasotzaileak igorlea kontrolatzen du, igorleak ez du jasotzailearen bufferra beteko arinegi transmitituz

TCP flow control

- Jasotzaileak "jakinarazten" du buffer-ean duen tokia, TCP goiburuan **`rwnd`** balioa sartuta *receiver-to-sender* segmentuetan
  - **`RcvBuffer`** balioa socketen aukeren bidez ezartzen da (balio tipikoa 4096 bytes da)
  - Sistema eragile batzuek **`RcvBuffer`**-en balioa ajustatzen du
- Igorleak ez-ACK informazioa mugatzen du jasotzailearen **`rwnd`** balioaren bidez
- Jasotzailearen buferra ez dela gaineratuko bermatzen du



receiver-side buffering

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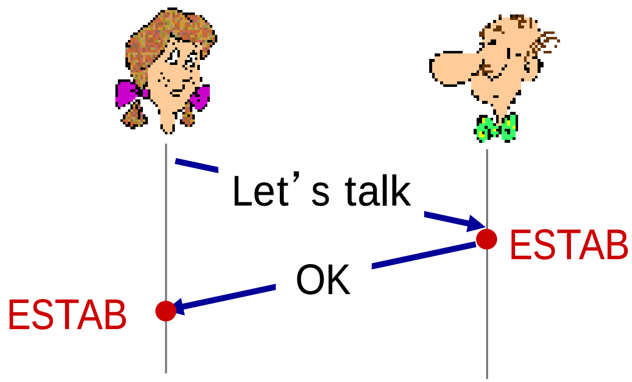
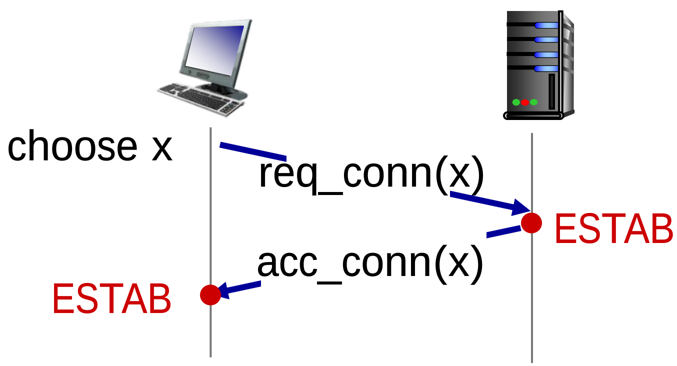
Konexio bat onartzen

2-way handshake:





Q: sareetan funtzionatuko du beti?

- Atzerapen aldakorrak
- Birbidalitako mezuak (e.g. req_conn(x)) mezuak galtzen direnean
- Mezuen berantolaketa
- Beste aldea ezin ikusi

TCP Konexioaren kudeaketa

Recall: TCP igorleak eta jasotzaileak konexioa ezartzen dute informazio segmentuak trukatzen hasi aurretik

- TCP aldagaien hasieratze:
  - seq. #s
  - buffers, fluxu kontrolaren informaziorako (ad. **RcvWindow**)
- *client:* konexioa abiaraztendu

Socket clientSocket = new Socket("hostname","port number");

- *server:* erantzuten du

Socket connectionSocket = welcomeSocket.accept();

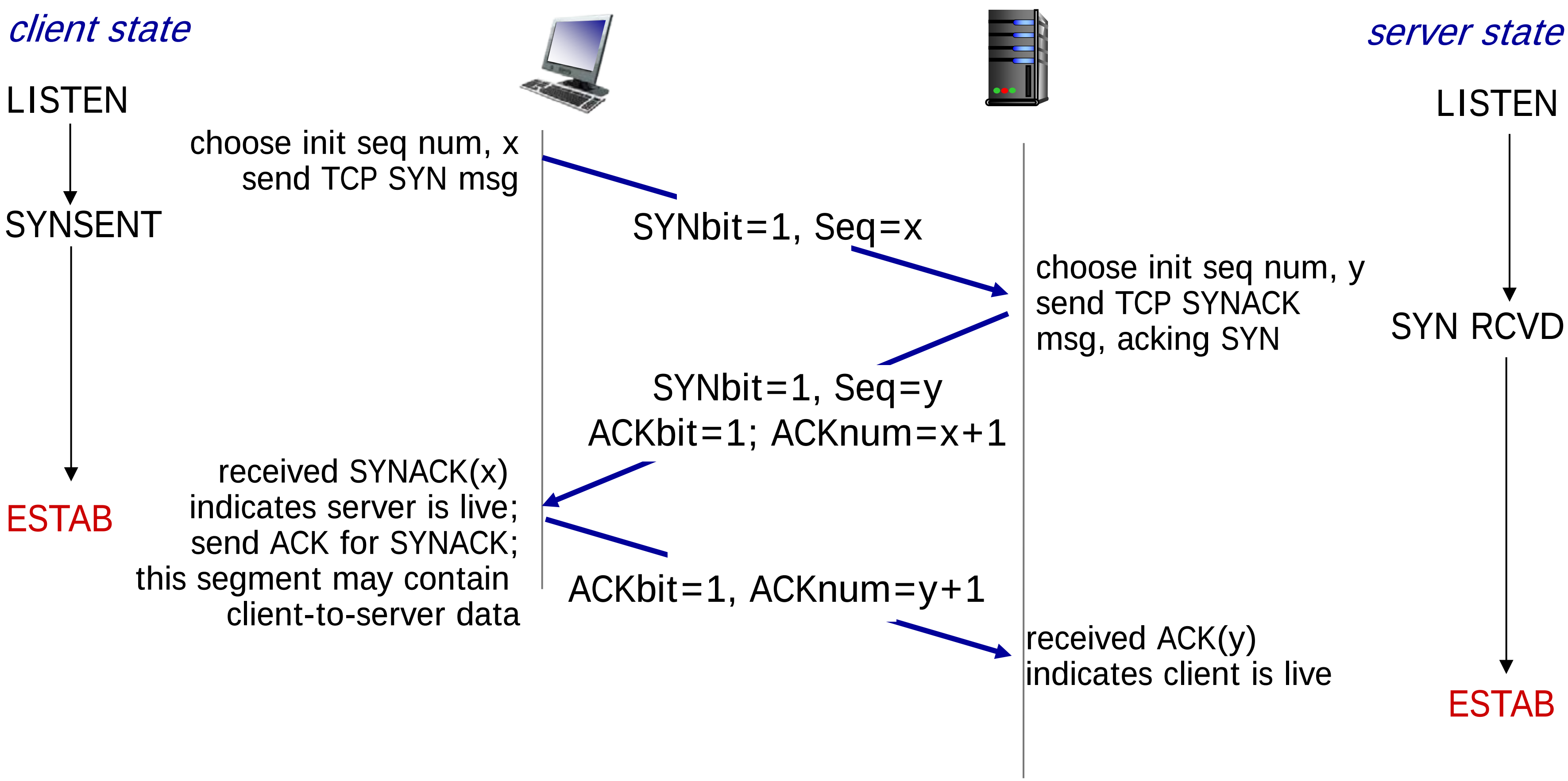
Three way handshake:

1. Pausua: bezeroak TCP SYN segmentua bidaltzen dio zerbitzariari
   - Hasierako **seq #**-ekin
   - Informazioarik gabe

2. Pausua: Zerbitzariak SYN jasotzen du, SYNACK segmentuarekin erantzuten du:
   - Zerbitzariak gordetako bufferekin
   - Zerbitzariaren hasierako **seq #**

3. Pausua: bezeroak SYNACK jasotzen du, ACK segmentuarekin erantzuten du, honek informazioa eduki dezake

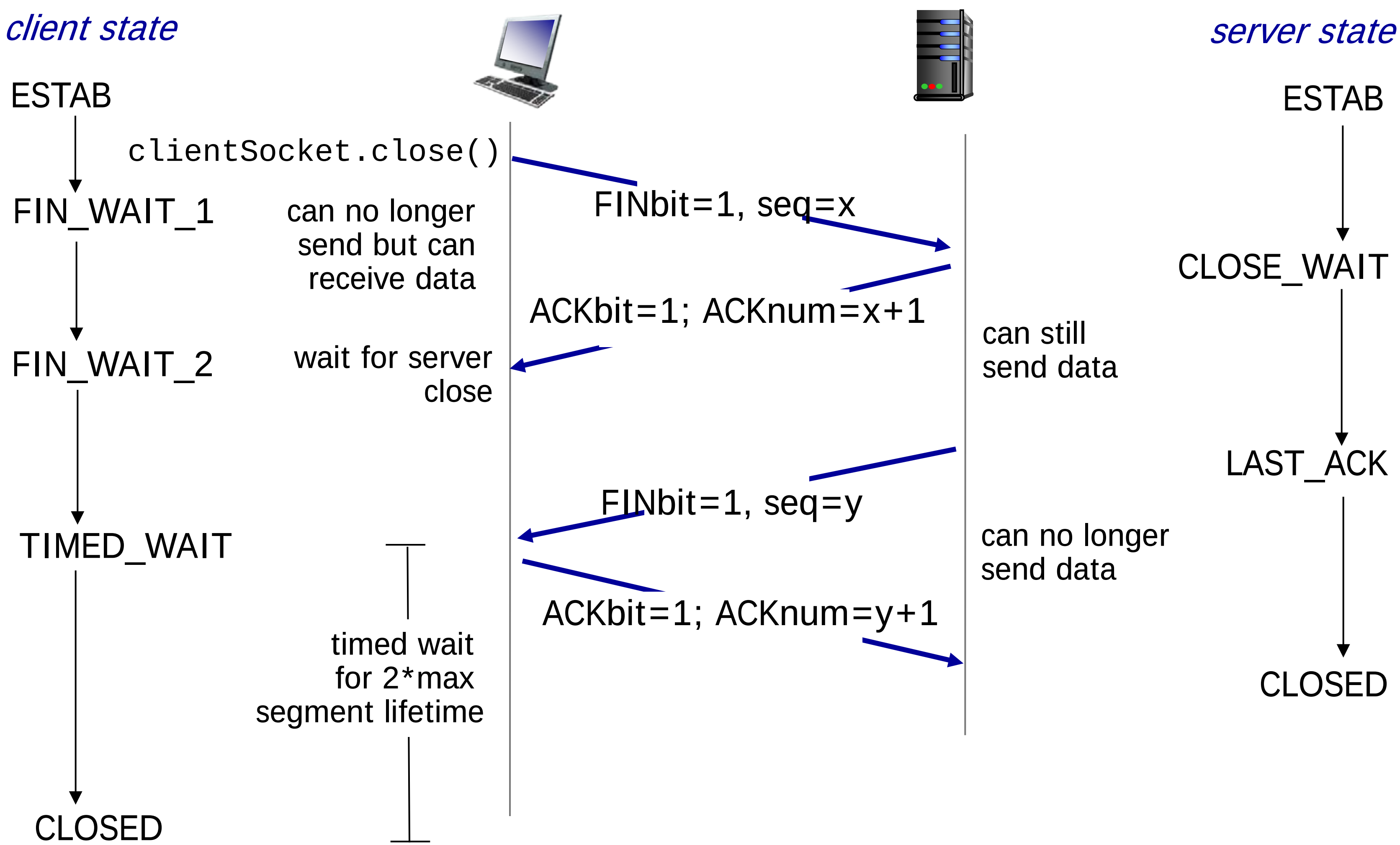
TCP 3-way handshake



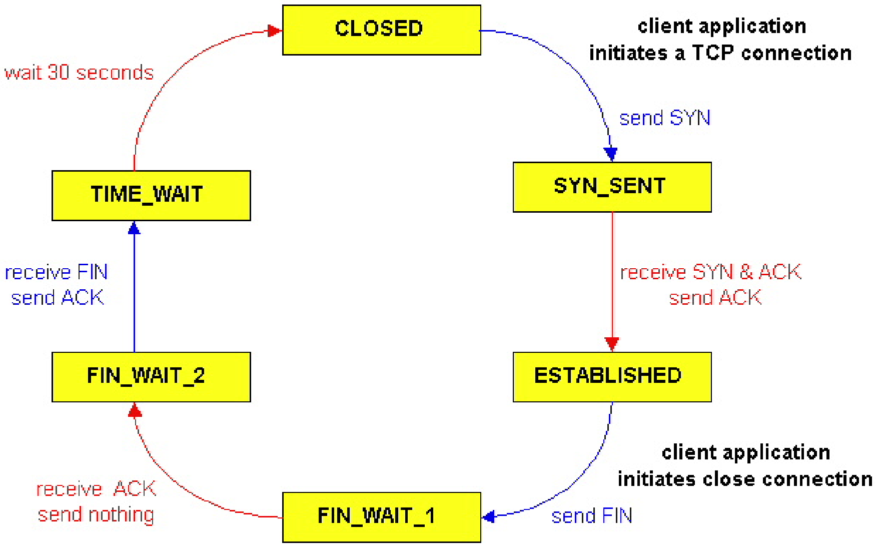
TCP: closing a connection

- Bezeroak eta zerbitzariak, biak izten dute konexioa beraren aldean
  - send TCP segment with FIN bit = 1
- Erantzun jasotako FIN, ACK-rekin
  - FIN jasotzen denean, ACK beste aldeko FIN-ekin nahas daiteke
- Batera ematen diren FIN trukaketak kudea daitezke

TCP: closing a connection

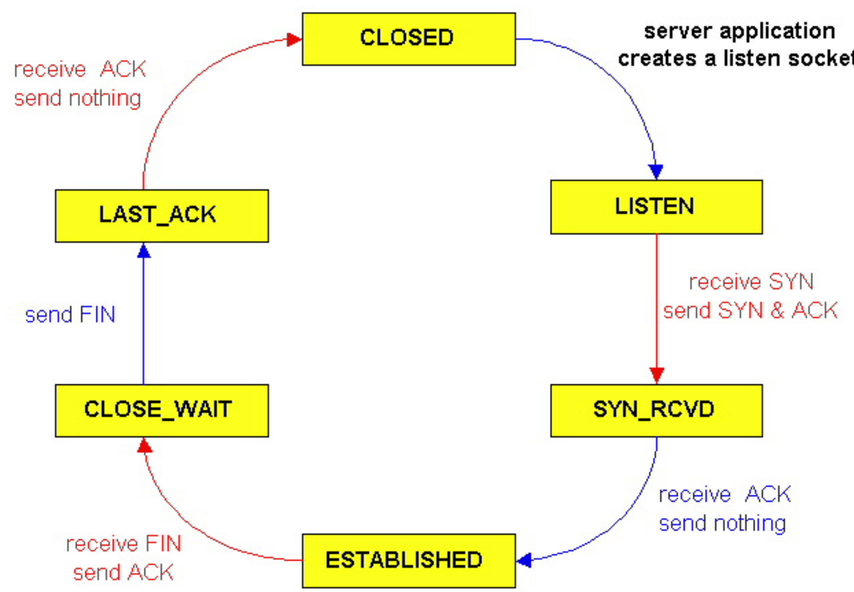


TCP Konexioaren kudeaketa (jarraipena)



TCP client lifecycle

TCP server lifecycle



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- 3.7 TCP-ren pilaketen kontrola (Bukaeran)
- 3.8 Bukaerako kontzeptuak

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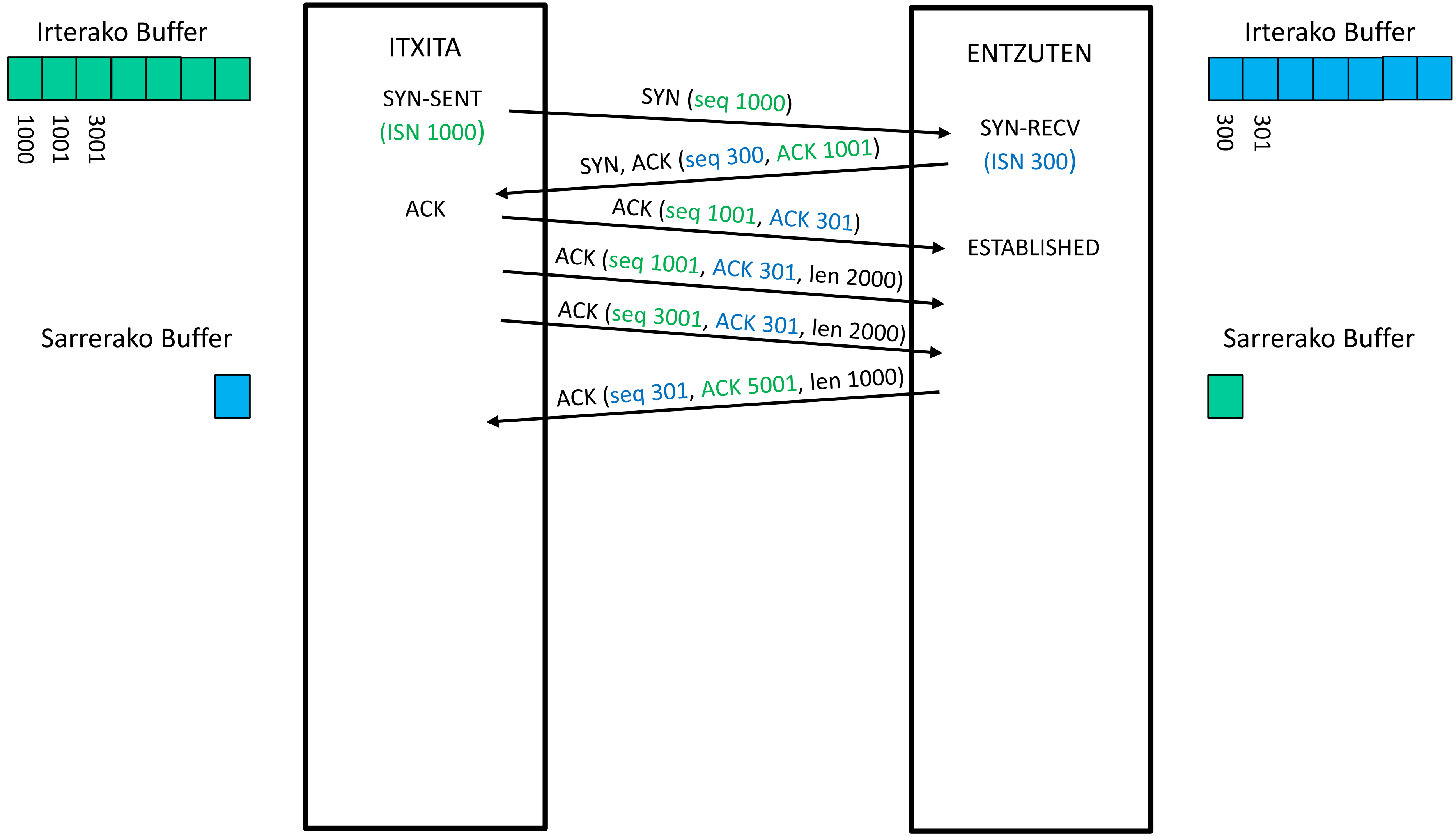
- 3.8 Bukaerako kontzeptuak

