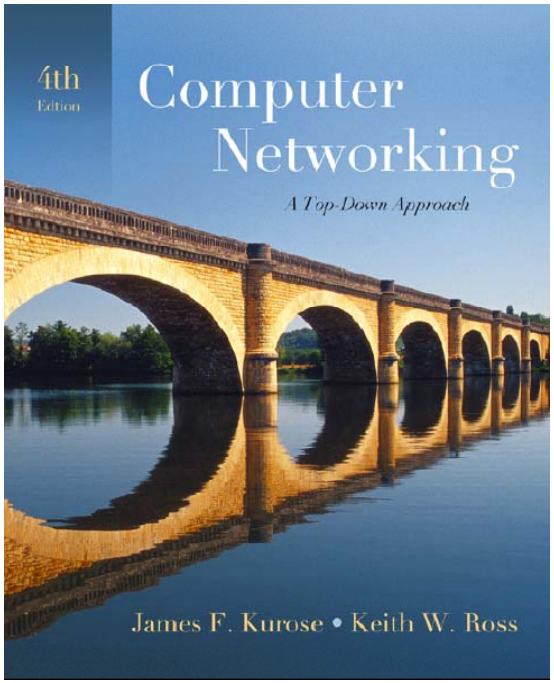
Chapter 3
Transport Layer
傳輸層



Computer Networking: A Top Down Approach 4th edition.
Jim Kurose, Keith Ross
Addison-Wesley, July 2007.

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Chapter 3: Transport Layer

Our goals:

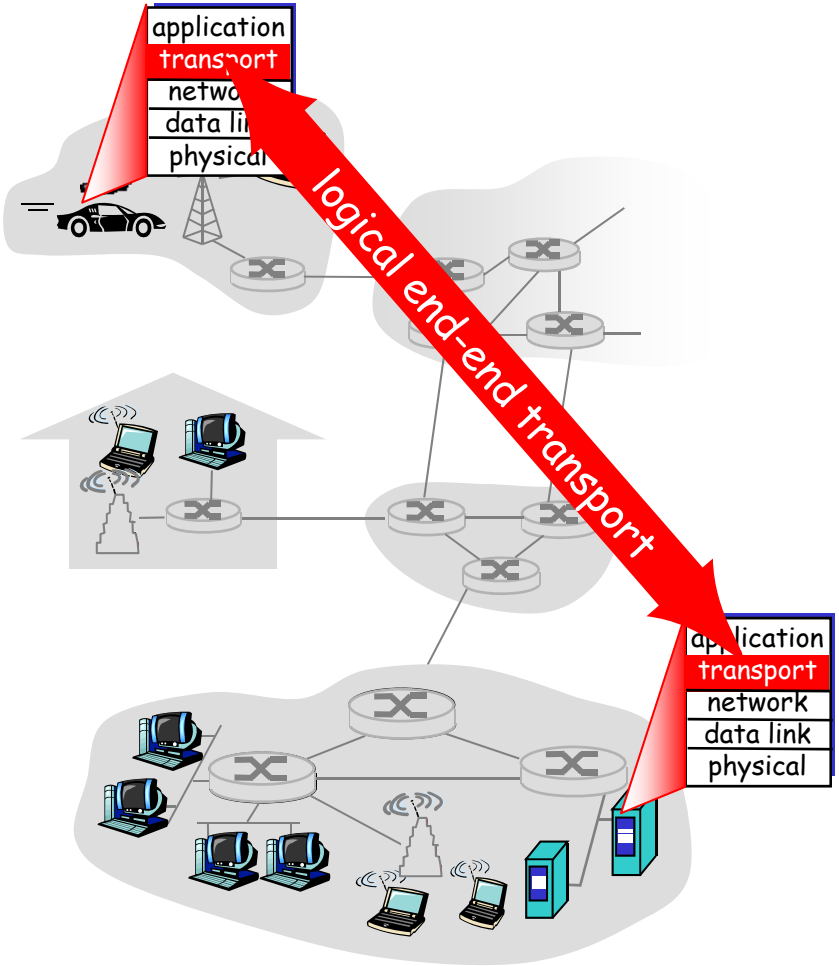
- understand principles behind transport layer services:
  - multiplexing/demultiplexing
    多工與解多工
  - reliable data transfer
    可信賴的資料傳輸
  - flow control
    流量控制
  - congestion control
    壅塞控制
- learn about transport layer protocols in the Internet:
  - UDP: connectionless transport
  - TCP: connection-oriented transport
  - TCP congestion control

Chapter 3 outline

- 3.1 Transport-layer services
  傳輸層服務
- 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

Transport services and protocols

- provide *logical communication* 邏輯通訊 between app processes running on different hosts
- transport protocols run in end systems
  - send side: breaks app messages into segments 區段, passes to network layer
  - rcv side: reassembles segments into messages, passes to app layer
- more than one transport protocol available to apps
  - Internet: TCP and UDP



Transport vs. network layer

傳輸層與網路層之間的關係

- *network layer:* logical communication between hosts
- *transport layer:* logical communication between processes
  - relies on, enhances, network layer services

Household analogy:例子

12 kids sending letters to 12 kids

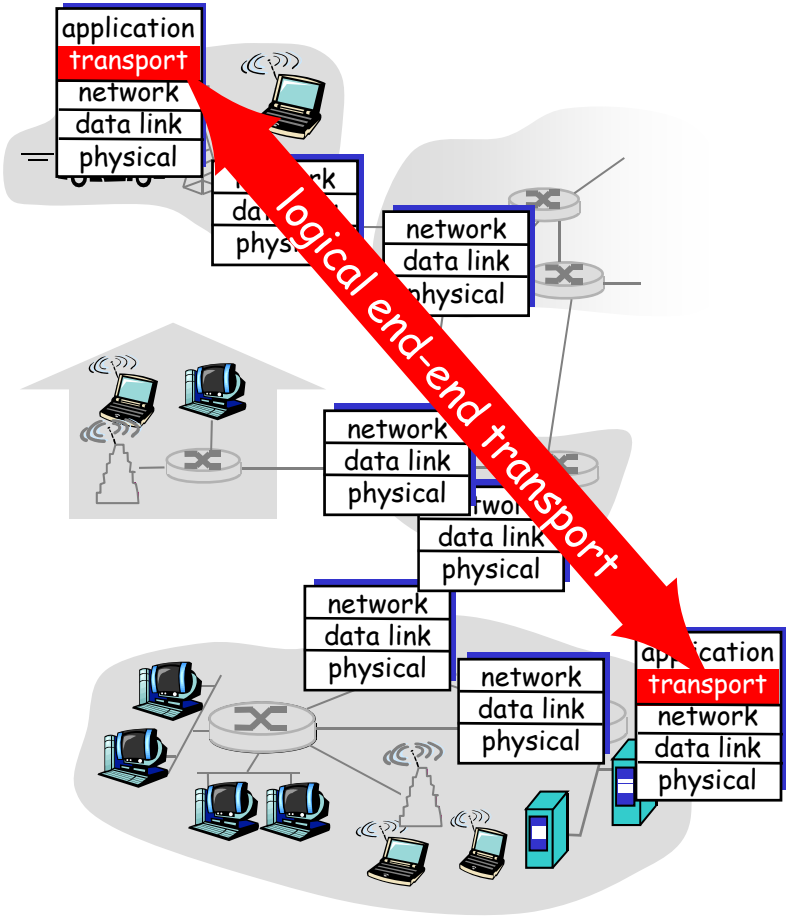
12小孩寄信給12個小孩

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

Internet transport-layer protocols

網際網路傳輸層協定

- reliable, in-order delivery (TCP)
  - congestion control
  - flow control
  - connection setup
- unreliable, unordered delivery: UDP
  - no-frills extension of "best-effort" IP
- services not available:
  - delay guarantees
  - bandwidth guarantees



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  多工與解多工
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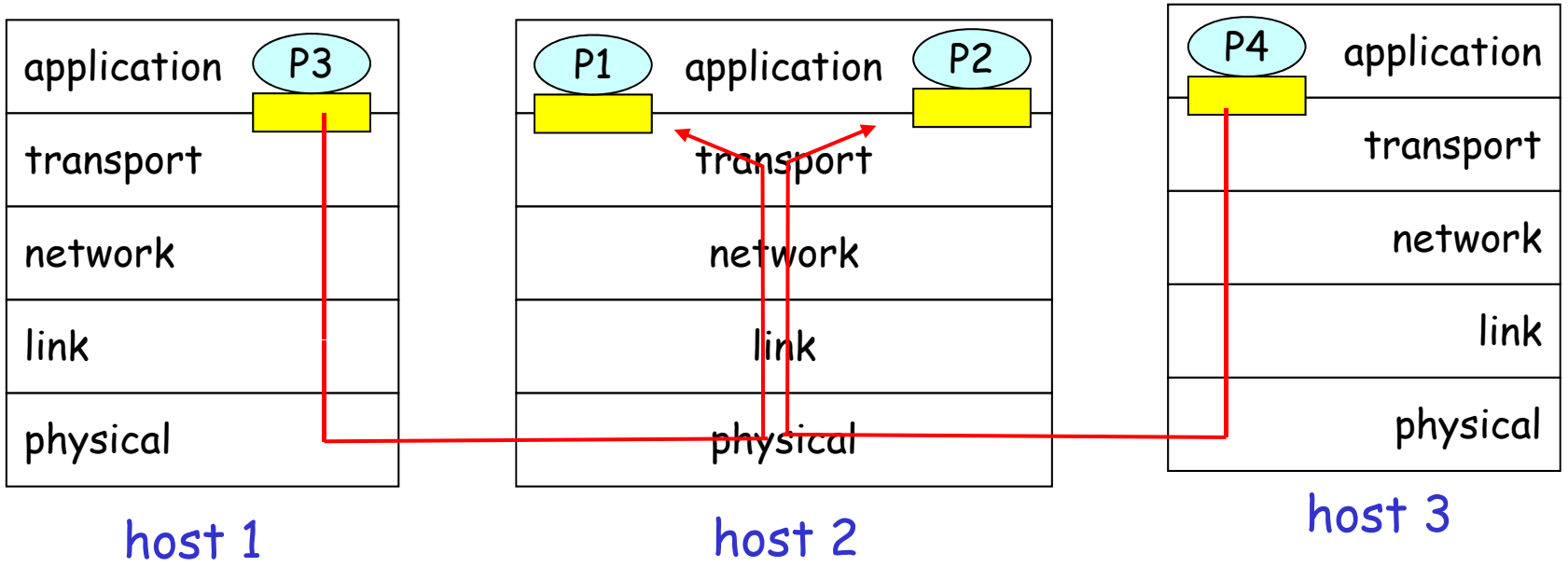
Multiplexing/demultiplexing

多工與解多工

Demultiplexing at rcv host:
delivering received segments to correct socket
在接收端解多工

Multiplexing at send host:
gathering data from multiple sockets, enveloping data with header (later used for demultiplexing)
在傳送端多工

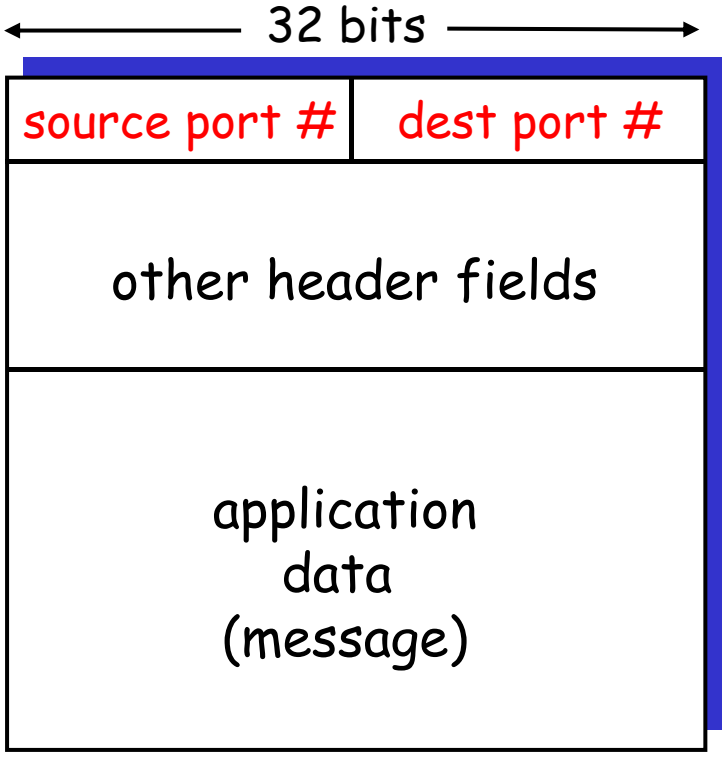
= socket = process



How demultiplexing works

解多工的運作方式

- host receives IP datagrams
  主機收到IP封包後
  - each datagram has source IP address, destination IP address
  - each datagram carries 1 transport-layer segment
    每個封包都攜帶一個傳輸層的區段
  - each segment has source, destination port number
    每個區段都有port number
- host uses IP addresses & port numbers to direct segment to appropriate socket
  用IP Address及port number決定



TCP/UDP segment format

Connectionless demultiplexing
UDP的解多工

- Create sockets with port numbers:

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

- UDP socket identified by two-tuple:

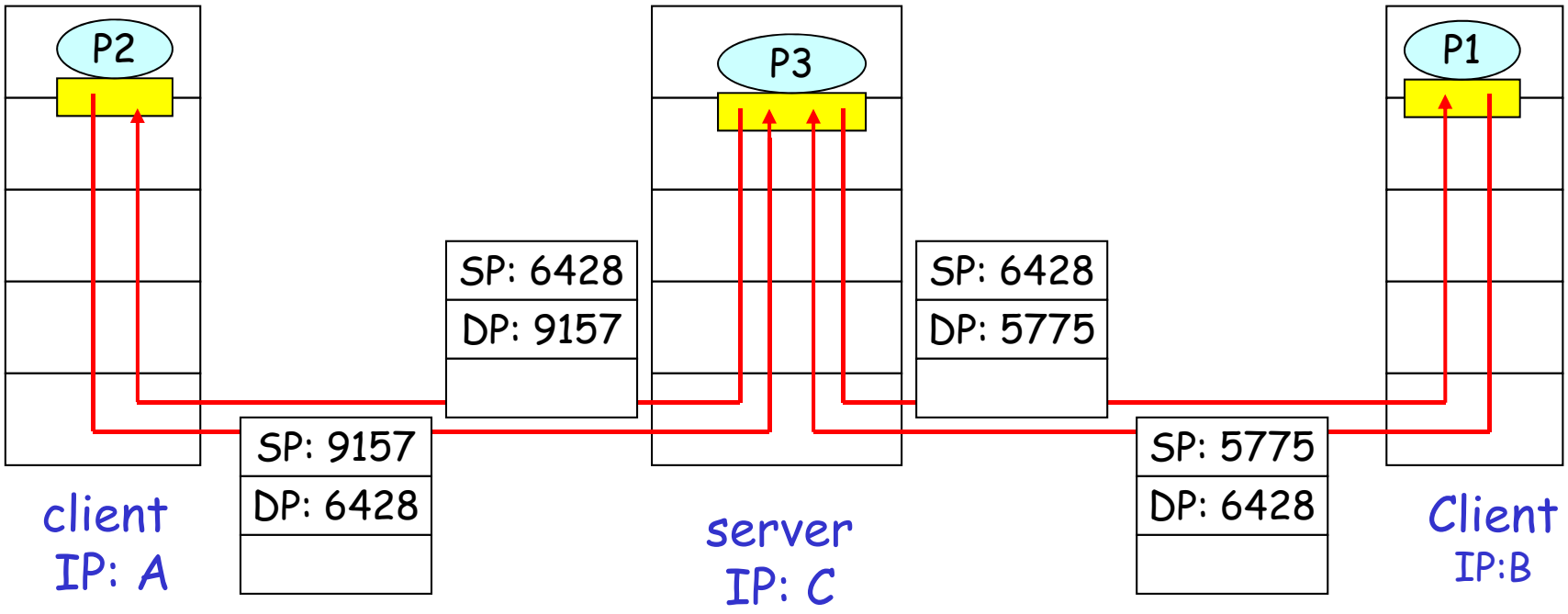
(dest IP address, dest port number)

- When host receives UDP segment:
  - checks destination port number in segment
  - directs UDP segment to socket with that port number
- IP datagrams with different source IP addresses and/or source port numbers directed to same socket

返回位址

Connectionless demux (cont)