TCP/UDP Portuak

- **Portu zenbakia**: 2 byte (0-65535)
- Hiru esparru:
  - **0-1023: Well Know Ports**, Normalean erabiltzen diren zerbitsuetan erabiltzen dira
  - **1024-49151: Portu erregistratuak**, IANAk asignatzen ditu aplikazio desberdinetan erabiltzeko garatzailearen eskariz
  - **49152-65535: Pribatuak edo dinamikoak**, Bezeroetan erabilitak. Era dinamikoan (ausazkoan) esleitzen dira konexia sortzerakoan

Zerbitzua	Portua	TCP	UDP
ECHO	7	X	X
Day time	13	X	X
FTP	20-21	X	
SSH	22	X	
TelNet	23	X	
SMPT	24	X	
DNS	53	X	X
BOOTP	67		X
TFPT	69		X
HTTP	80	X	
POP3	110	X	
NTP	123		X
SNMP	189		X
LDAP	389	X	
HTTPS	443		X

TCP: Datu trukaketa



Chapter 3: Summary

- principles behind transport layer services:
  - multiplexing, demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- instantiation and implementation in the Internet
  - UDP
  - TCP

Next:

- leaving the network “edge” (application, transport layers)
- into the network “core”

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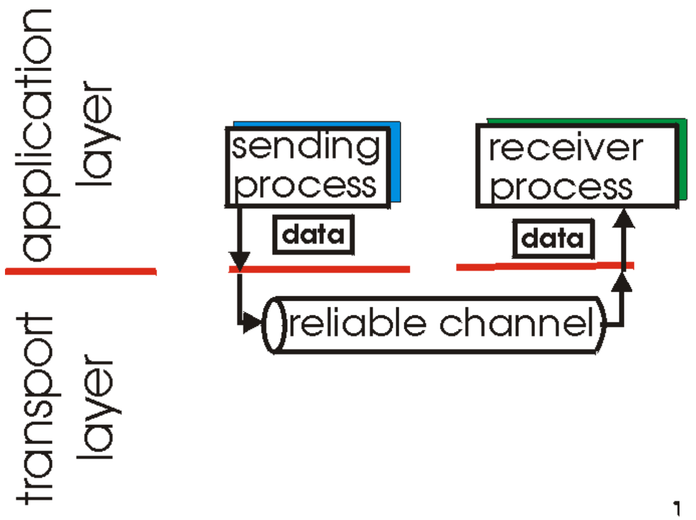
Informazio garraio fidagarriaren oinarriak

- Garrantzitsua aplikazioetan, garraioan, lotura geruzetan
- top-10 list of important networking topics!

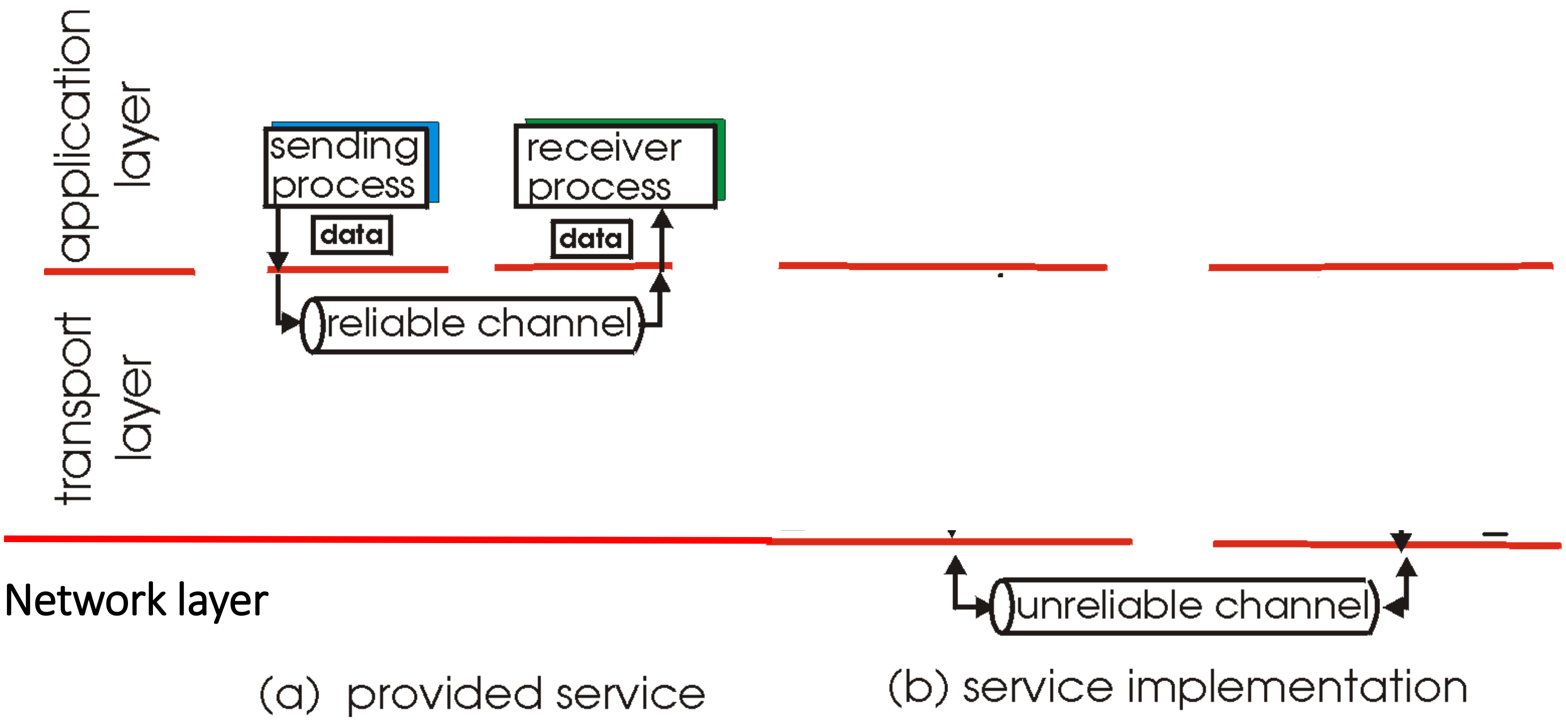


(a) provided service

- Kanal ez fidagarriaren ezaugarriek, informazio fidagarriaren garraio protokoloaren (rdt) konplexutasuna definitzen dute

Informazio garraio fidagarriaren oinarriak

- Garrantzitsua aplikazioetan, garraioan, lotura geruzetan
- top-10 list of important networking topics!



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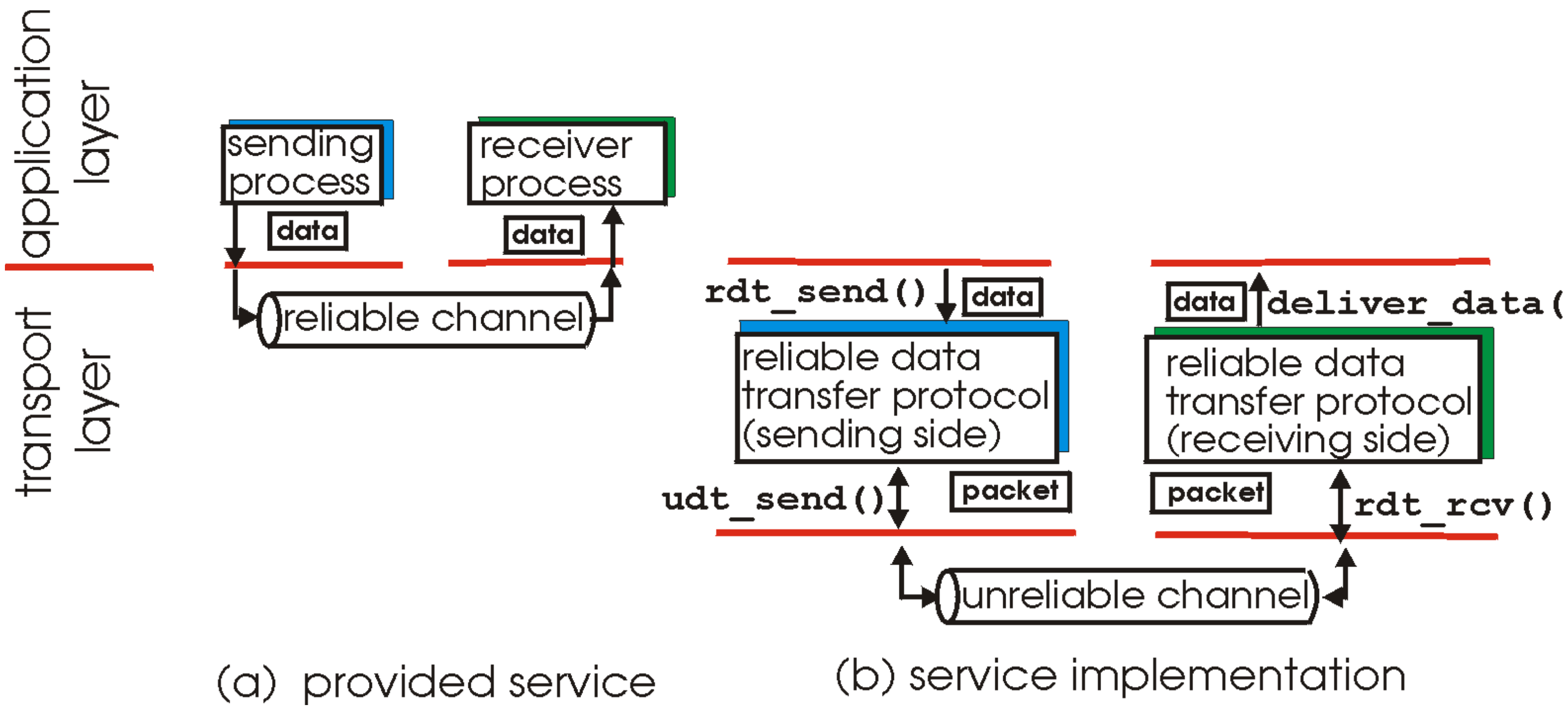
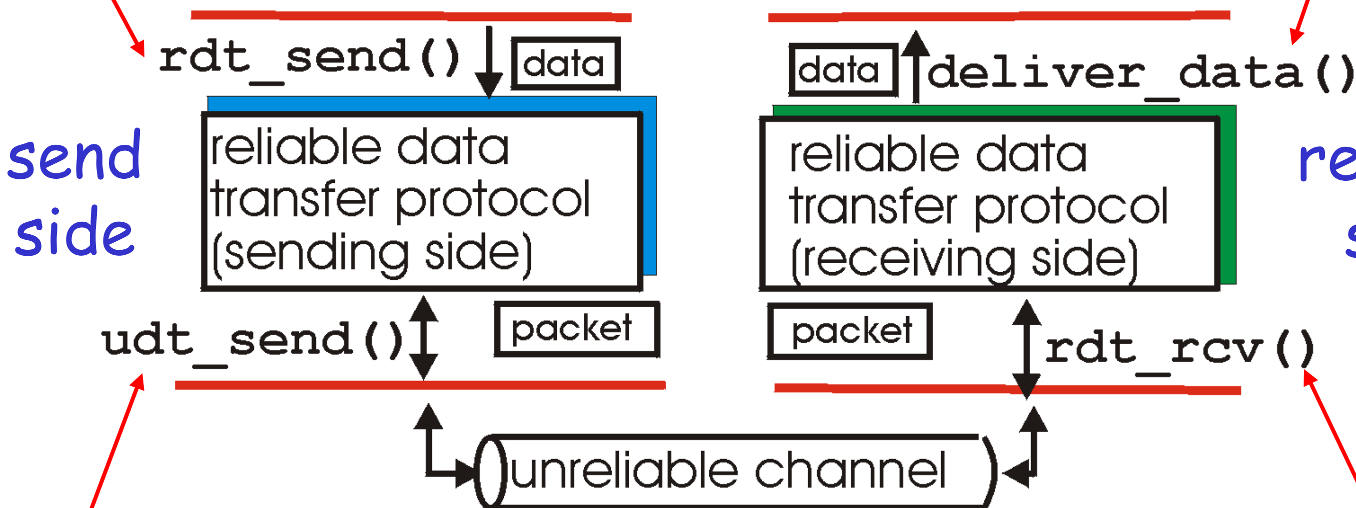


- Kanal ez fidagarriaren ezaugarriek, informazio fidagarriaren garraio protokoloaren (rdt) konplexutasuna definitzen dute

Informazio garraio fidagarria: hasiera

rdt_send(): called from above, (e.g., by app.). Passed data to deliver to receiver upper layer

deliver_data(): called by **rdt** to deliver data to upper



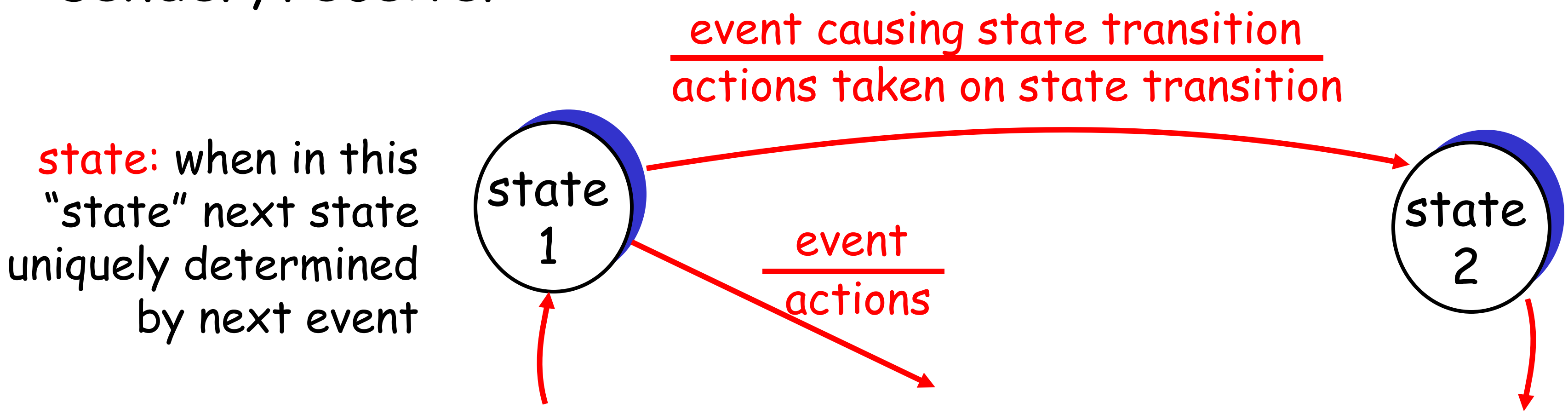
udt_send(): called by rdt, to transfer packet over unreliable channel to receiver

rdt_rcv(): called when packet arrives on rcv-side of channel

Informazio garraio fidagarria: hasiera

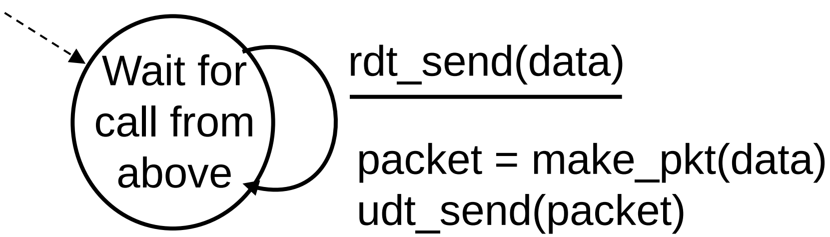
We'll:

- incrementally develop sender, receiver sides of reliable data transfer protocol (rdt)
- consider only unidirectional data transfer
  - but control info will flow on both directions!
- use finite state machines (FSM) to specify sender, receiver