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



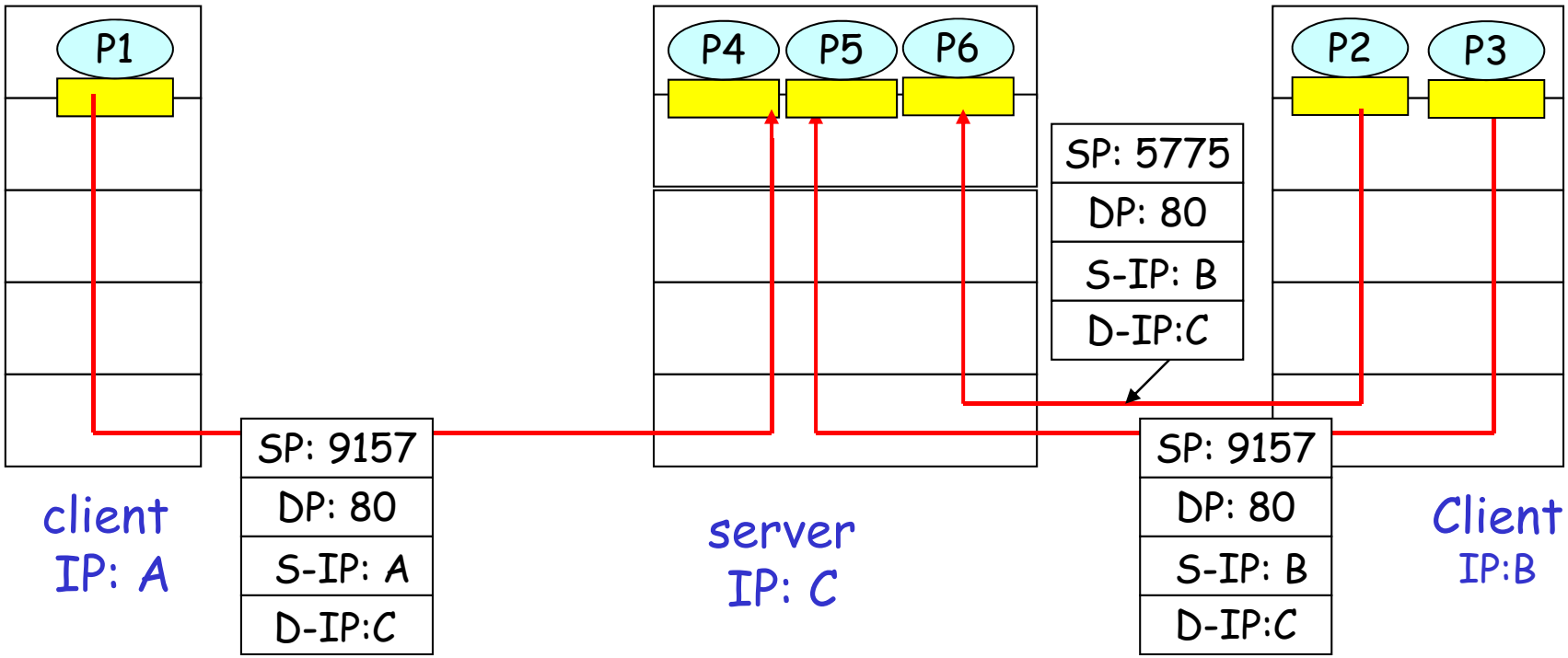
SP provides "return address"

Connection-oriented demux
TCP的解多工

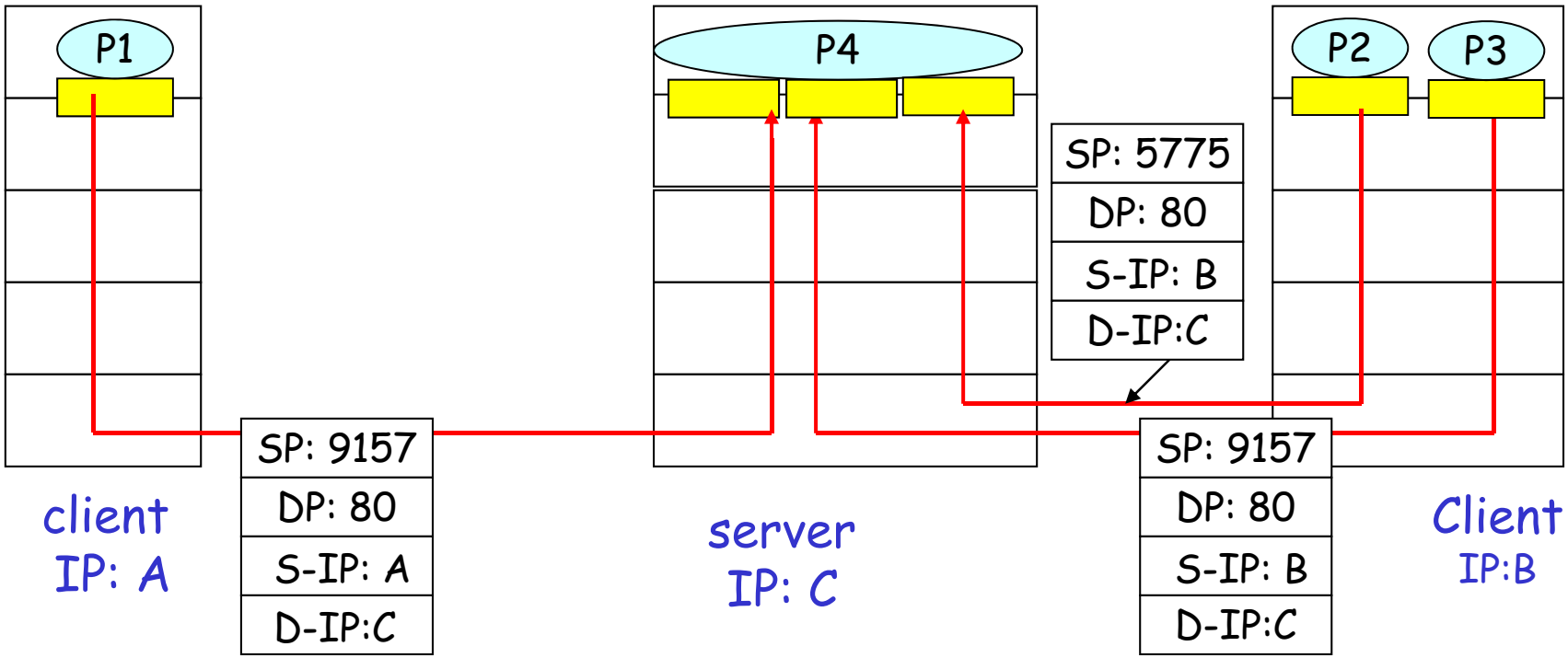
- TCP socket identified by 4-tuple:
  - source IP address
  - source port number
  - dest IP address
  - dest port number

    因為需要事先建立連線
- recv host uses all four values to direct segment to appropriate socket
- Server host may support many simultaneous TCP sockets:
  - each socket identified by its own 4-tuple
- Web servers have different sockets for each connecting client
  - non-persistent HTTP will have different socket for each request

Connection-oriented demux (cont)



Connection-oriented demux: Threaded Web Server



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  無連結傳輸：UDP
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UDP: User Datagram Protocol [RFC 768]

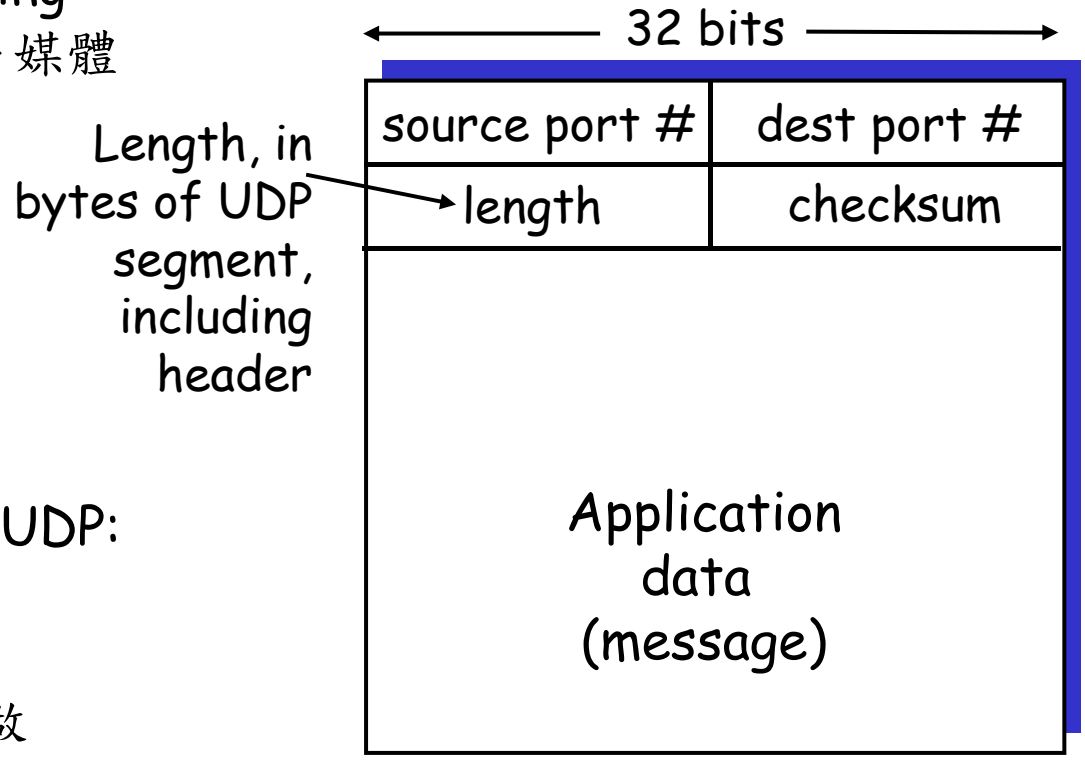
- "no frills," "bare bones" Internet transport protocol
- "best effort 盡力去做" service, UDP segments may be:
  - Lost 封包可能遺失
  - delivered out of order to app 封包可能不依順序
- *connectionless:*
  - no handshaking between UDP sender, receiver 沒有握手機制
  - each UDP segment handled independently of others 各自獨立

Why is there a UDP?

- no connection establishment (which can add delay) 減少delay
- simple: no connection state at sender, receiver
- small segment header 標頭負擔較小
- no congestion control: UDP can blast away as fast as desired 沒有壅塞控制的機制

UDP: more

- often used for streaming multimedia apps 串流多媒體
  - loss tolerant
  - rate sensitive
- other UDP uses
  - DNS
  - SNMP
- reliable transfer over UDP: add reliability at application layer
  未達成的服務由應用層做
  - application-specific error recovery!



UDP segment format

UDP checksum 錯誤偵測

Goal: detect "errors" (e.g., flipped bits) in transmitted segment 偵測錯誤

Sender:

- treat segment contents as sequence of 16-bit integers
- checksum: addition (1's complement sum) of segment contents
- sender puts checksum value into UDP checksum field

Receiver:

- compute checksum of received segment
- check if computed checksum equals checksum field value:
  - NO - error detected
  - YES - no error detected. *But maybe errors nonetheless?* More later ....

身份證號碼驗證 check-sum

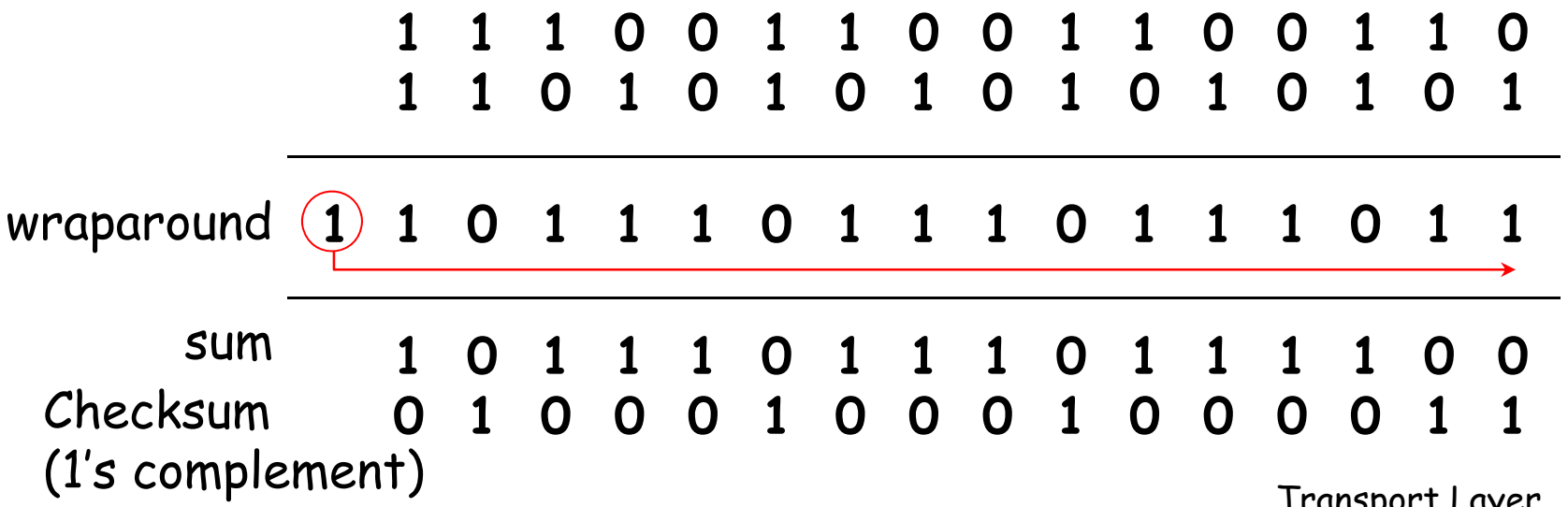
1. 英文代號以下表轉換成數字
   A=10 台北市　J=18 新竹縣 S=26 高雄縣
   B=11 台中市　K=19 苗栗縣 T=27 屏東縣
   C=12 基隆市　L=20 台中縣 U=28 花蓮縣
   D=13 台南市　M=21 南投縣 V=29 台東縣
   E=14 高雄市　N=22 彰化縣 * W=32 金門縣
   F=15 台北縣　O=35 新竹市 X=30 澎湖縣
   G=16 宜蘭縣　P=23 雲林縣 Y=31 陽明山
   H=17 桃園縣　Q=24 嘉義縣 * Z=33 連江縣
   * I=34 嘉義市 R=25 台南縣
2. 英文轉成的數字, 個位數乘９再加上十位數
3. 各數字從右到左依次乘１、２、３、４・・・・８
4. 求出(2),(3)之和
5. 求出(4)除10後之餘數,用10減該餘數,結果就是檢查碼，若餘數為0，檢查碼就是 0。

6. http://web.htps.tn.edu.tw/cen/other/files/pp/index.htm

Internet Checksum Example

網際網路驗證碼

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



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- 3.4 Principles of reliable data transfer
  可靠資料傳輸原理
- 3.5 Connection-oriented transport: TCP
  - segment structure
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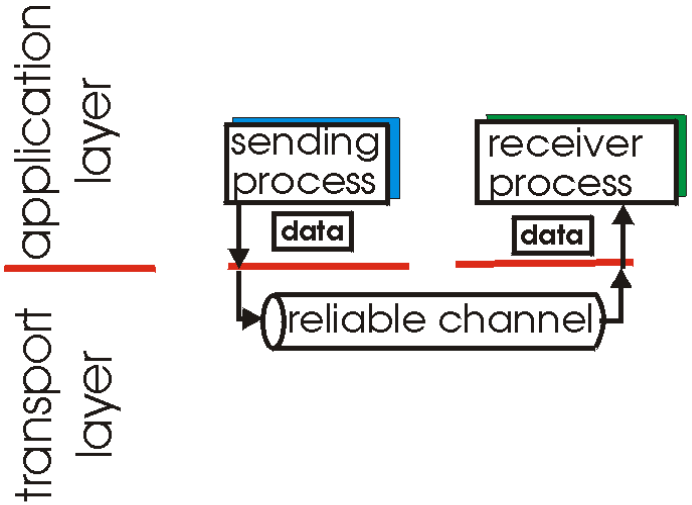
Principles of Reliable data transfer

- important in app., transport, link layers 應用層、傳輸層、連結層
- top-10 list of important networking topics!