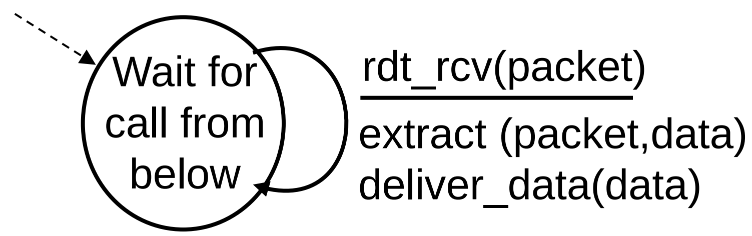


Rdt1.0: reliable transfer over a reliable channel

- underlying channel perfectly reliable
  - no bit errors
  - no loss of packets
- separate FSMs for sender, receiver:
  - sender sends data into underlying channel
  - receiver read data from underlying channel



sender

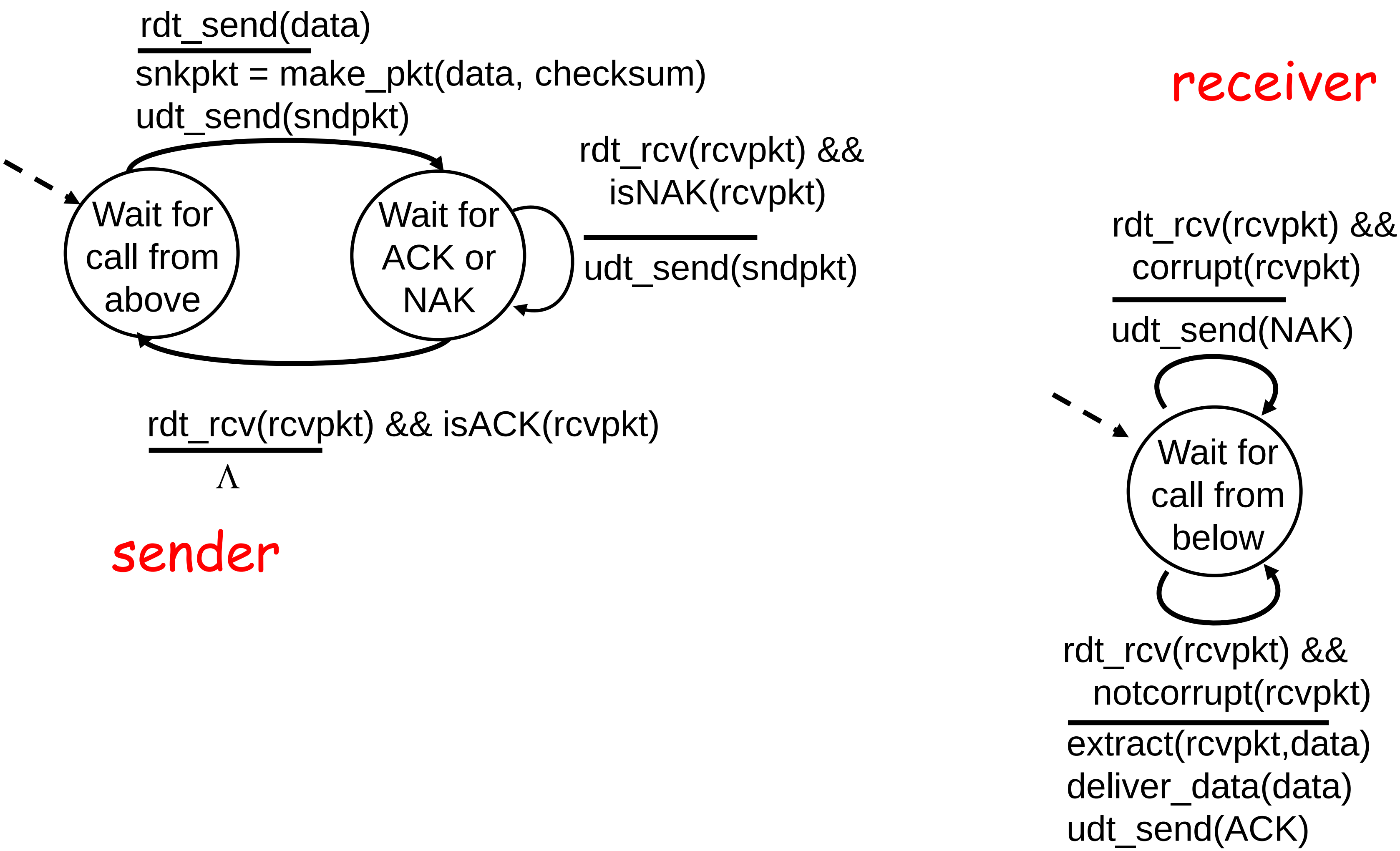


receiver

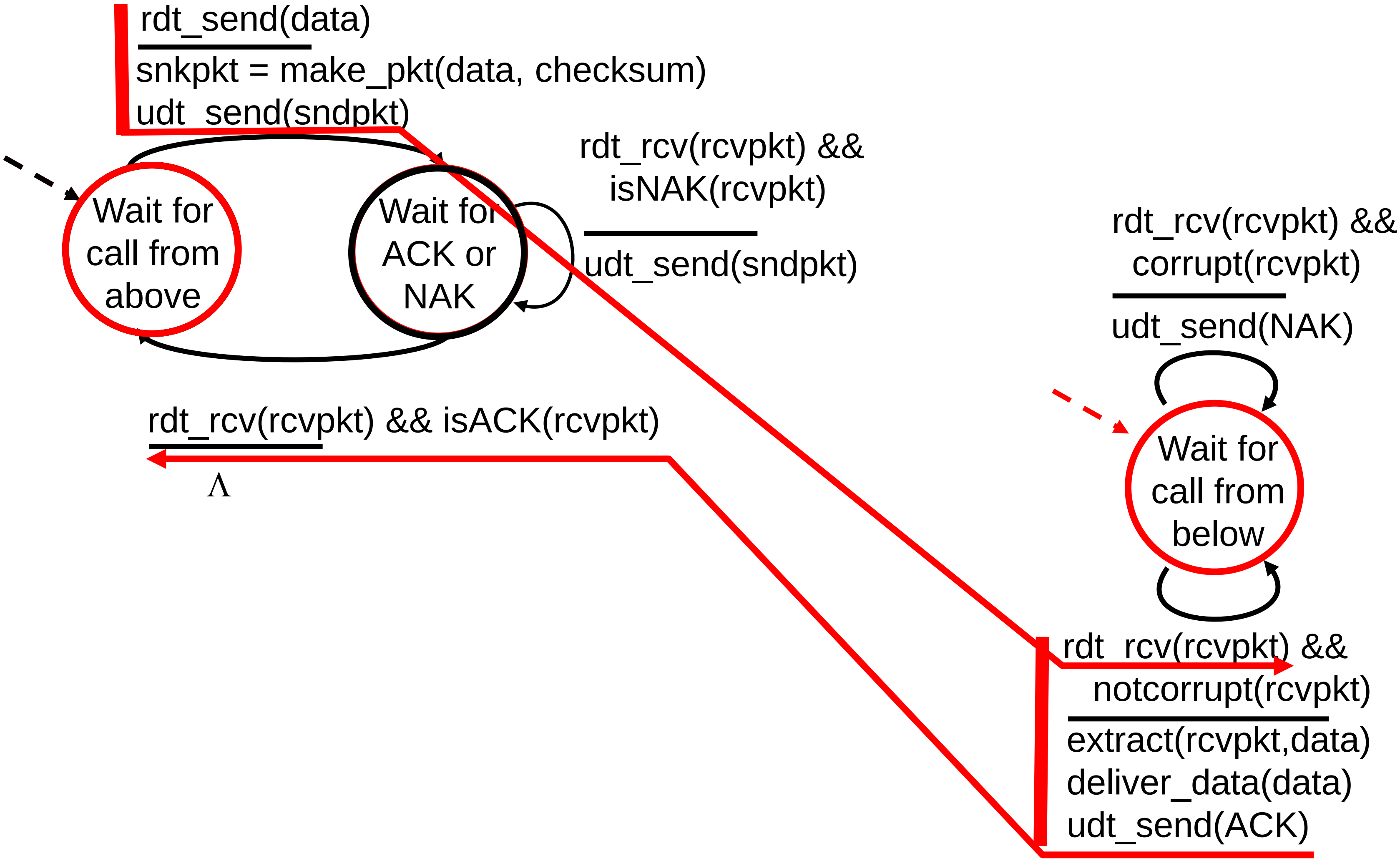
Rdt2.0: channel with bit errors

- underlying channel may flip bits in packet
  - checksum to detect bit errors
- *the* question: how to recover from errors:
  - *acknowledgements (ACKs):* receiver explicitly tells sender that pkt received OK
  - *negative acknowledgements (NAKs):* receiver explicitly tells sender that pkt had errors
  - sender retransmits pkt on receipt of NAK
- new mechanisms in **`rdt2.0`** (beyond **`rdt1.0`**):
  - error detection
  - receiver feedback: control msgs (ACK,NAK) rcvr->sender

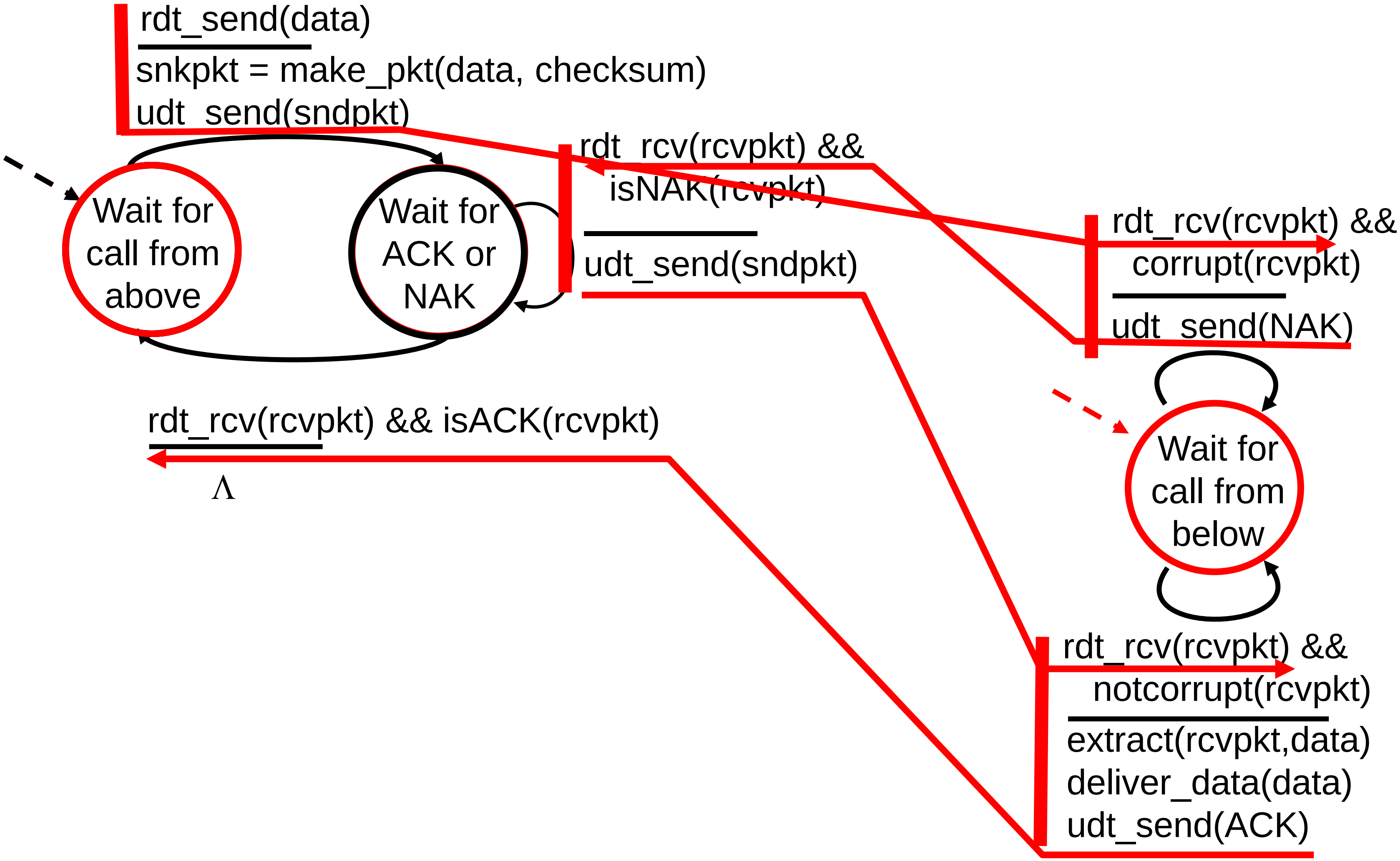
rdt2.0: FSM specification



rdt2.0: operation with no errors



rdt2.0: error scenario



rdt2.0 has a fatal flaw!

What happens if ACK/NAK corrupted?

- sender doesn't know what happened at receiver!
- can't just retransmit: possible duplicate

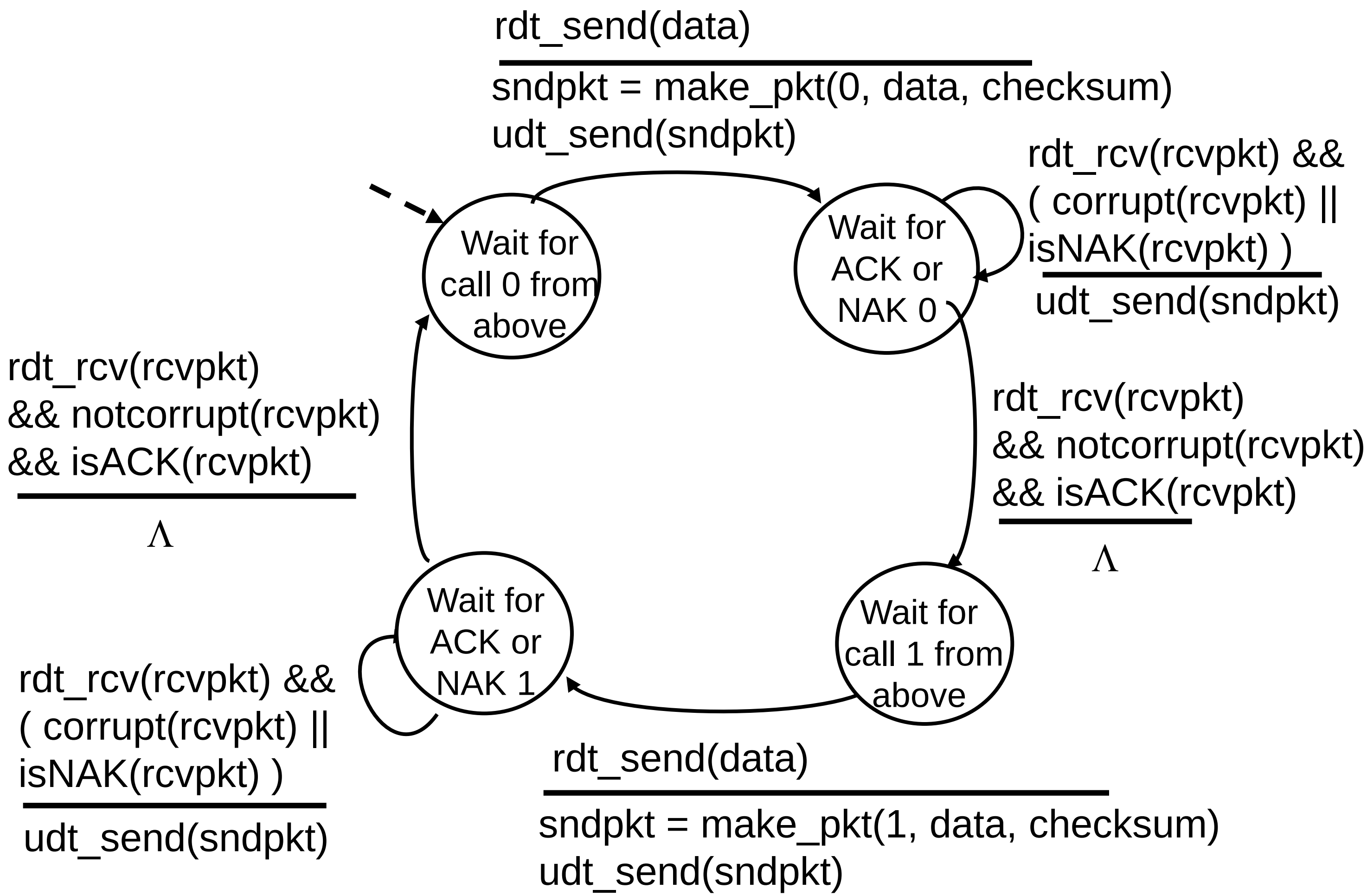
Handling duplicates:

- sender retransmits current pkt if ACK/NAK garbled
- sender adds *sequence number* to each pkt
- receiver discards (doesn't deliver up) duplicate pkt