(a) provided service

- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

Principles of Reliable data transfer

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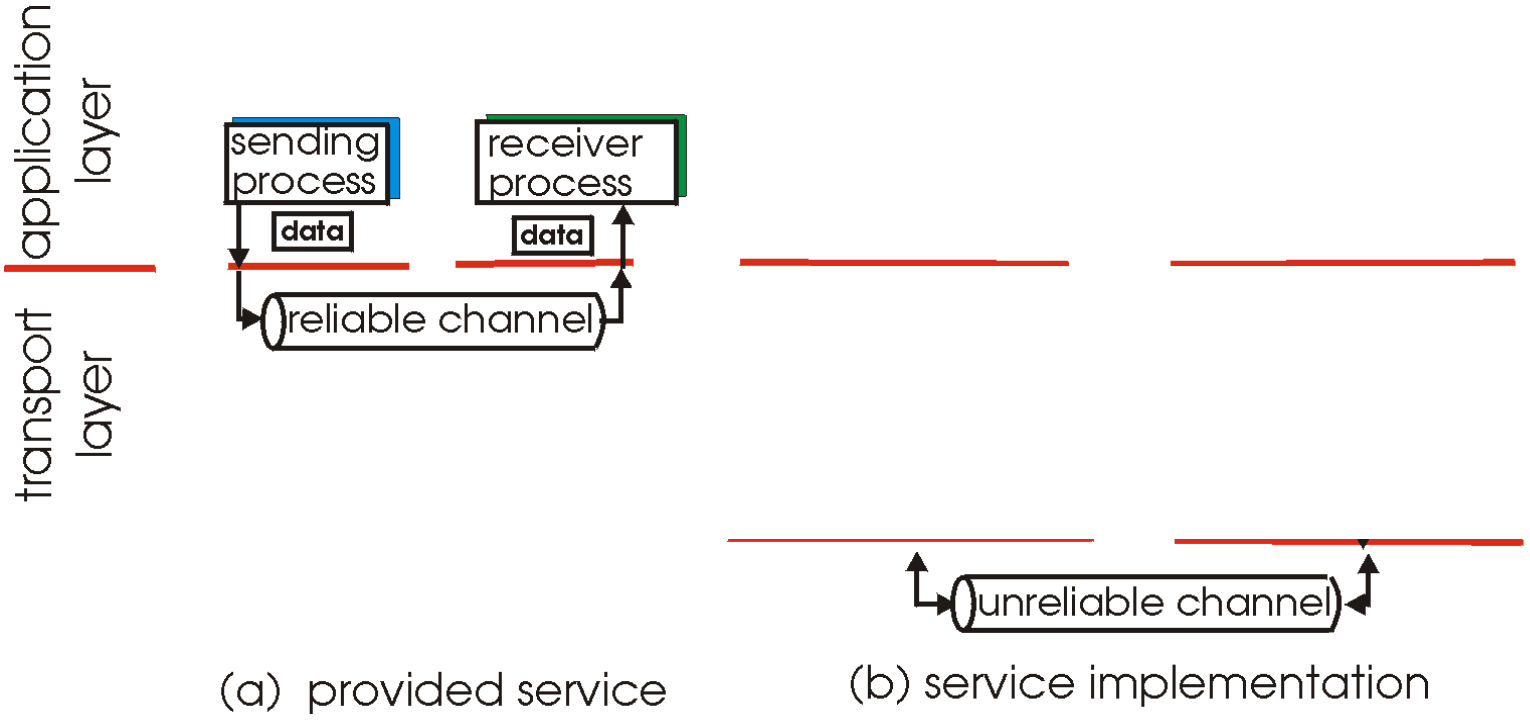

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Principles of Reliable data transfer

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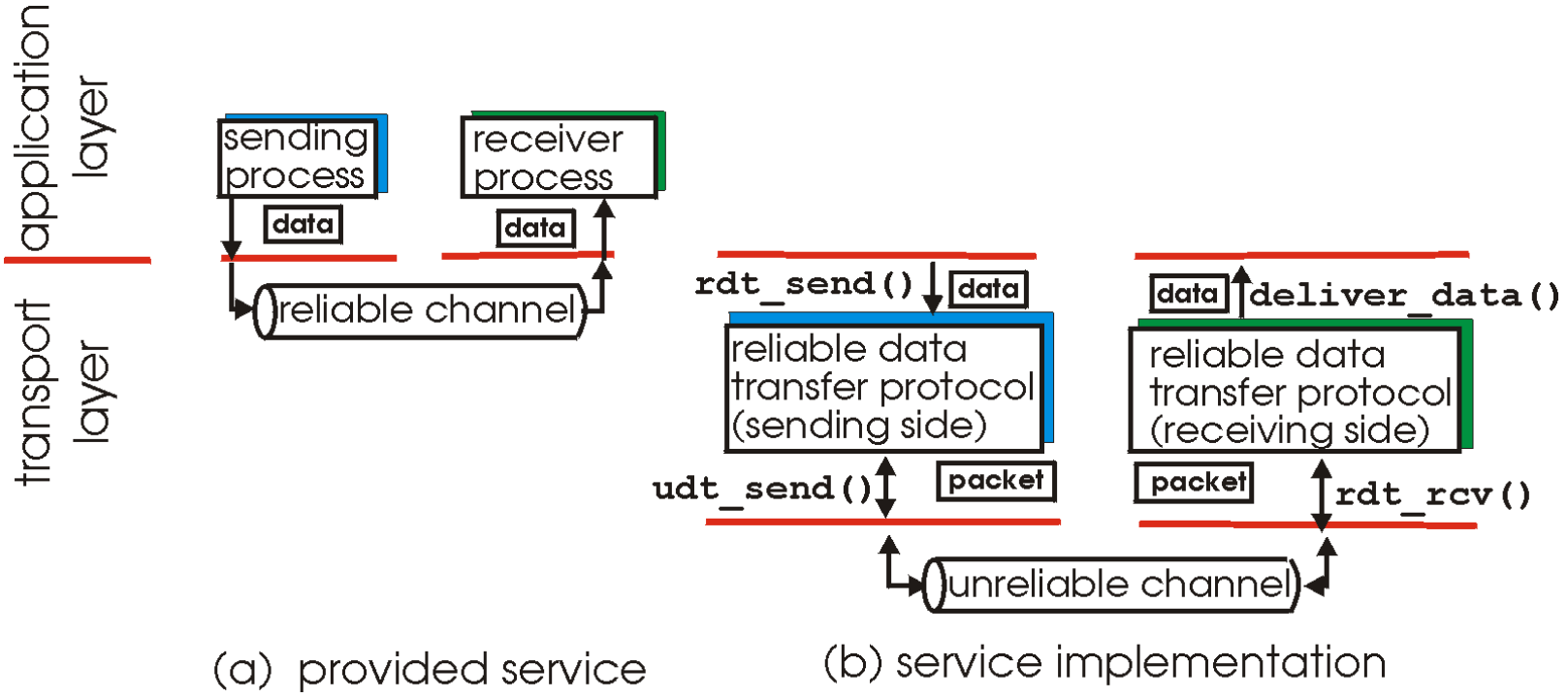
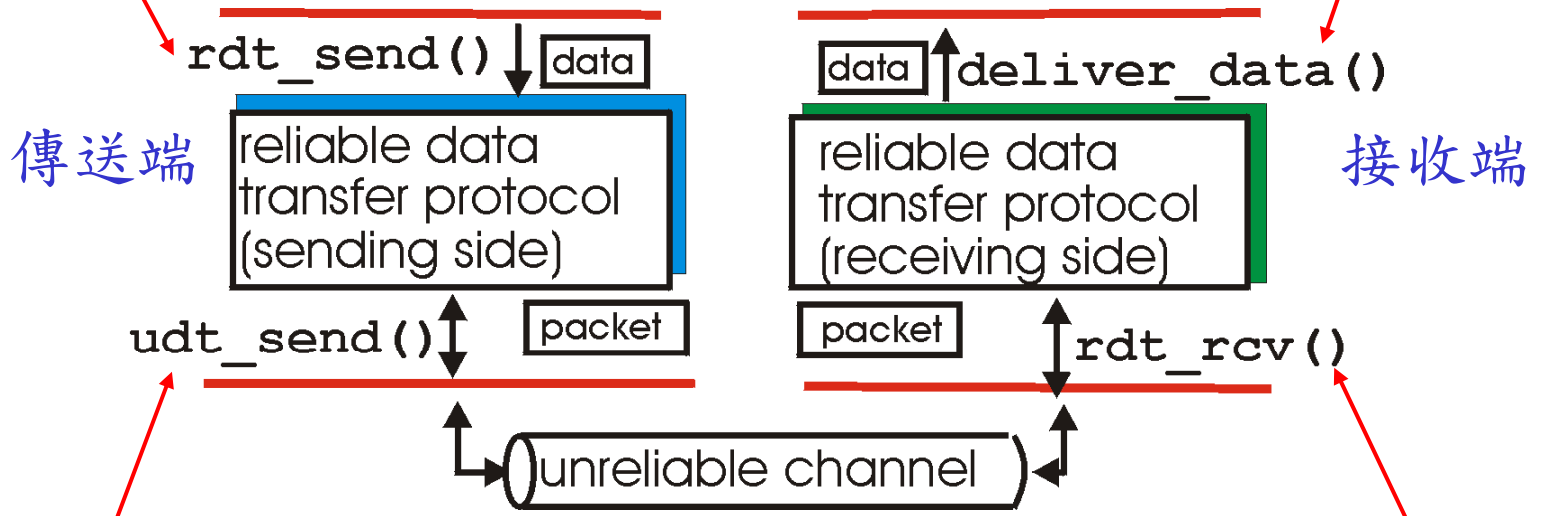


- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

Reliable data transfer: getting started

rdt_send(): 由上層應用程式呼叫，可以把資料傳送給接收端的上層

deliver_data(): 將資料傳送到上層



udt_send(): 由rdt呼叫，可以將封包傳送給另一端

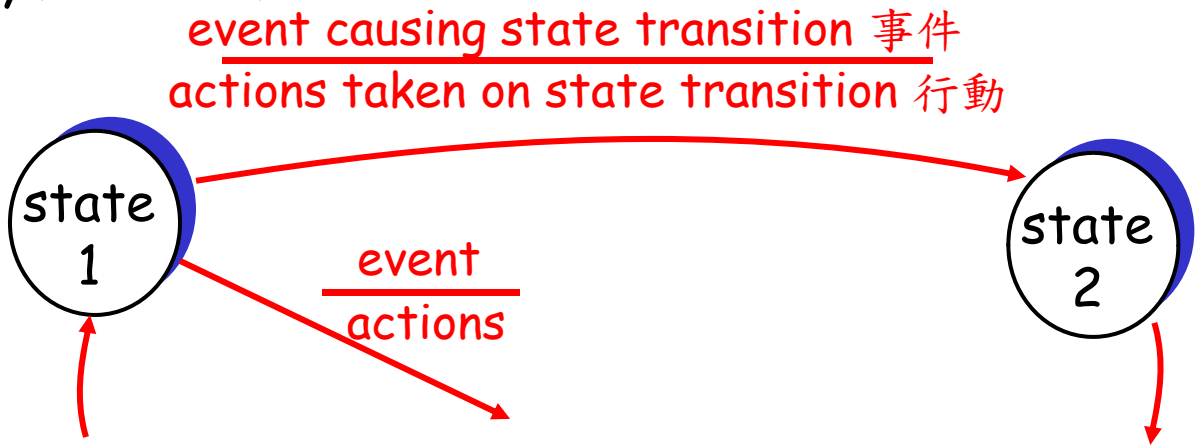
rdt_rcv(): 當封包抵達時，呼叫此函式

Reliable data transfer: getting started

We'll:

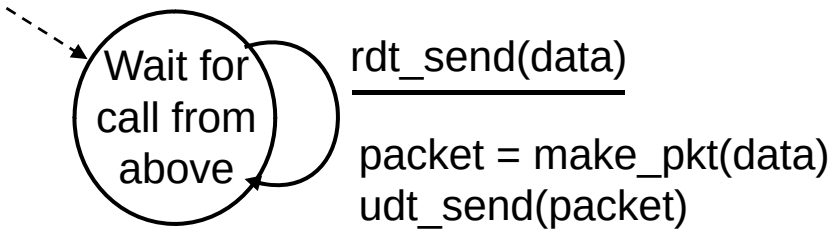
- incrementally develop sender, receiver sides of reliable data transfer protocol (rdt)
- consider only unidirectional data transfer
  - but control info will flow on both directions!
- use finite state machines (FSM有限狀態機) to specify sender, receiver

state: when in this "state" next state uniquely determined by next event

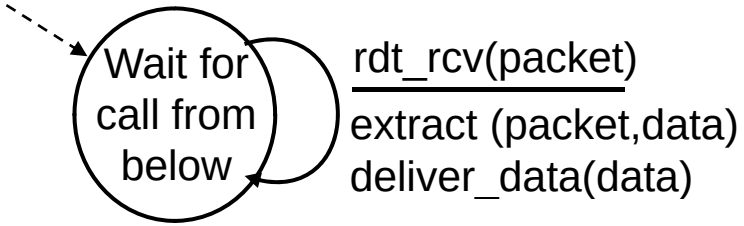


Rdt1.0: reliable transfer over a reliable channel

- underlying channel perfectly reliable
  完全可信賴的底層通道
  - no bit errors 資料不會錯誤
  - no loss of packets 封包不會遺失
- separate FSMs for sender, receiver:
  - sender sends data into underlying channel
  - receiver read data from underlying channel