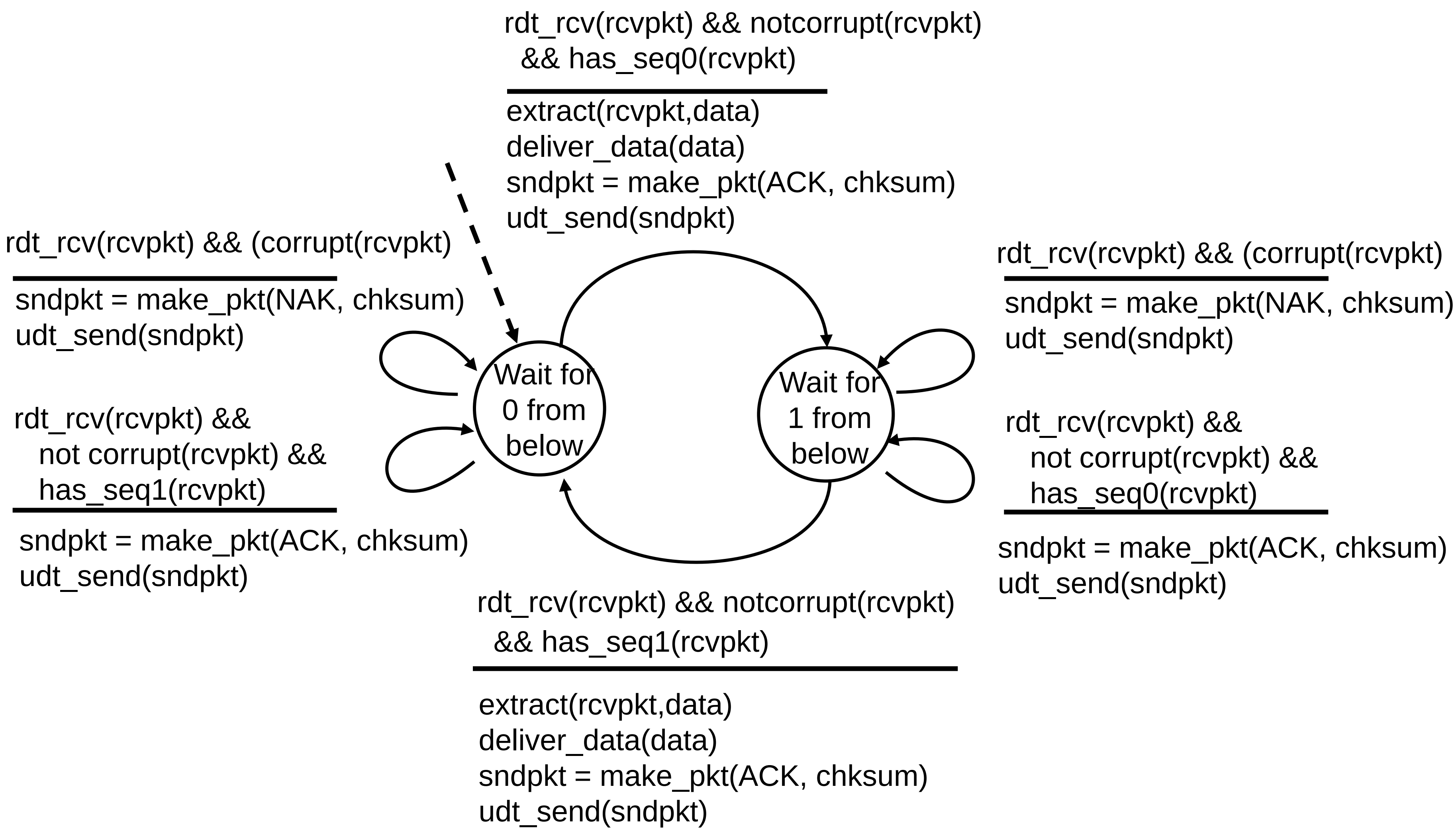
stop and wait

Sender sends one packet, then waits for receiver response

rdt2.1: sender, handles garbled ACK/NAKs



rdt2.1: receiver, handles garbled ACK/NAKs



rdt2.1: discussion

Sender:

- seq # added to pkt
- two seq. #'s (0,1) will suffice. Why?
- must check if received ACK/NAK corrupted
- twice as many states
  - state must "remember" whether "current" pkt has 0 or 1 seq. #

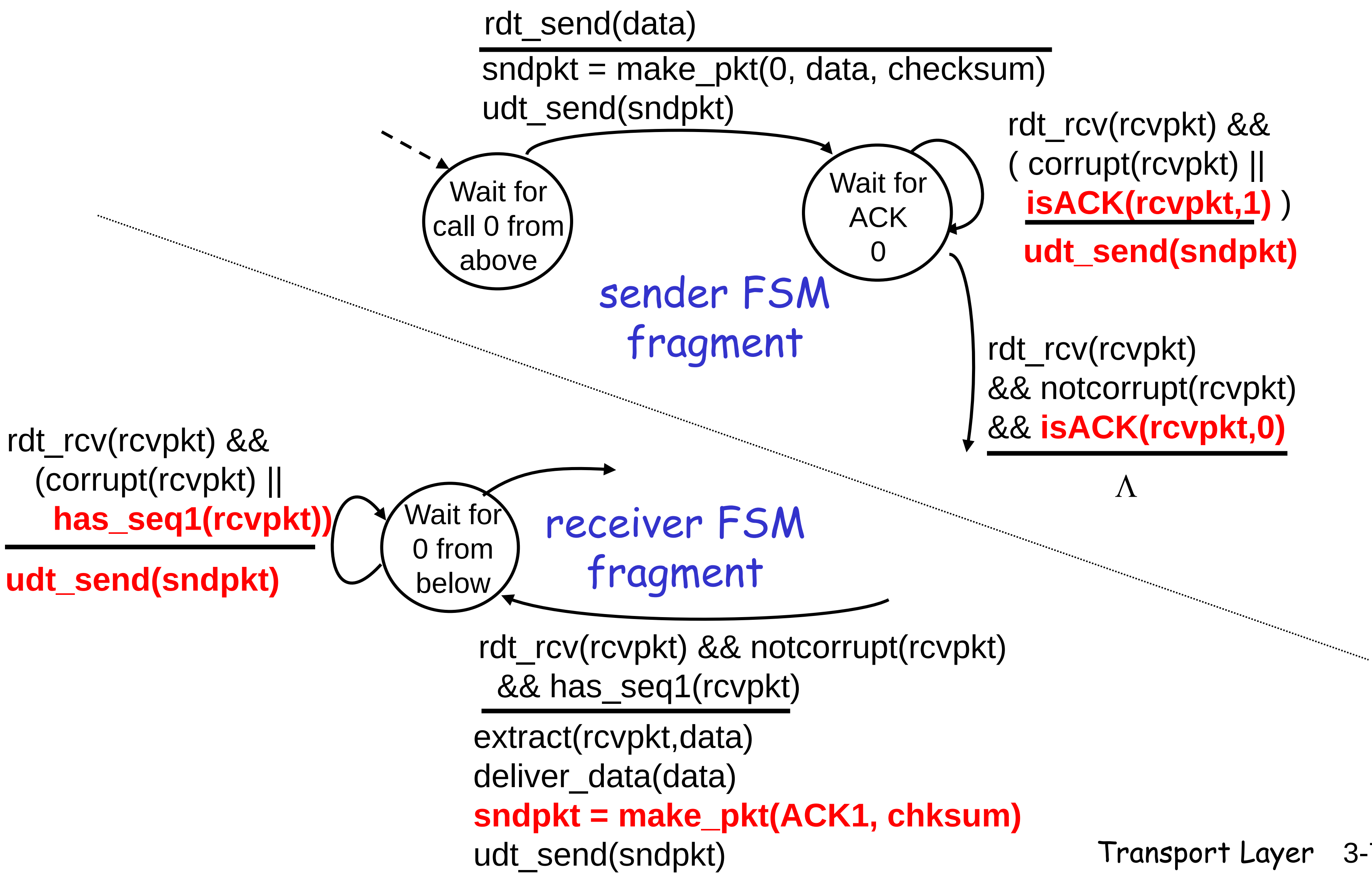
Receiver:

- must check if received packet is duplicate
  - state indicates whether 0 or 1 is expected pkt seq #
- note: receiver can *not* know if its last ACK/NAK received OK at sender

rdt2.2: a NAK-free protocol

- same functionality as rdt2.1, using ACKs only
- instead of NAK, receiver sends ACK for last pkt received OK
  - receiver must *explicitly* include seq # of pkt being ACKed
- duplicate ACK at sender results in same action as NAK: *retransmit current pkt*

rdt2.2: sender, receiver fragments



rdt3.0: channels with errors *and* loss

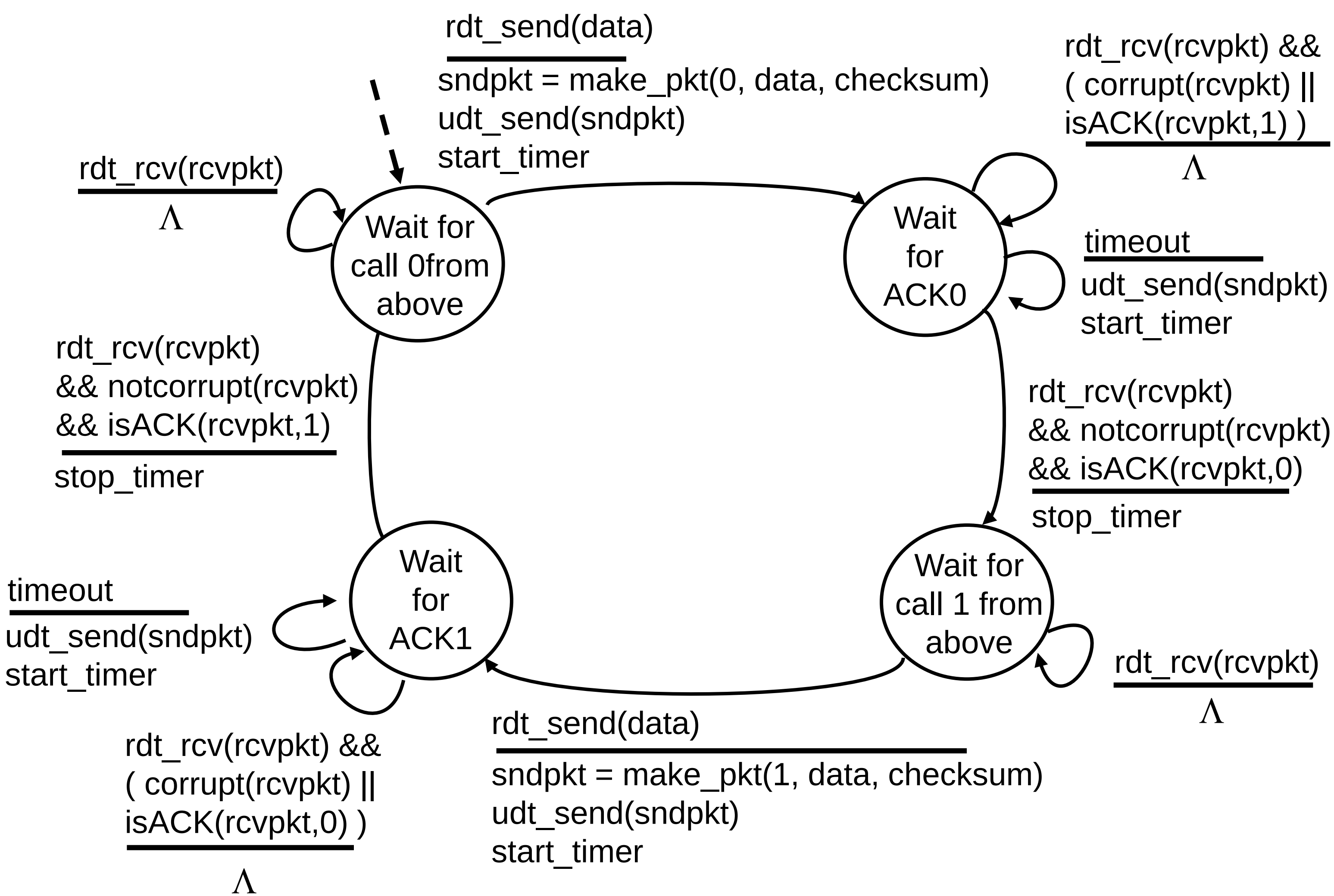
New assumption: underlying channel can also lose packets (data or ACKs)

- checksum, seq. #, ACKs, retransmissions will be of help, but not enough

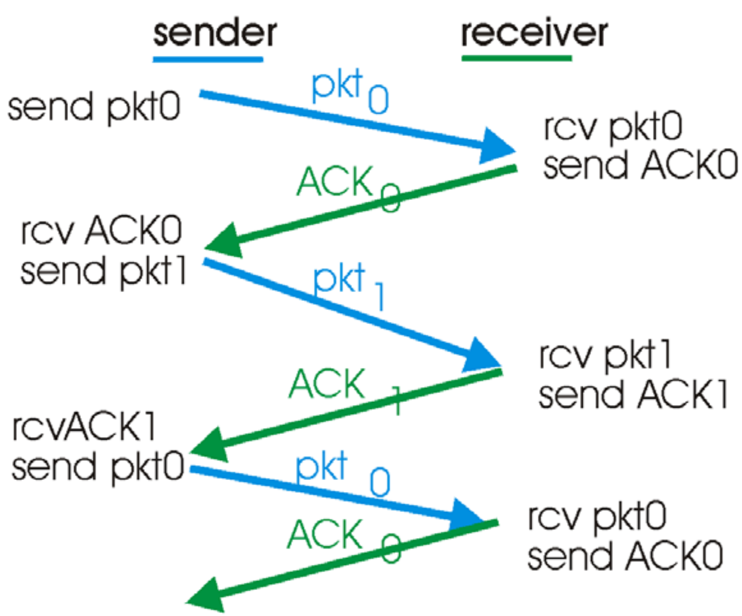
Approach: sender waits "reasonable" amount of time for ACK

- retransmits if no ACK received in this time
- if pkt (or ACK) just delayed (not lost):
  - retransmission will be duplicate, but use of seq. #'s already handles this
  - receiver must specify seq # of pkt being ACKed
- requires countdown timer

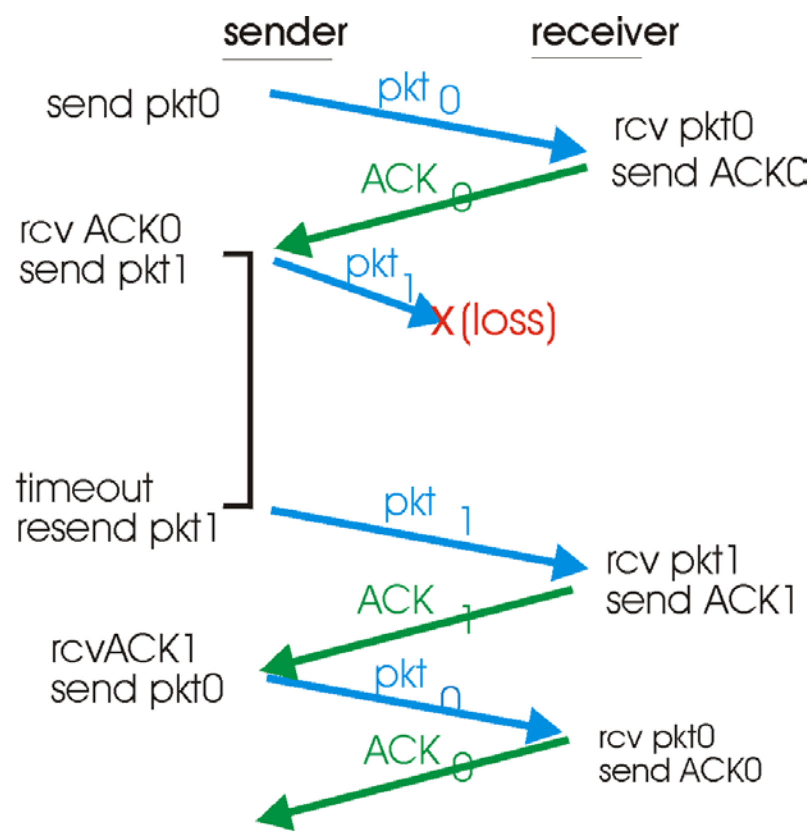
rdt3.0 sender



rdt3.0 in action

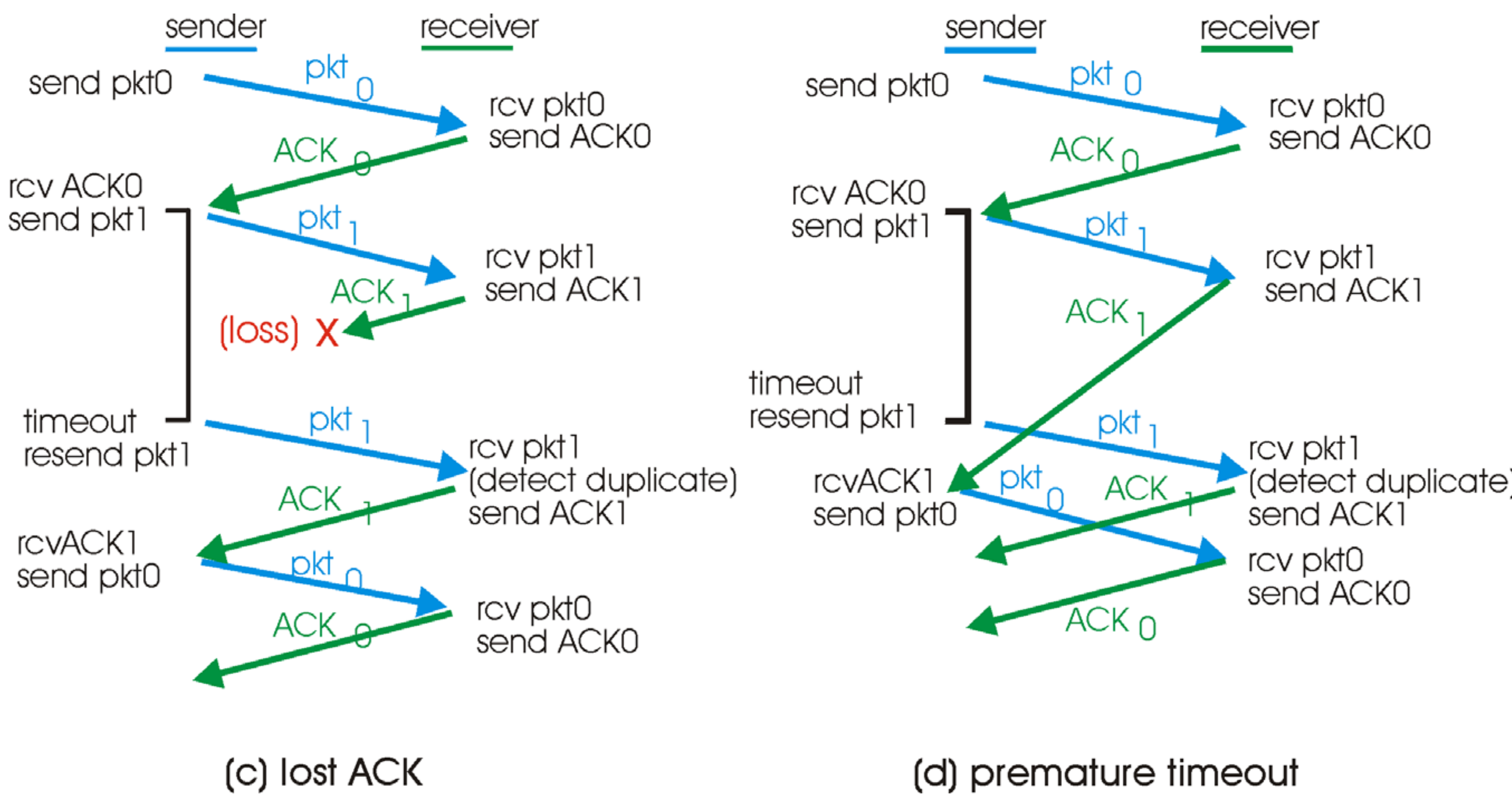


(a) operation with no loss



(b) lost packet

rdt3.0 in action



Performance of rdt3.0

- rdt3.0 works, but performance stinks
- ex: 1 Gbps link, 15 ms prop. delay, 8000 bit packet:

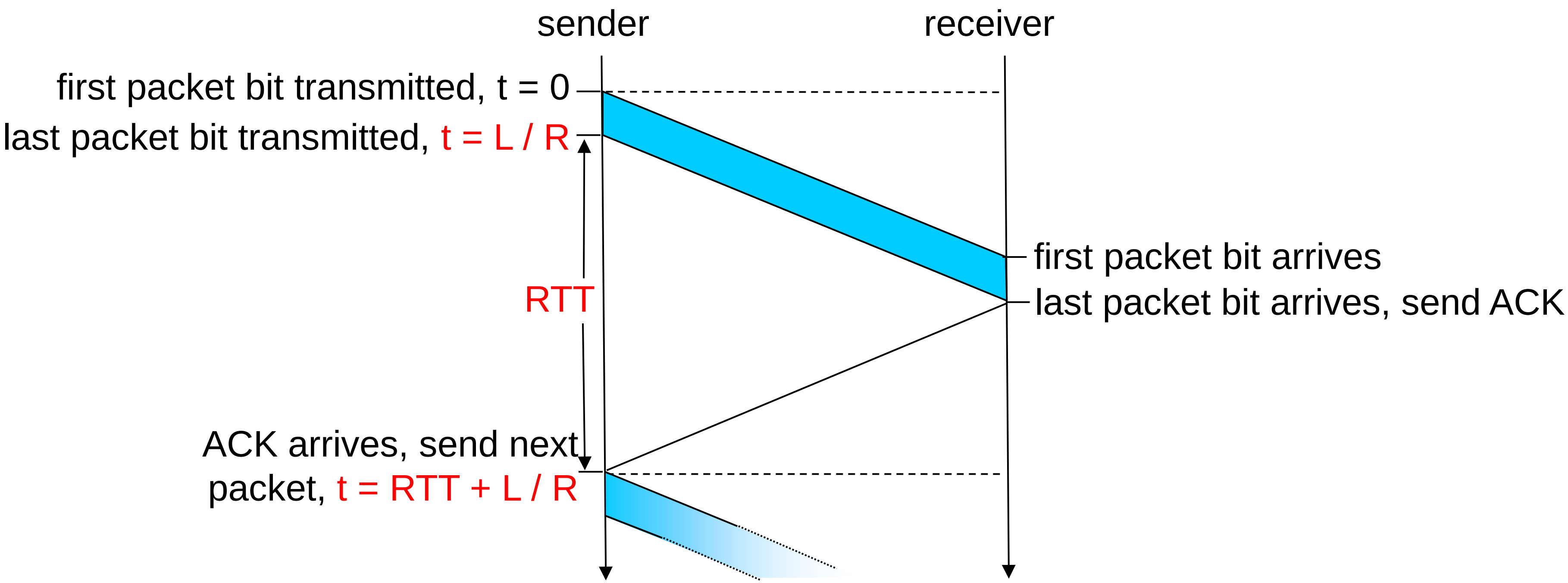
$$d_{trans} = \frac{L}{R} = \frac{8000\text{bits}}{10^9\,\text{bps}} = 8\,\text{microseconds}$$

  - U_{sender}: utilization – fraction of time sender busy sending

$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

  - 1KB pkt every 30 msec -> 33kB/sec thruput over 1 Gbps link
  - network protocol limits use of physical resources!

rdt3.0: stop-and-wait operation

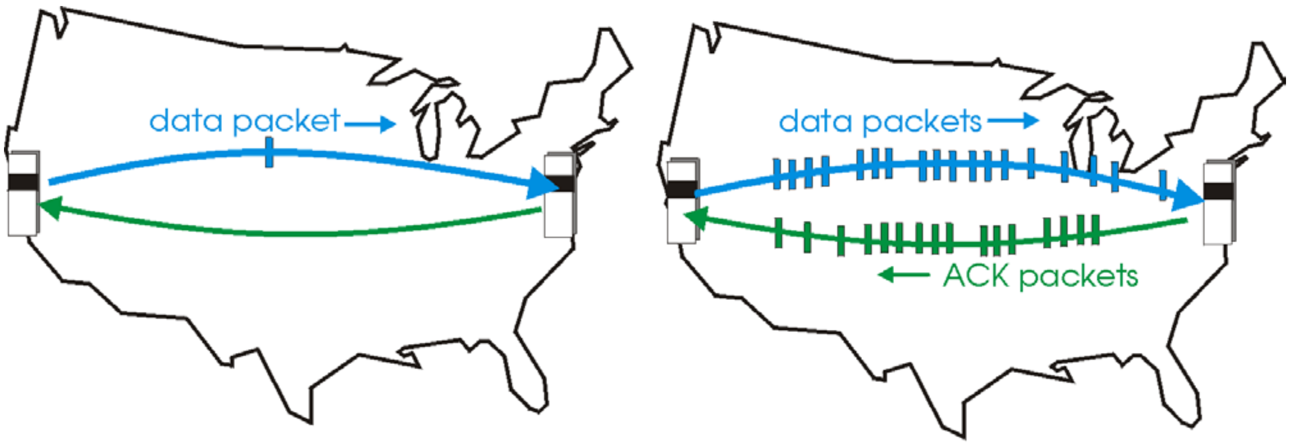


$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

Pipelined protocols

Pipelining: sender allows multiple, "in-flight", yet-to-be-acknowledged pkts

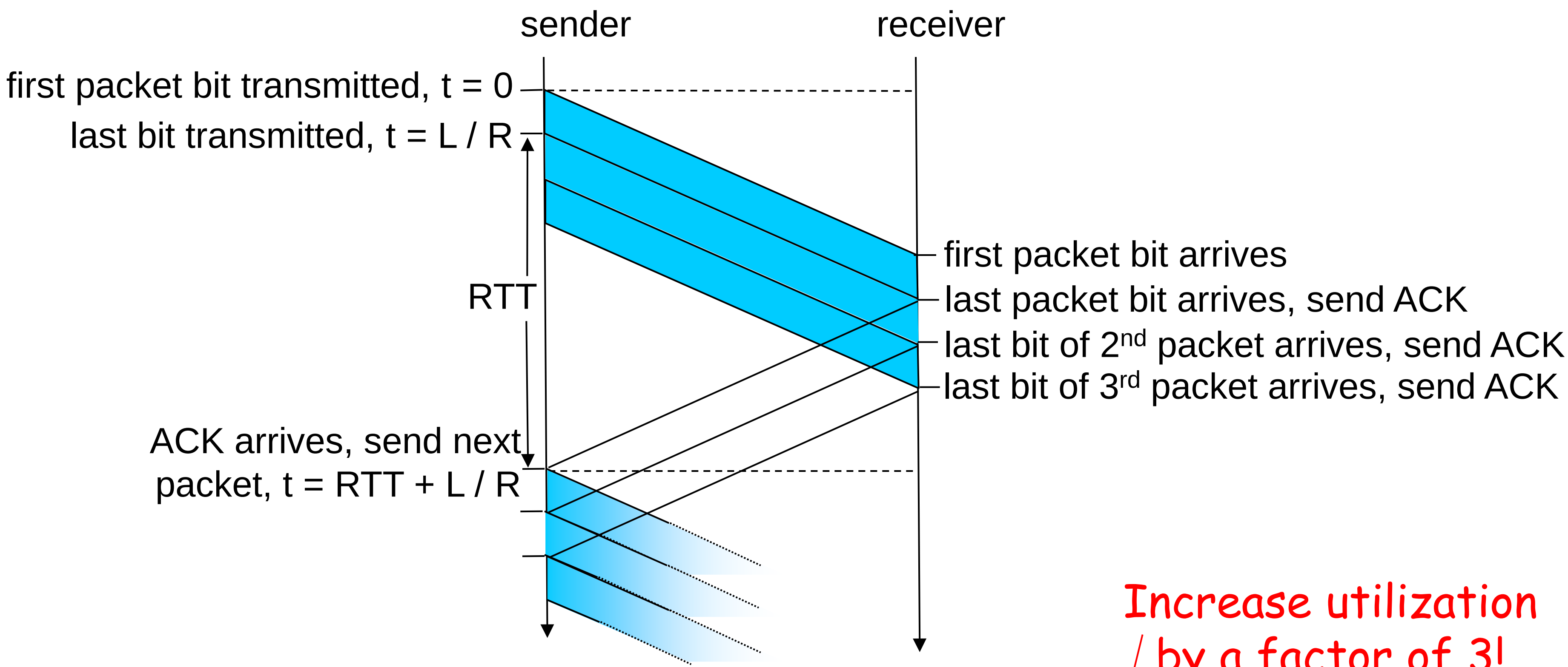
- range of sequence numbers must be increased
- buffering at sender and/or receiver



(a) a stop-and-wait protocol in operation (b) a pipelined protocol in operation

- Two generic forms of pipelined protocols: *go-Back-N, selective repeat*

Pipelining: increased utilization



Increase utilization by a factor of 3!

$$U_{sender} = \frac{3 * L / R}{RTT + L / R} = \frac{.024}{30.008} = 0.0008$$

Pipelining Protocols

Go-back-N: big picture:

- Sender can have up to N unacked packets in pipeline
- Rcvr only sends cumulative acks
  - Doesn't ack packet if there's a gap
- Sender has timer for oldest unacked packet
  - If timer expires, retransmit all unacked packets

Selective Repeat: big pic

- Sender can have up to N unacked packets in pipeline
- Rcvr acks individual packets
- Sender maintains timer for each unacked packet
  - When timer expires, retransmit only unack packet

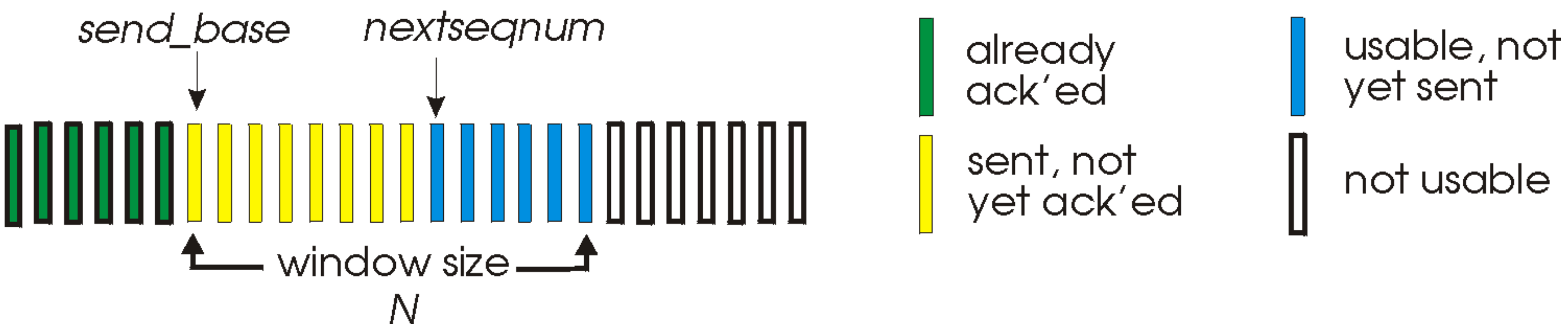
Selective repeat: big picture

- Sender can have up to N unacked packets in pipeline
- Rcvr acks individual packets
- Sender maintains timer for each unacked packet
  - When timer expires, retransmit only unack packet

Go-Back-N

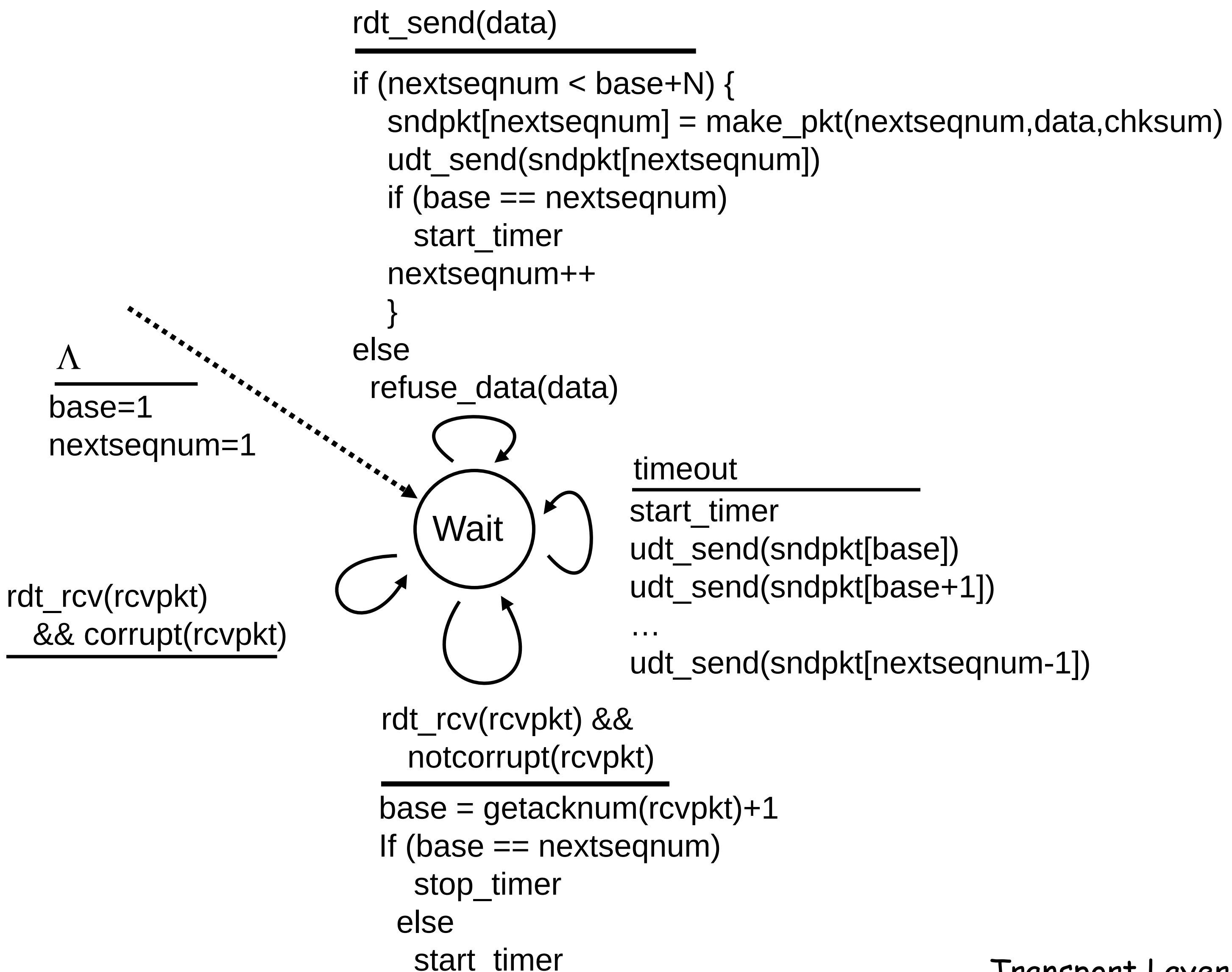
Sender:

- k-bit seq # in pkt header
- "window" of up to N, consecutive unack'ed pkts allowed