Sender
傳送端

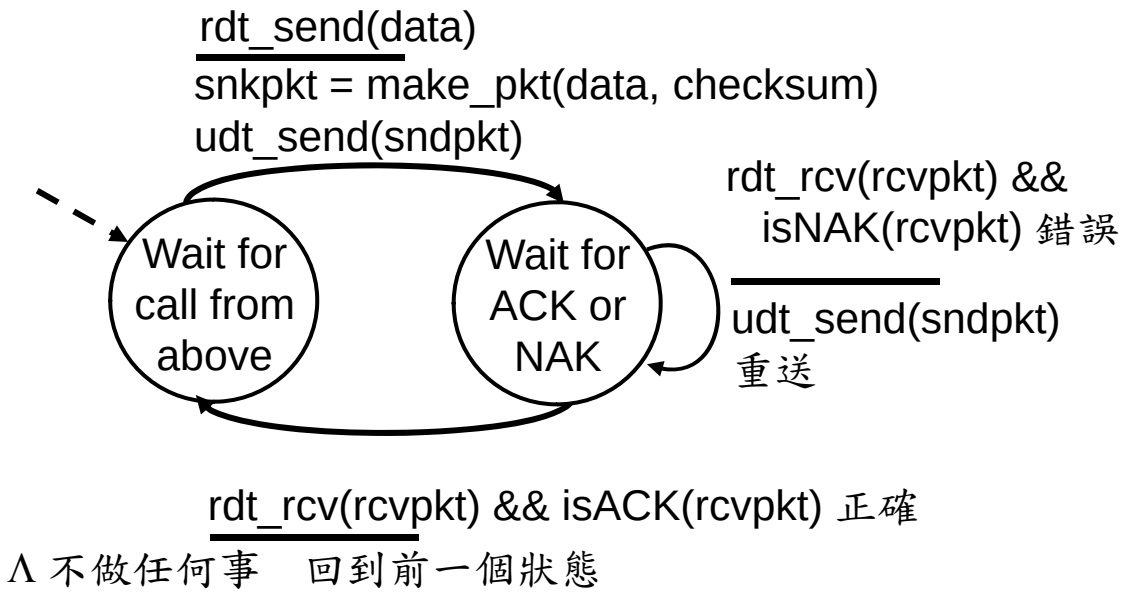


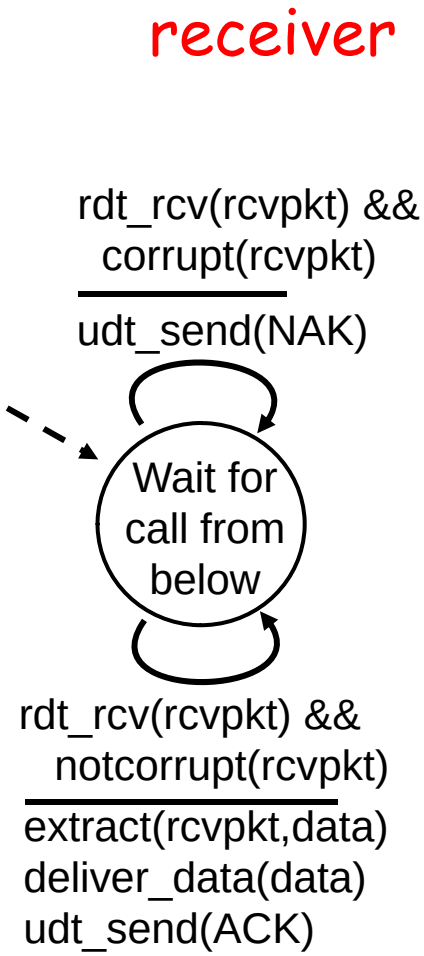
Receiver
接收端

Rdt2.0: channel with bit errors 會有位元錯誤

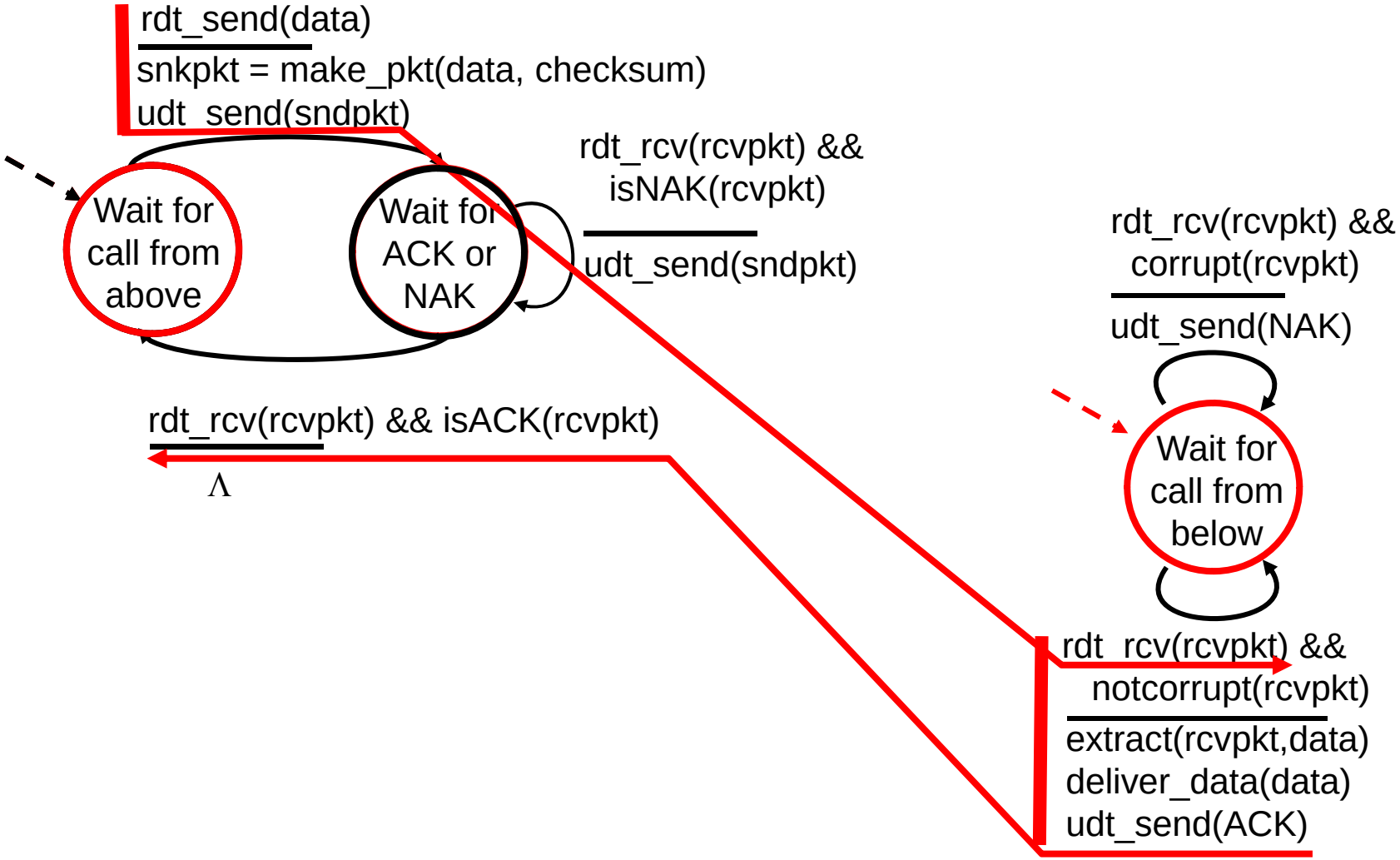
- underlying channel may flip bits in packet
  - checksum to detect bit errors 用驗證碼偵測錯誤
- *the* question: how to recover from errors:
  - *acknowledgements (ACKs):* receiver explicitly tells sender that pkt received OK 使用【正確】訊息
  - *negative acknowledgements (NAKs):* receiver explicitly tells sender that pkt had errors 使用【錯誤】訊息
  - sender retransmits pkt on receipt of NAK
    當沒收到【正確】或收到【錯誤】時重送
- new mechanisms in **`rdt2.0`** (beyond **`rdt1.0`**):
  - error detection
  - receiver feedback: control msgs (ACK,NAK) rcvr->sender