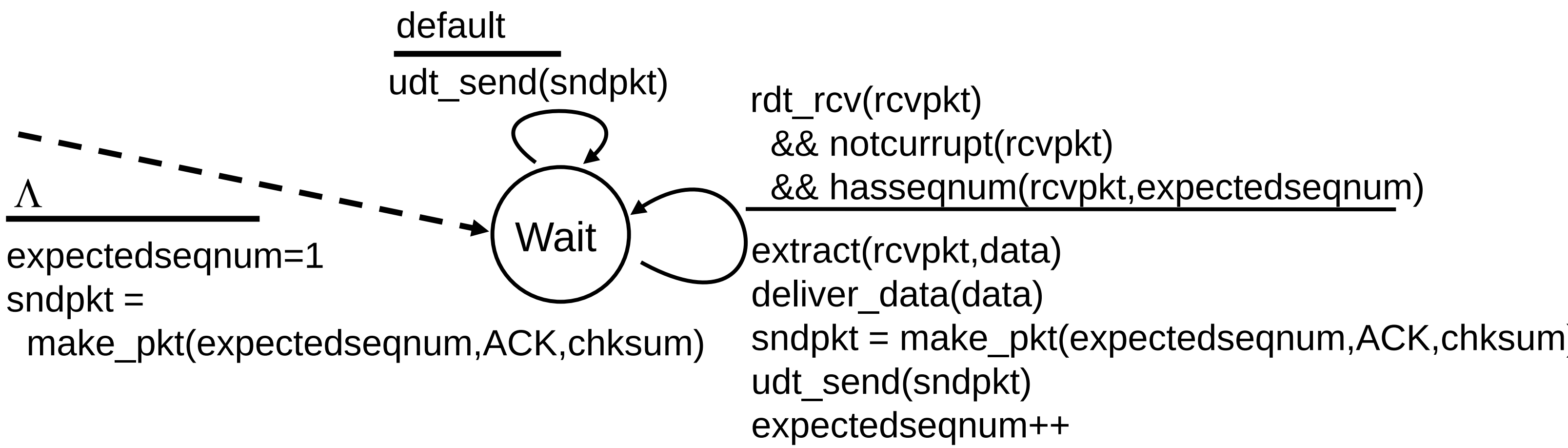


- ACK(n): ACKs all pkts up to, including seq # n - "cumulative ACK"
  - may receive duplicate ACKs (see receiver)
- timer for each in-flight pkt
- *timeout(n):* retransmit pkt n and all higher seq # pkts in window

GBN: sender extended FSM



GBN: receiver extended FSM

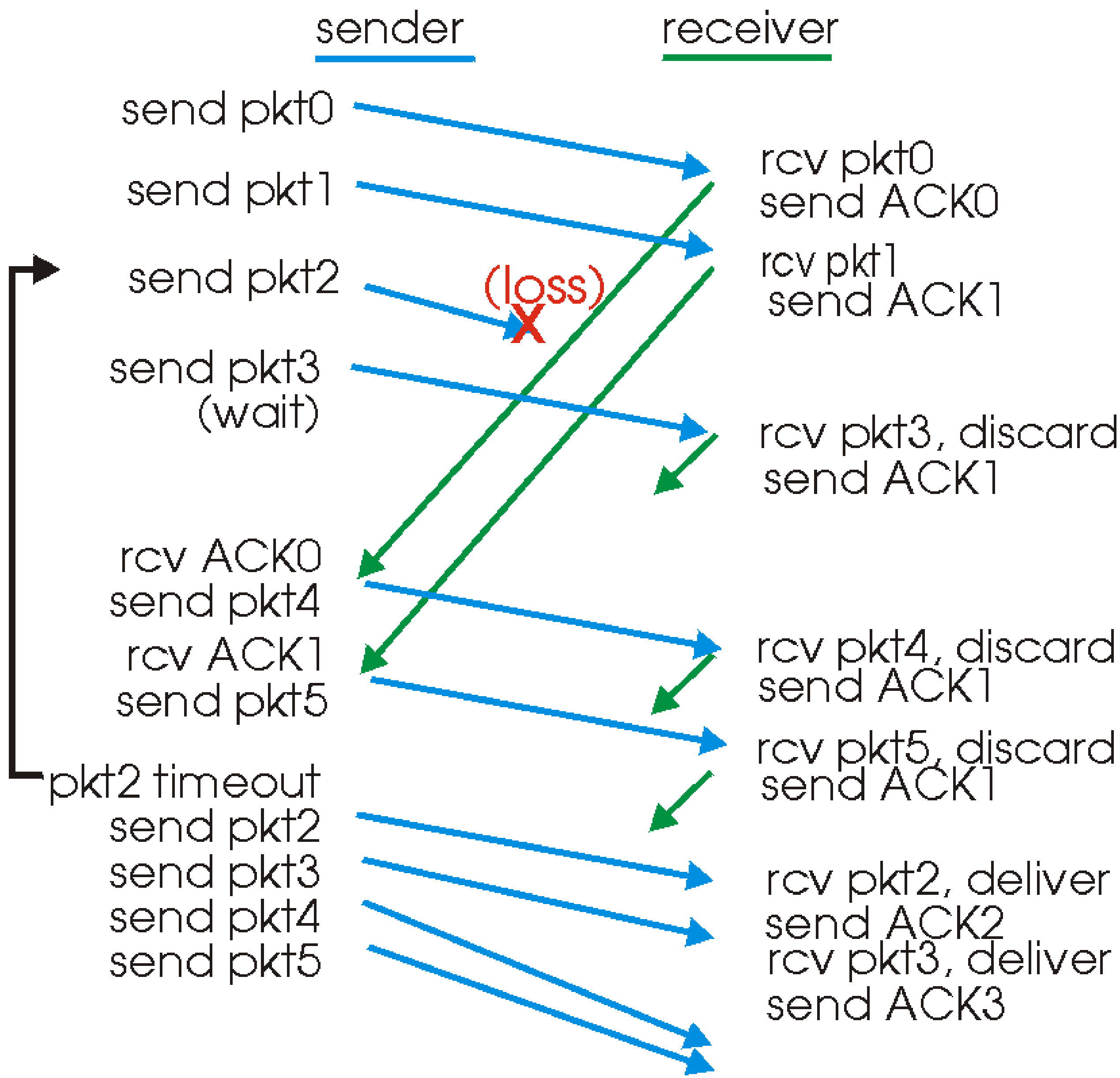


ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq #

- may generate duplicate ACKs
- need only remember **expectedseqnum**

- out-of-order pkt:
  - discard (don't buffer) -> no receiver buffering!
  - Re-ACK pkt with highest in-order seq #

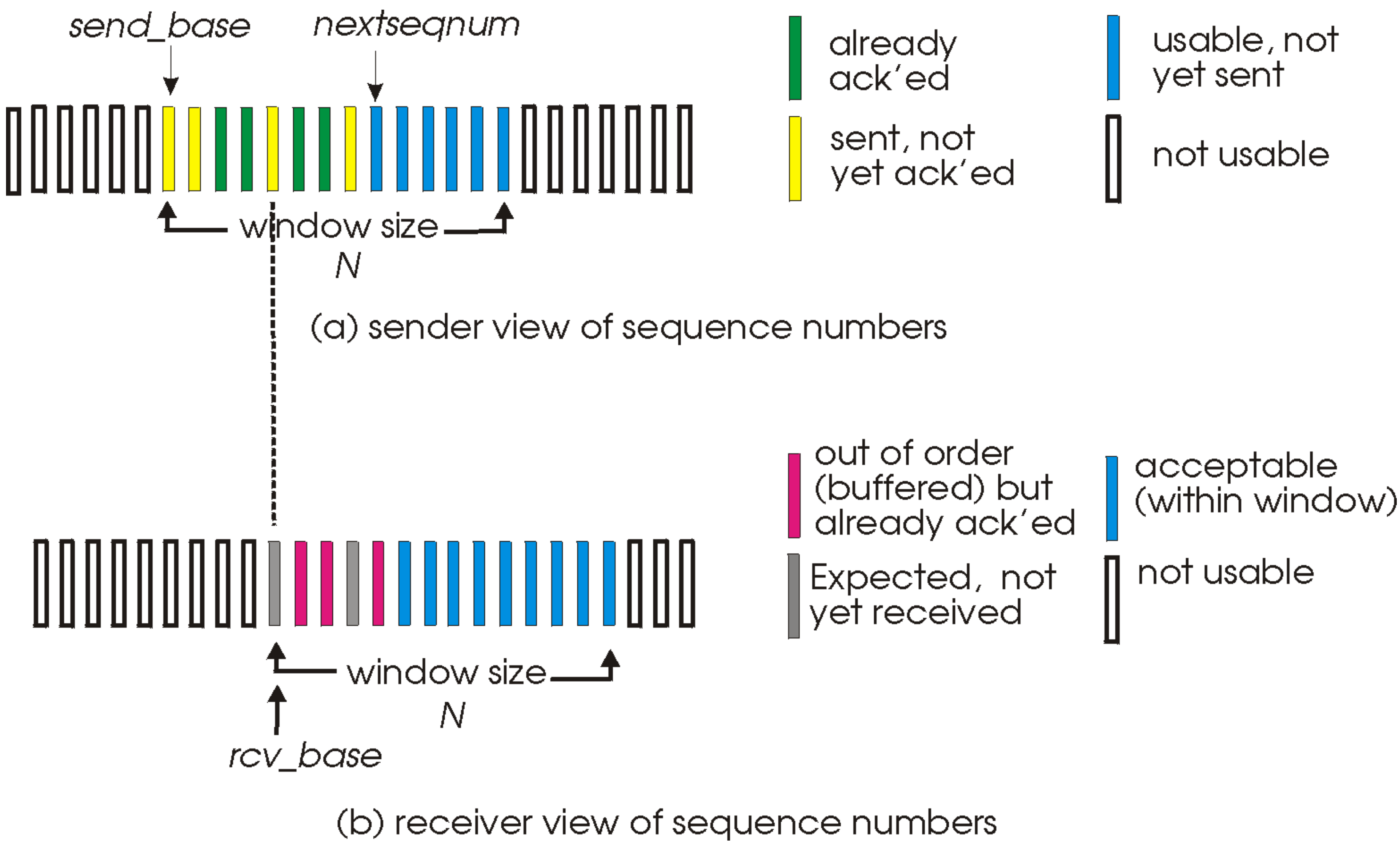
GBN in action



Selective Repeat

- receiver *individually* acknowledges all correctly received pkts
  - buffers pkts, as needed, for eventual in-order delivery to upper layer
- sender only resends pkts for which ACK not received
  - sender timer for each unACKed pkt
- sender window
  - N consecutive seq #'s
  - again limits seq #s of sent, unACKed pkts

Selective repeat: sender, receiver windows



Selective repeat

sender

data from above :

- if next available seq # in window, send pkt

timeout(n):

- resend pkt n, restart timer

ACK(n) in [sendbase,sendbase+N]:

- mark pkt n as received
- if n smallest unACKed pkt, advance window base to next unACKed seq #

receiver

pkt n in [rcvbase, rcvbase+N-1]

- send ACK(n)
- out-of-order: buffer
- in-order: deliver (also deliver buffered, in-order pkts), advance window to next not-yet-received pkt

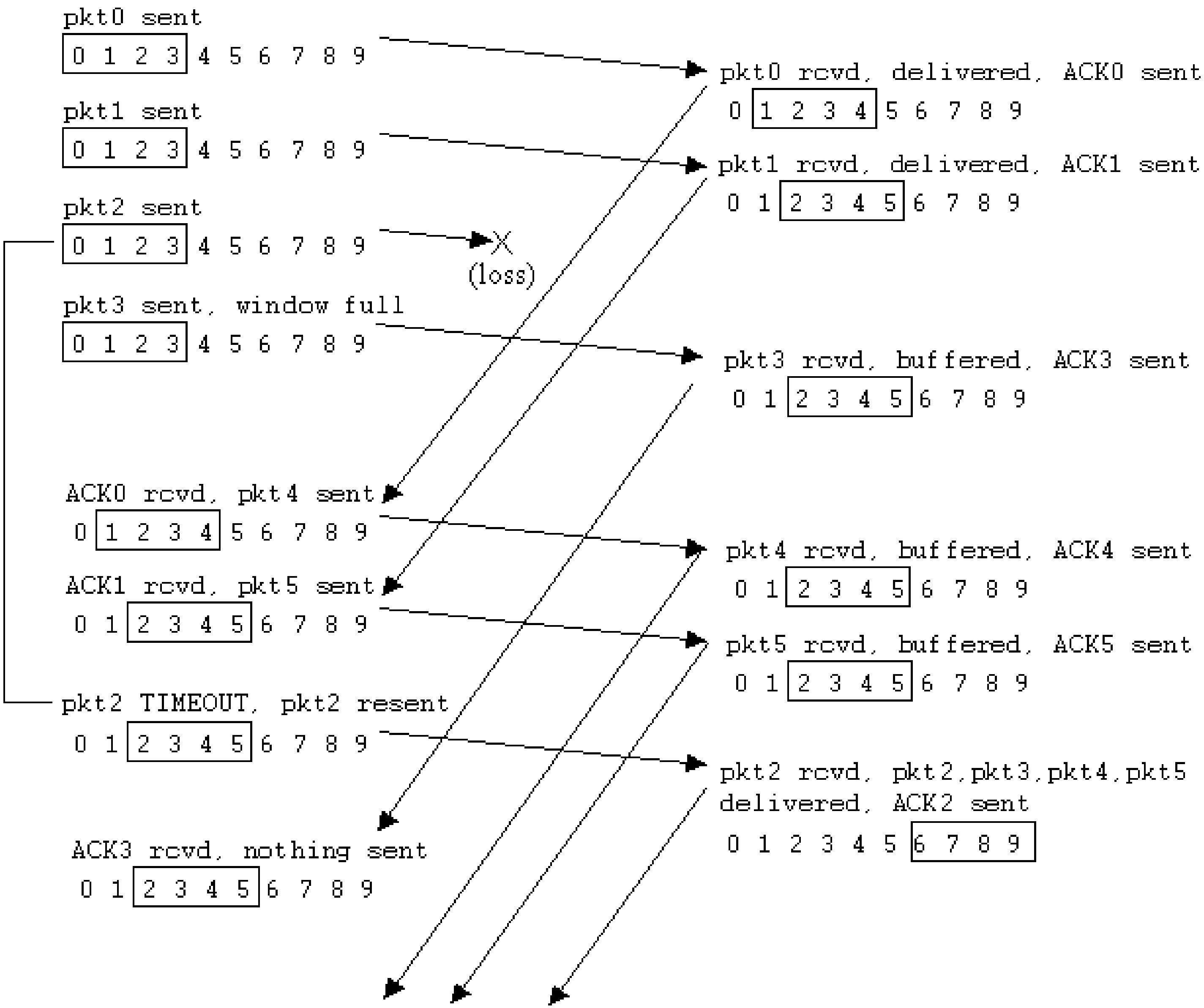
pkt n in [rcvbase-N,rcvbase-1]

- ACK(n)

otherwise:

- ignore

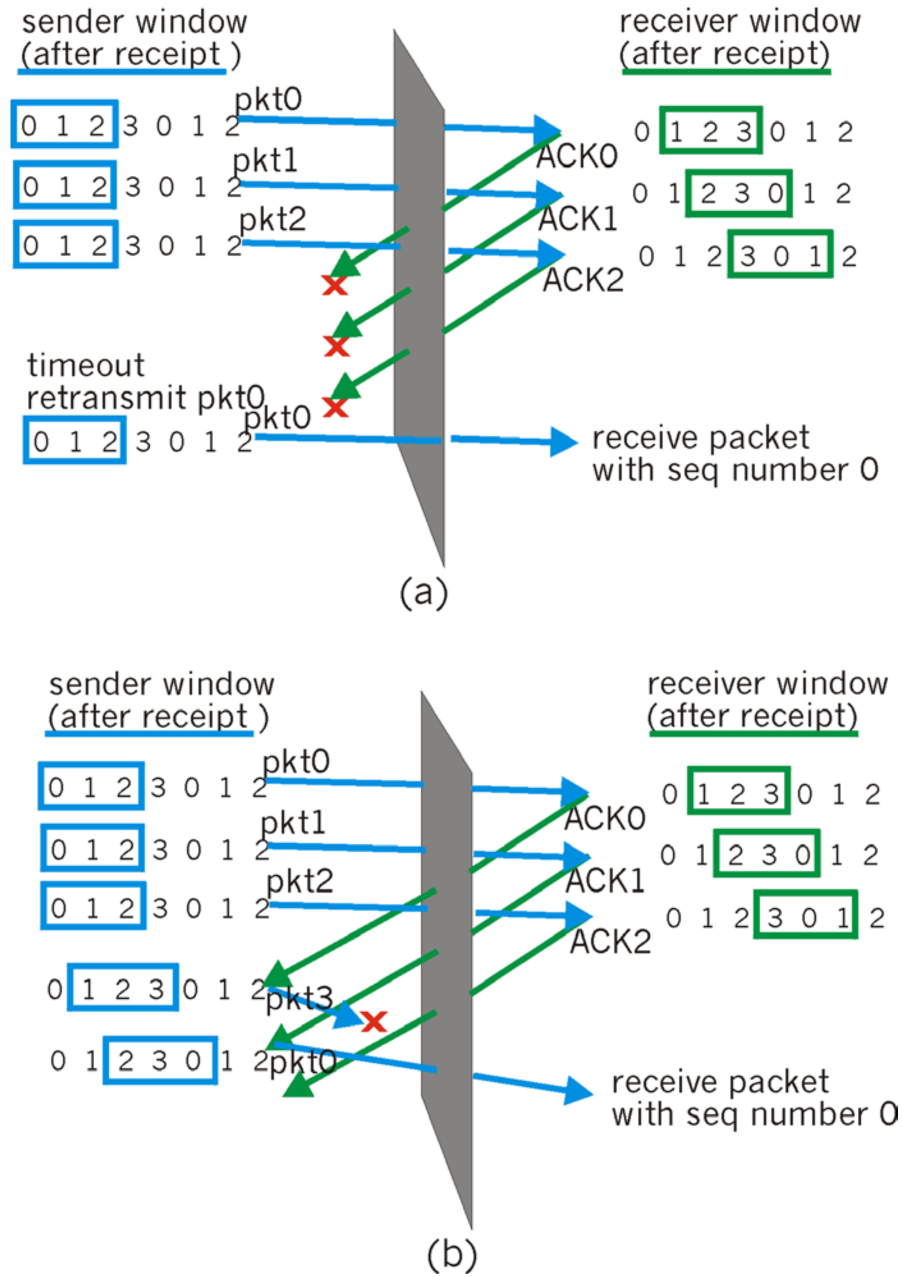
Selective repeat in action



Selective repeat: dilemma

Example:

- seq #'s: 0, 1, 2, 3
- window size=3



- receiver sees no difference in two scenarios!
- incorrectly passes duplicate data as new in (a)

Q: what relationship between seq # size and window size?

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

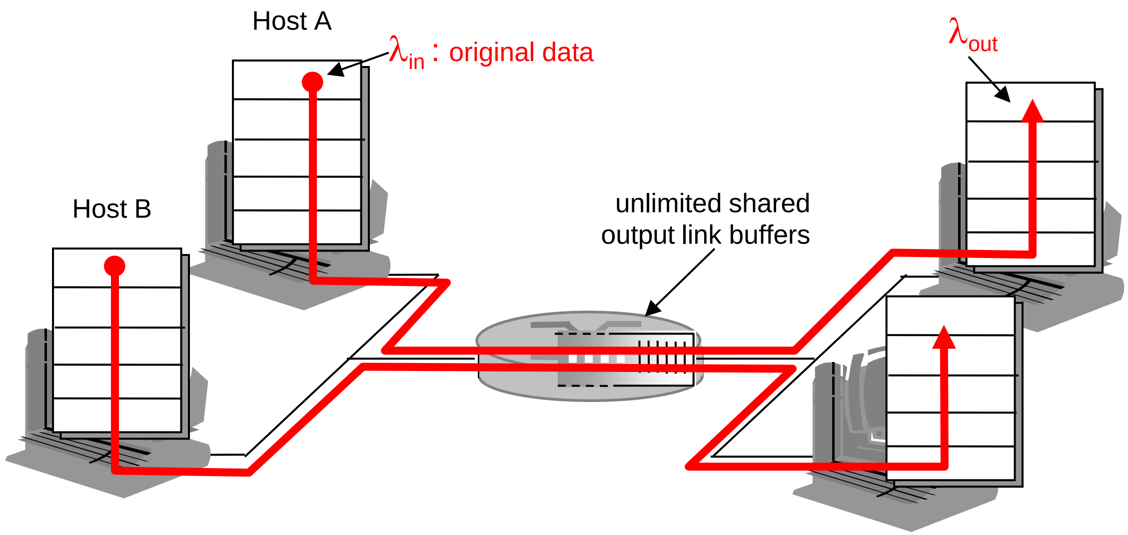
Principles of Congestion Control

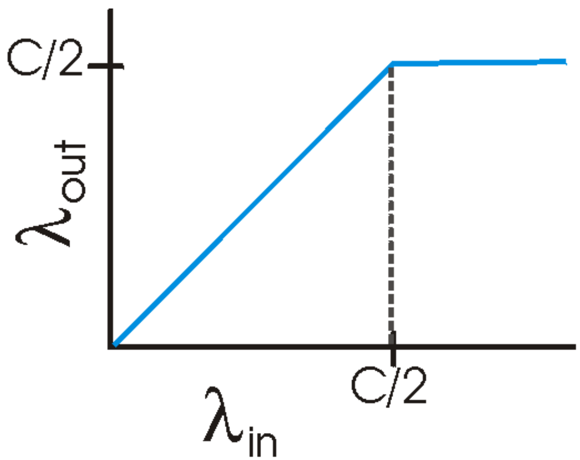
Congestion:

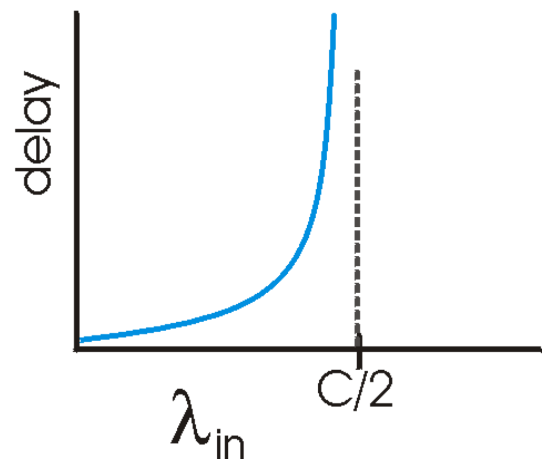
- informally: "too many sources sending too much data too fast for *network* to handle"
- different from flow control!
- manifestations:
  - lost packets (buffer overflow at routers)
  - long delays (queueing in router buffers)
- a top-10 problem!

Causes/costs of congestion: scenario 1

- two senders, two receivers
- one router, infinite buffers
- no retransmission



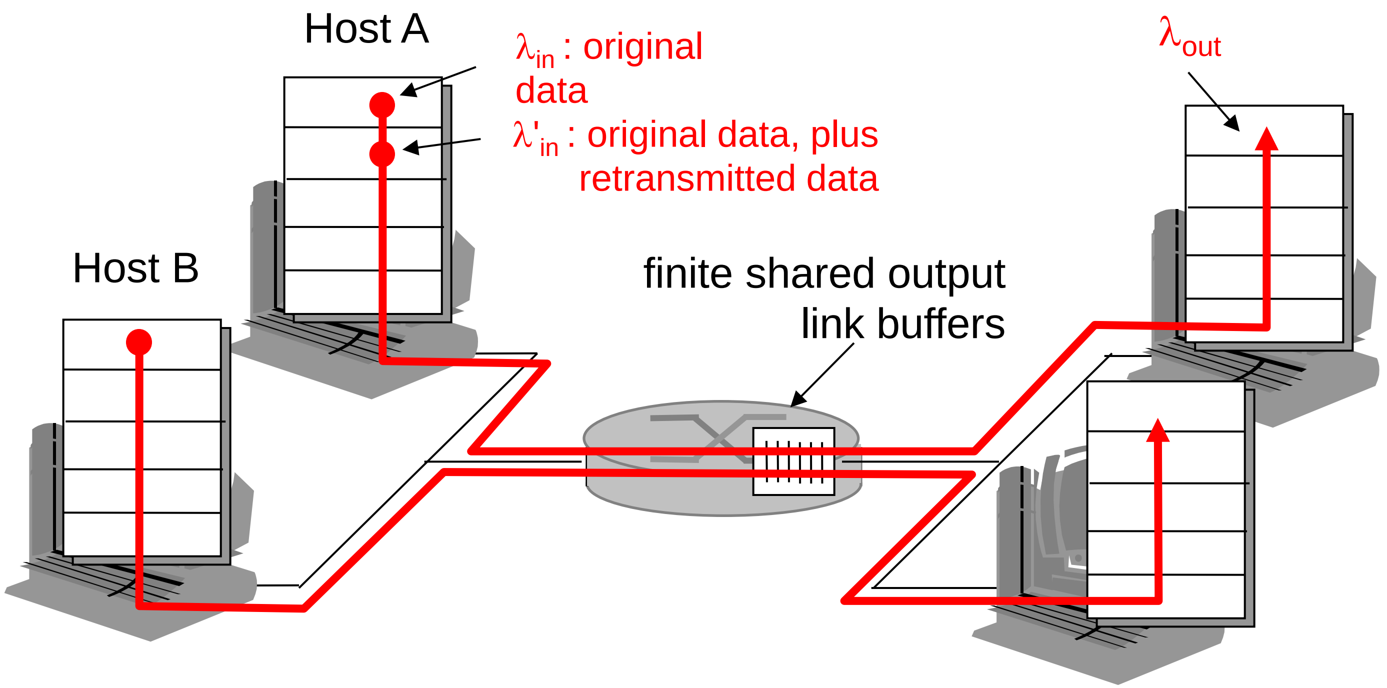




- large delays when congested
- maximum achievable throughput

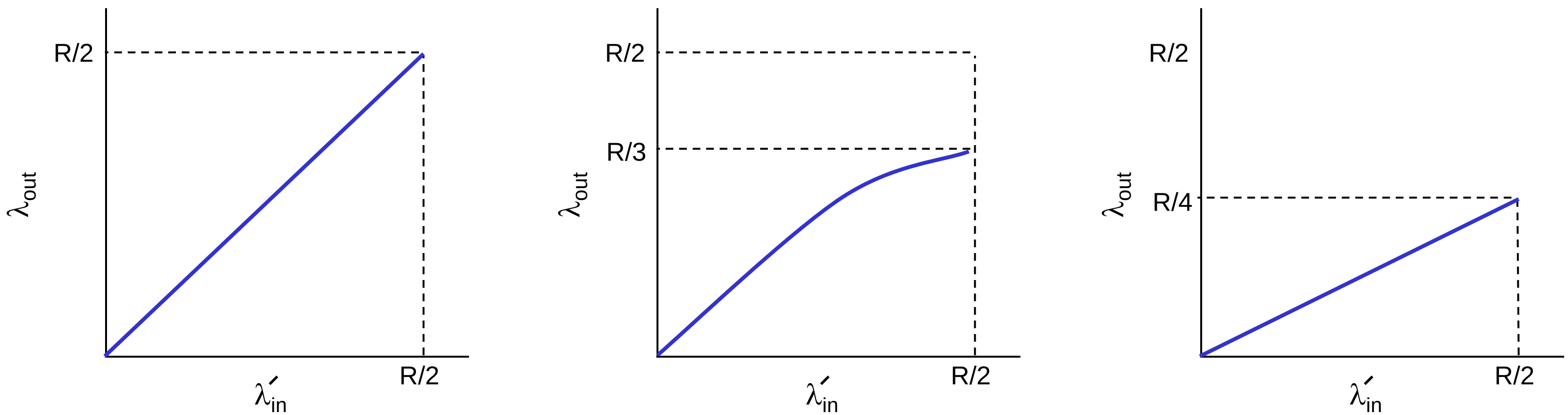
Causes/costs of congestion: scenario 2

- one router, *finite* buffers
- sender retransmission of lost packet



Causes/costs of congestion: scenario 2

- always: $\lambda_{in} = \lambda_{out}$ (goodput)
- "perfect" retransmission only when loss: $\lambda'_{in} > \lambda_{out}$
- retransmission of delayed (not lost) packet makes λ'_{in} larger (than perfect case) for same λ_{out}



a. b. c.

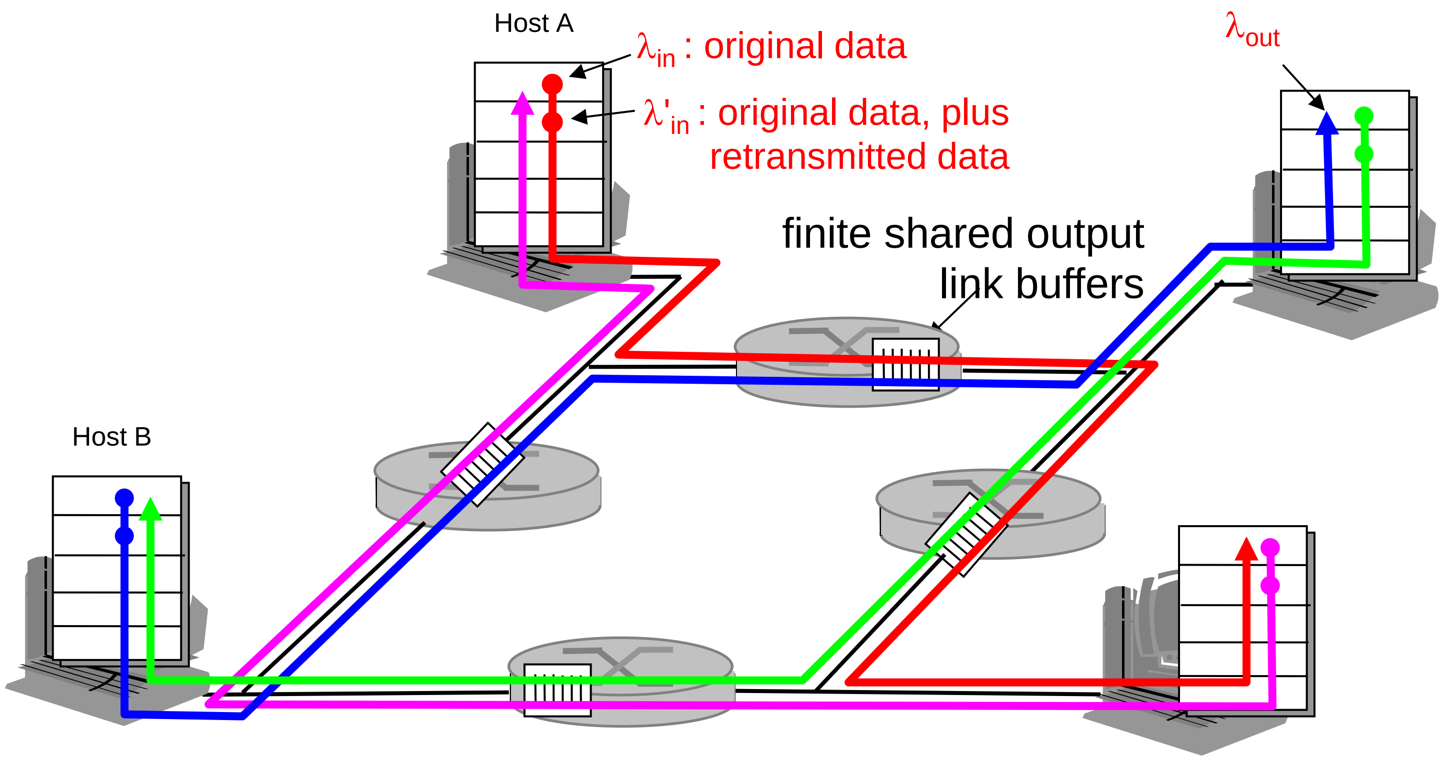
"costs" of congestion:

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

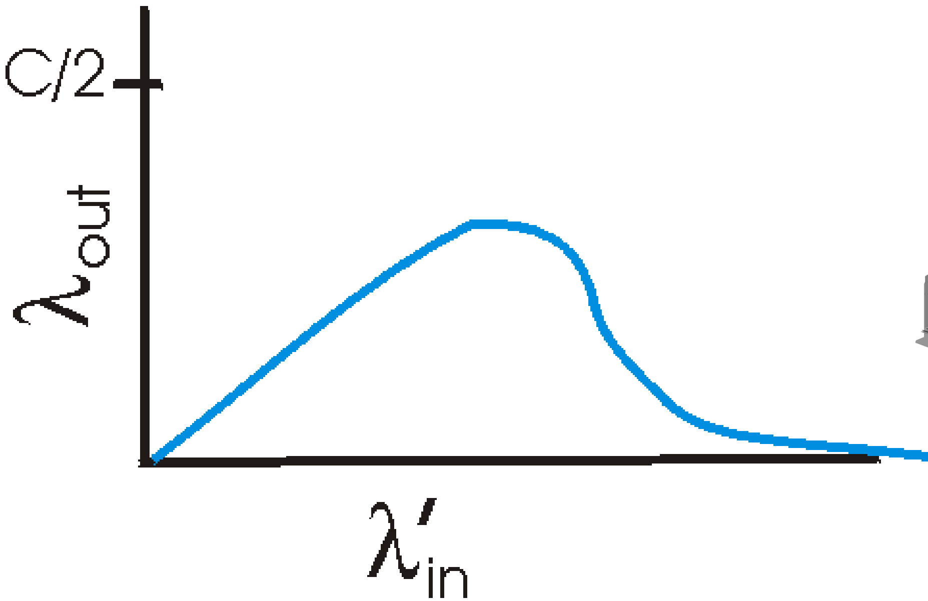
Causes/costs of congestion: scenario 3

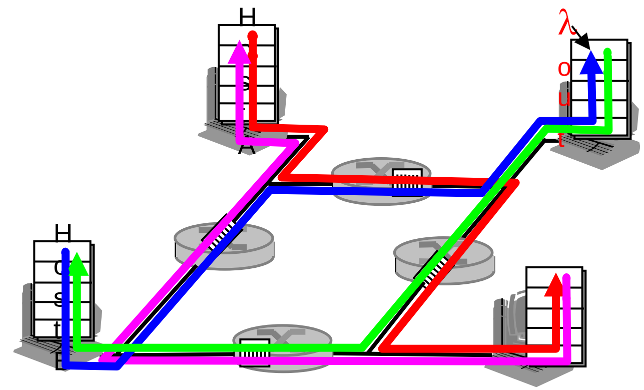
- four senders
- multihop paths
- timeout/retransmit

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



Causes/costs of congestion: scenario 3





Another "cost" of congestion:

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

Approaches towards congestion control

Two broad approaches towards congestion control:

End-end congestion control:

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

Network-assisted congestion control:

- routers provide feedback to end systems
  - single bit indicating congestion (SNA, DECbit, TCP/IP ECN, ATM)
  - explicit rate sender should send at

Case study: ATM ABR congestion control

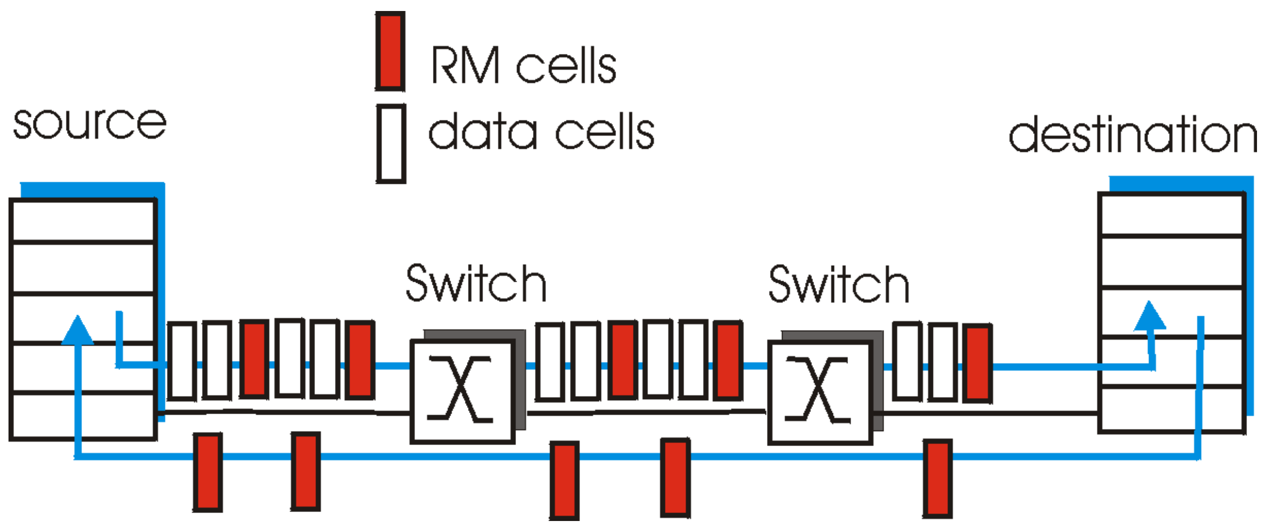
ABR: available bit rate:

- "elastic service"
- if sender's path "underloaded":
  - sender should use available bandwidth
- if sender's path congested:
  - sender throttled to minimum guaranteed rate

RM (resource management) cells:

- sent by sender, interspersed with data cells
- bits in RM cell set by switches ("*network-assisted*")
  - NI bit: no increase in rate (mild congestion)
  - CI bit: congestion indication
- RM cells returned to sender by receiver, with bits intact

Case study: ATM ABR congestion control



- two-byte ER (explicit rate) field in RM cell
  - congested switch may lower ER value in cell
  - sender' send rate thus maximum supportable rate on path
- EFCI bit in data cells: set to 1 in congested switch
  - if data cell preceding RM cell has EFCI set, sender sets CI bit in returned RM cell

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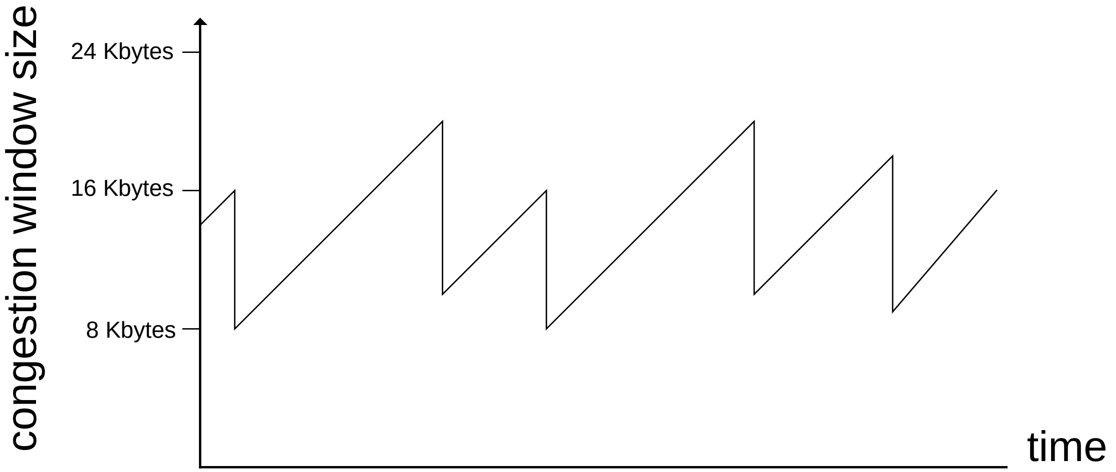
TCP congestion control: additive increase, multiplicative decrease

- *Approach:* increase transmission rate (window size), probing for usable bandwidth, until loss occurs
  - *additive increase:* increase **CongWin** by 1 MSS every RTT until loss detected
  - *multiplicative decrease*: cut **CongWin** in half after loss

Saw tooth behavior: probing for bandwidth



TCP Congestion Control: details

- sender limits transmission:

  `LastByteSent-LastByteAcked ≤ CongWin`

- Roughly,

$$\text{rate} = \frac{\text{CongWin}}{\text{RTT}} \text{ Bytes/sec}$$

- **CongWin** is dynamic, function of perceived network congestion

How does sender perceive congestion?

- loss event = timeout *or* 3 duplicate acks
- TCP sender reduces rate (**CongWin**) after loss event

three mechanisms:

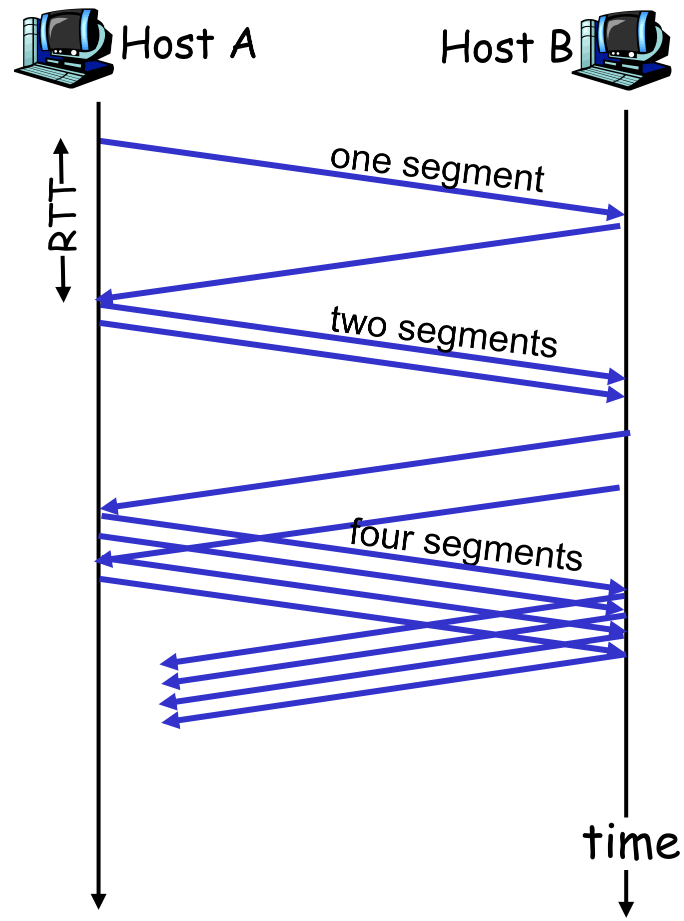
  - AIMD
  - slow start
  - conservative after timeout events

TCP Slow Start

- When connection begins, **`CongWin`** = 1 MSS
  - Example: MSS = 500 bytes & RTT = 200 msec
  - initial rate = 20 kbps
- available bandwidth may be >> MSS/RTT
  - desirable to quickly ramp up to respectable rate
- When connection begins, increase rate exponentially fast until first loss event

TCP Slow Start (more)

- When connection begins, increase rate exponentially until first loss event:
  - double **CongWin** every RTT
  - done by incrementing **CongWin** for every ACK received
- Summary: initial rate is slow but ramps up exponentially fast



Refinement: inferring loss

- After 3 dup ACKs:
  - **CongWin** is cut in half
  - window then grows linearly
- But after timeout event:
  - **CongWin** instead set to 1 MSS;
  - window then grows exponentially
  - to a threshold, then grows linearly

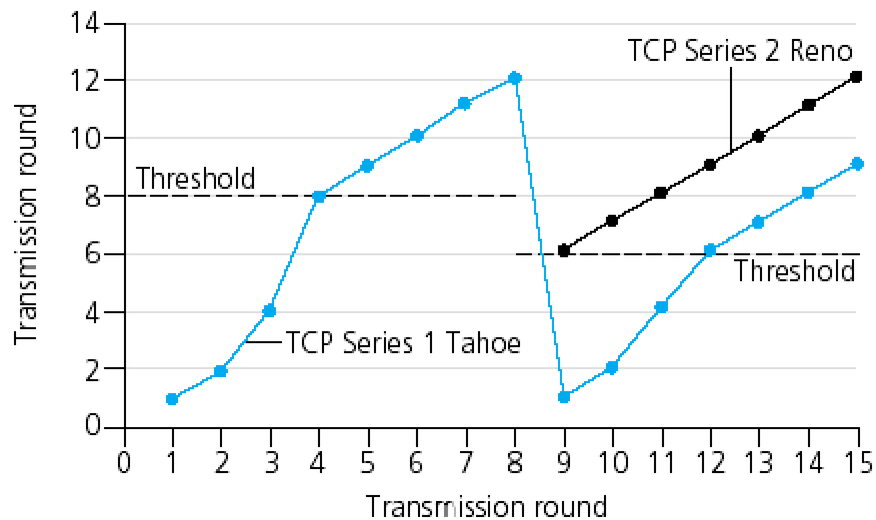
Philosophy:

- 3 dup ACKs indicates network capable of delivering some segments
- timeout indicates a "more alarming" congestion scenario

Refinement

Q: When should the exponential increase switch to linear?

A: When `CongWin` gets to 1/2 of its value before timeout.



Implementation:

- Variable Threshold
- At loss event, Threshold is set to 1/2 of CongWin just before loss event

Summary: TCP Congestion Control

- When **CongWin** is below **Threshold**, sender in slow-start phase, window grows exponentially.
- When **CongWin** is above **Threshold**, sender is in congestion-avoidance phase, window grows linearly.
- When a triple duplicate ACK occurs, **Threshold** set to **CongWin/2** and **CongWin** set to **Threshold**.
- When timeout occurs, **Threshold** set to **CongWin/2** and **CongWin** is set to 1 MSS.

TCP sender congestion control

State	Event	TCP Sender Action	Commentary
Slow Start (SS)	ACK receipt for previously unacked data	CongWin = CongWin + MSS, If (CongWin > Threshold) set state to “Congestion Avoidance”	Resulting in a doubling of CongWin every RTT
Congestion Avoidance (CA)	ACK receipt for previously unacked data	CongWin = CongWin+MSS * (MSS/CongWin)	Additive increase, resulting in increase of CongWin by 1 MSS every RTT
SS or CA	Loss event detected by triple duplicate ACK	Threshold = CongWin/2, CongWin = Threshold, Set state to “Congestion Avoidance”	Fast recovery, implementing multiplicative decrease. CongWin will not drop below 1 MSS.
SS or CA	Timeout	Threshold = CongWin/2, CongWin = 1 MSS, Set state to “Slow Start”	Enter slow start
SS or CA	Duplicate ACK	Increment duplicate ACK count for segment being acked	CongWin and Threshold not changed

TCP throughput

- What's the average throughout of TCP as a function of window size and RTT?
  - Ignore slow start
- Let W be the window size when loss occurs.
- When window is W, throughput is W/RTT
- Just after loss, window drops to W/2, throughput to W/2RTT.
- Average throughout: .75 W/RTT

TCP Futures: TCP over "long, fat pipes"

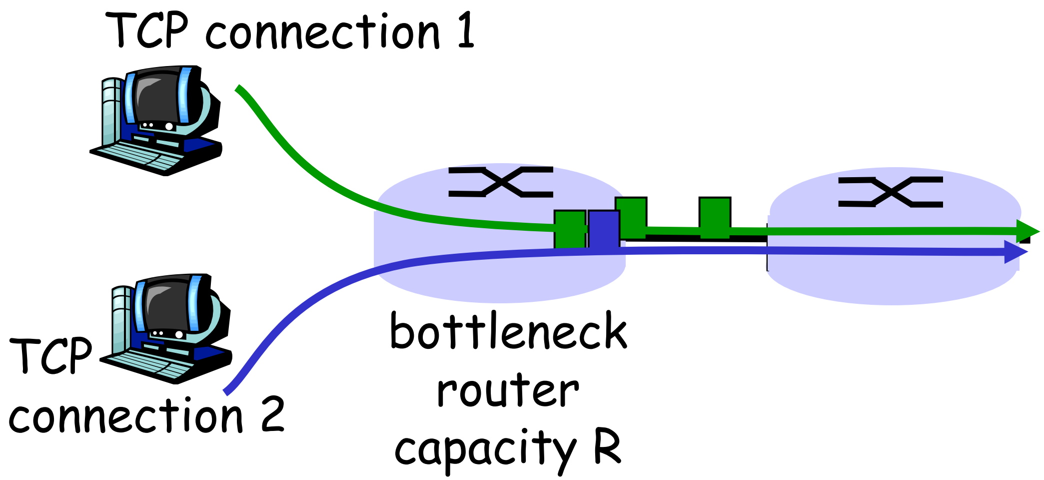
- Example: 1500 byte segments, 100ms RTT, want 10 Gbps throughput
- Requires window size W = 83,333 in-flight segments
- Throughput in terms of loss rate:

$$\frac{1.22 \cdot MSS}{RTT\sqrt{L}}$$

- ➜ $L = 2 \cdot 10^{-10}$ *Wow*
- New versions of TCP for high-speed

TCP Fairness

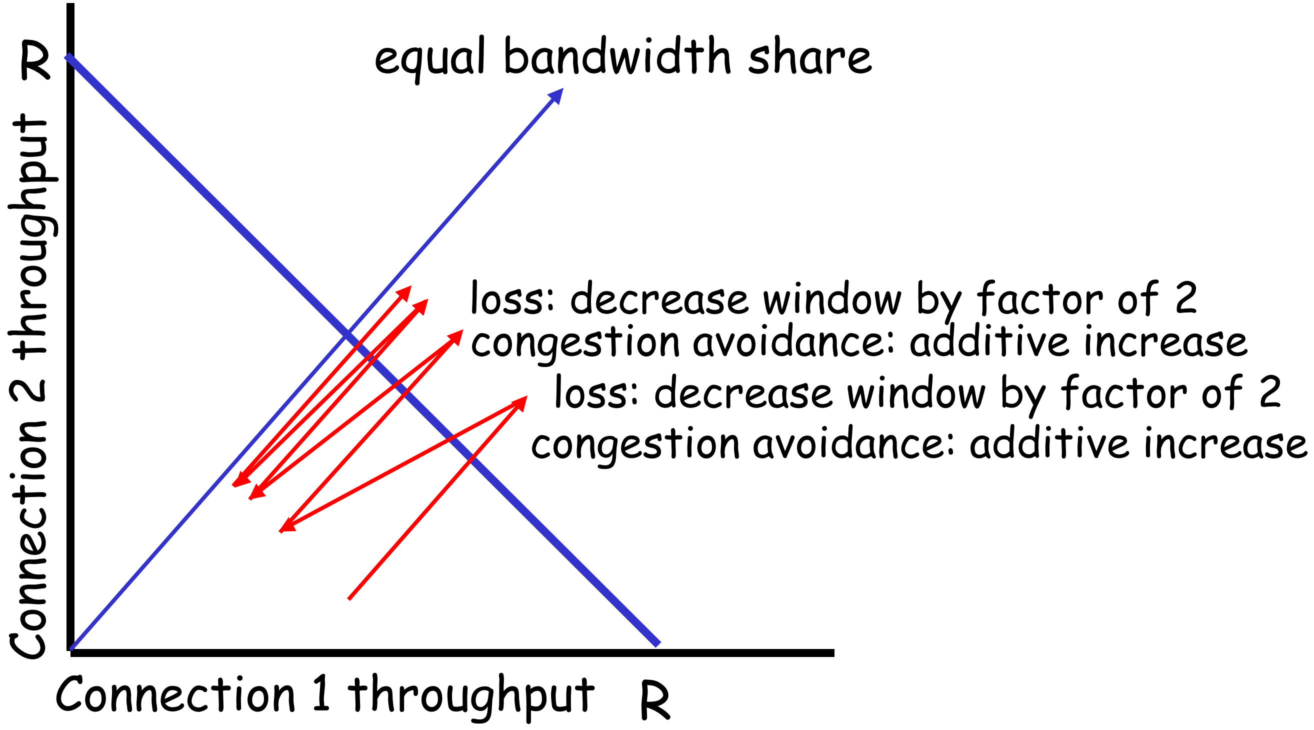
Fairness goal: if K TCP sessions share same bottleneck link of bandwidth R, each should have average rate of R/K



Why is TCP fair?

Two competing sessions:

- Additive increase gives slope of 1, as throughout increases
- multiplicative decrease decreases throughput proportionally



Fairness (more)

Fairness and UDP

- Multimedia apps often do not use TCP
  - do not want rate throttled by congestion control
- Instead use UDP:
  - pump audio/video at constant rate, tolerate packet loss
- Research area: TCP friendly

Fairness and parallel TCP connections

- nothing prevents app from opening parallel connections between 2 hosts.
- Web browsers do this
- Example: link of rate R supporting 9 connections;
  - new app asks for 1 TCP, gets rate R/10
  - new app asks for 11 TCPs, gets R/2 !