rdt2.0: FSM specification

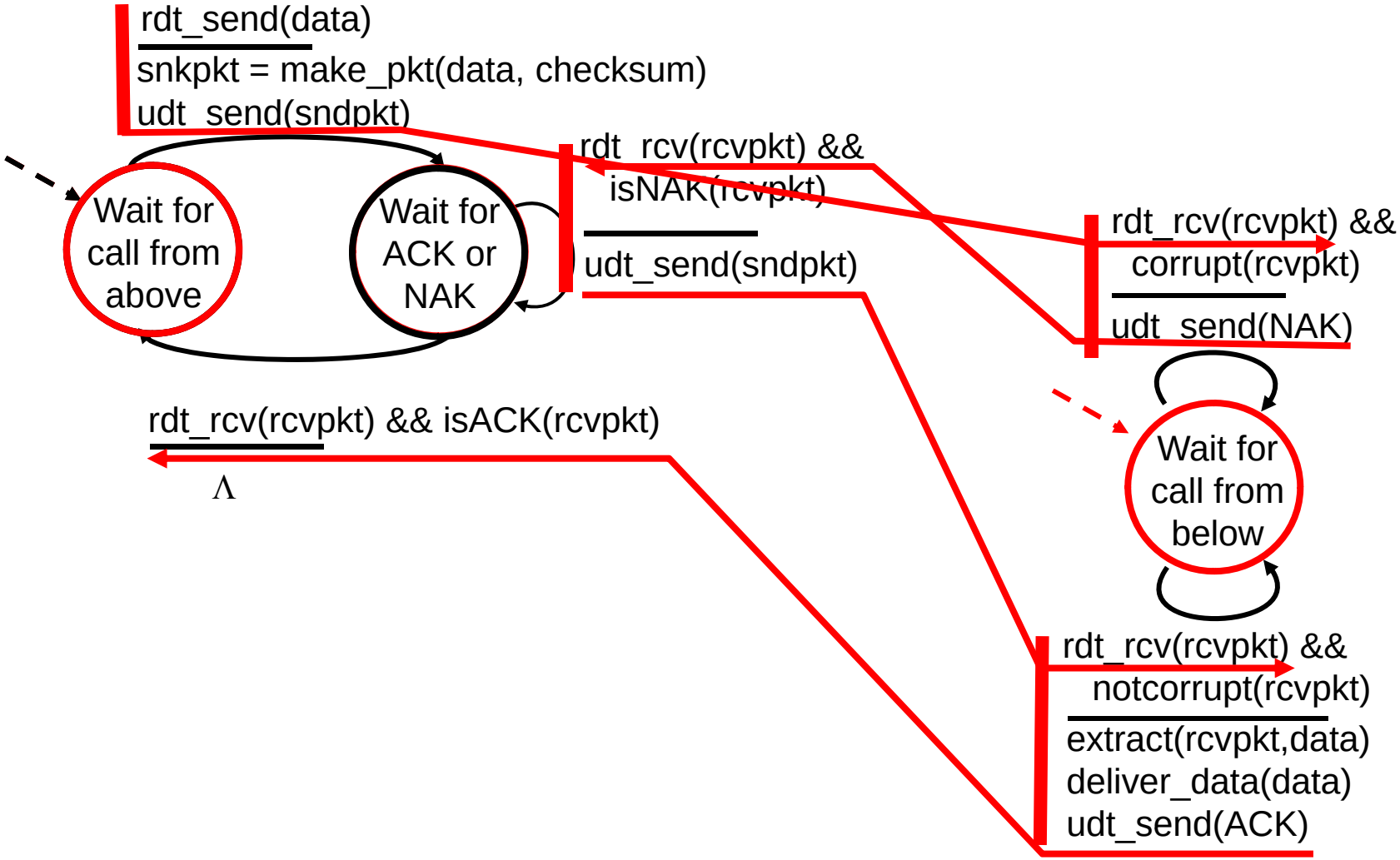




rdt2.0: operation with no errors



rdt2.0: error scenario



rdt2.0 has a fatal flaw! 致命問題

What happens if ACK/NAK corrupted?
如果連ACK/NAK也錯了

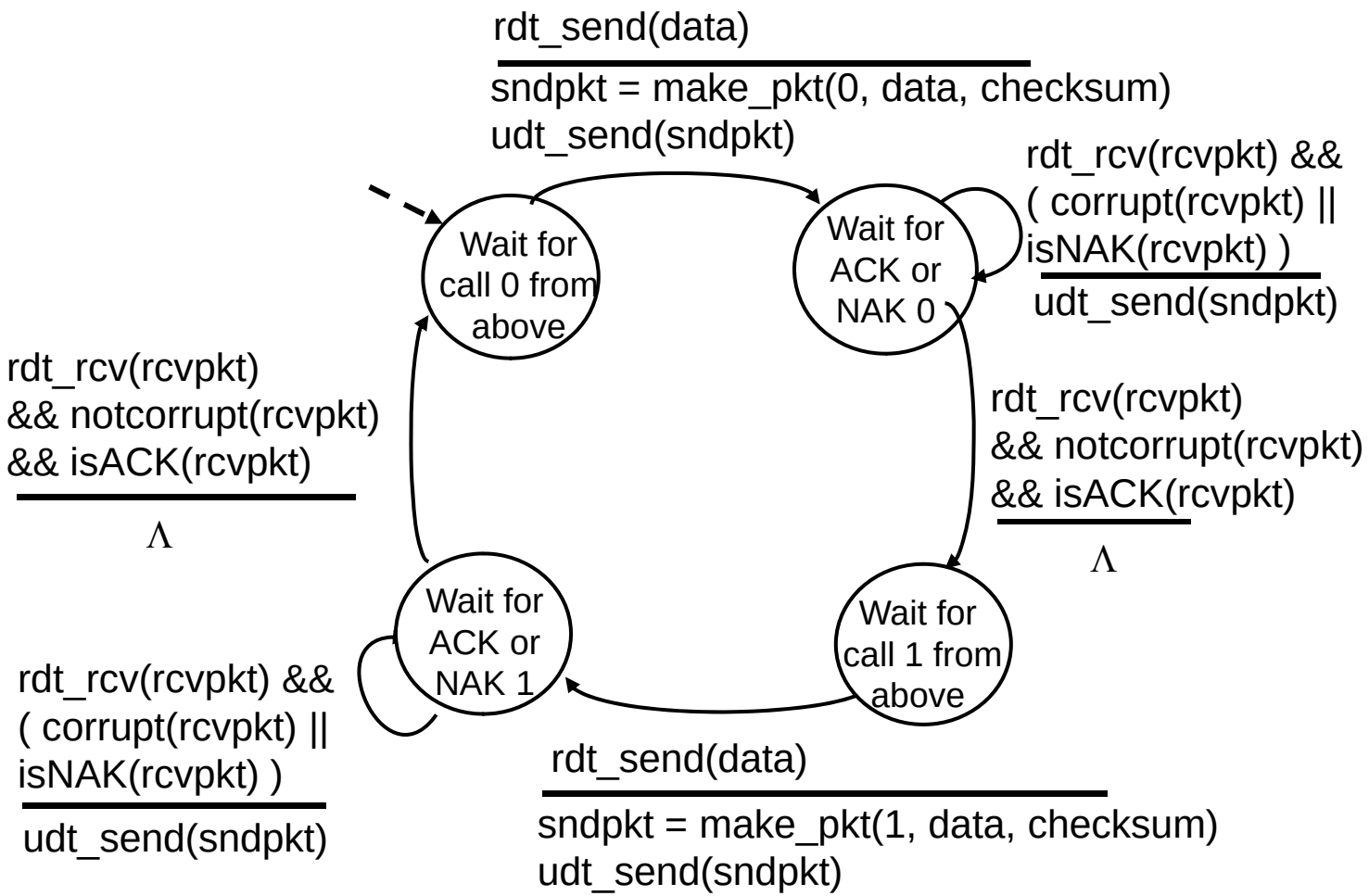
- sender doesn't know what happened at receiver!
  傳送端不知道接收端的狀態
- can't just retransmit: possible duplicate
  不能重送：可能發生重覆傳送的問題

Handling duplicates:

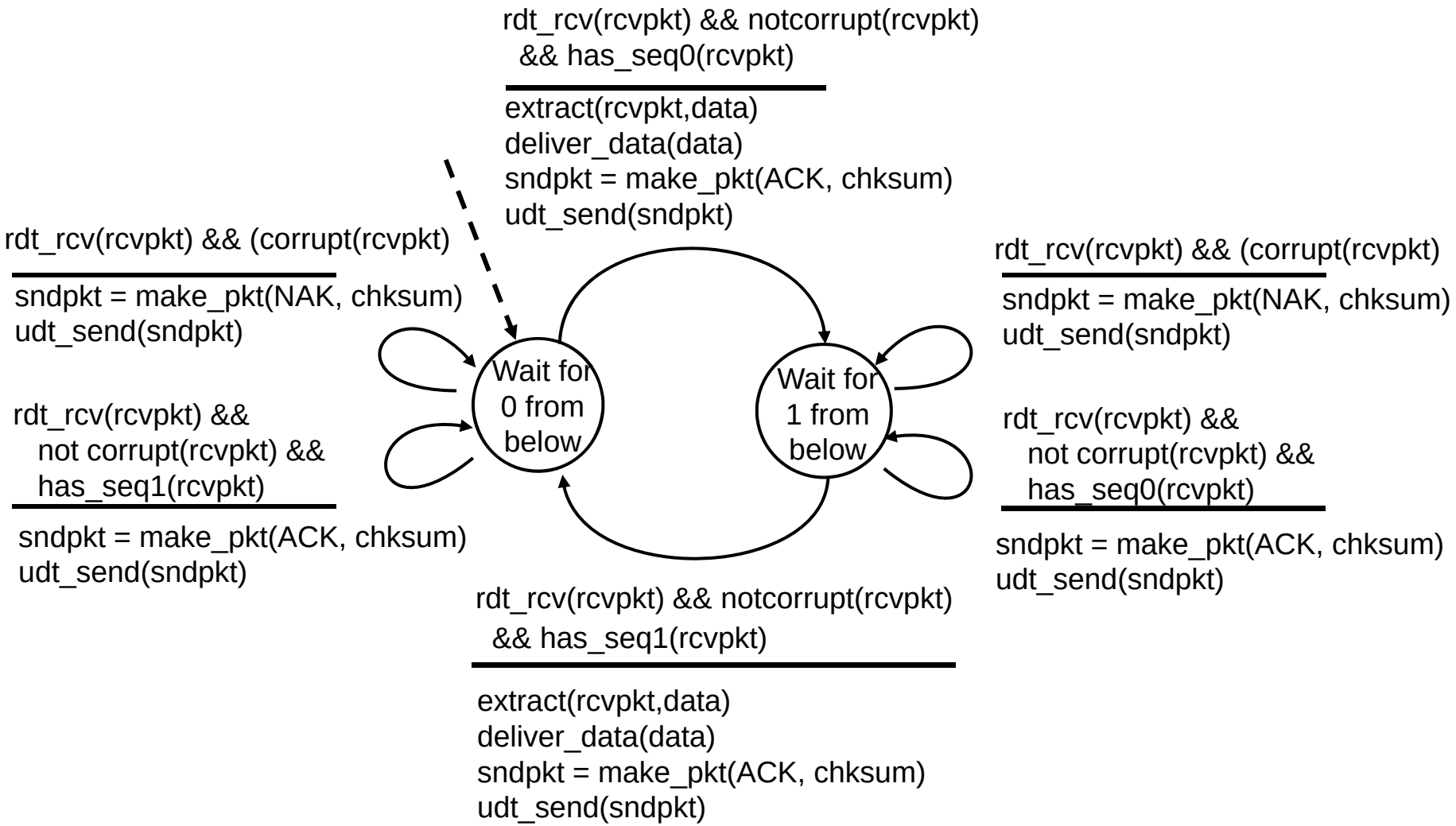
- sender retransmits current pkt if ACK/NAK garbled
- sender adds *sequence number* to each pkt
  增加序號
- receiver discards (doesn't deliver up) duplicate pkt
  丟棄相同序號的封包

stop and wait
Sender sends one packet, then waits for receiver Response
停止與等待協定

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?
  為什麼用兩個序號(0,1)就夠了
- 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

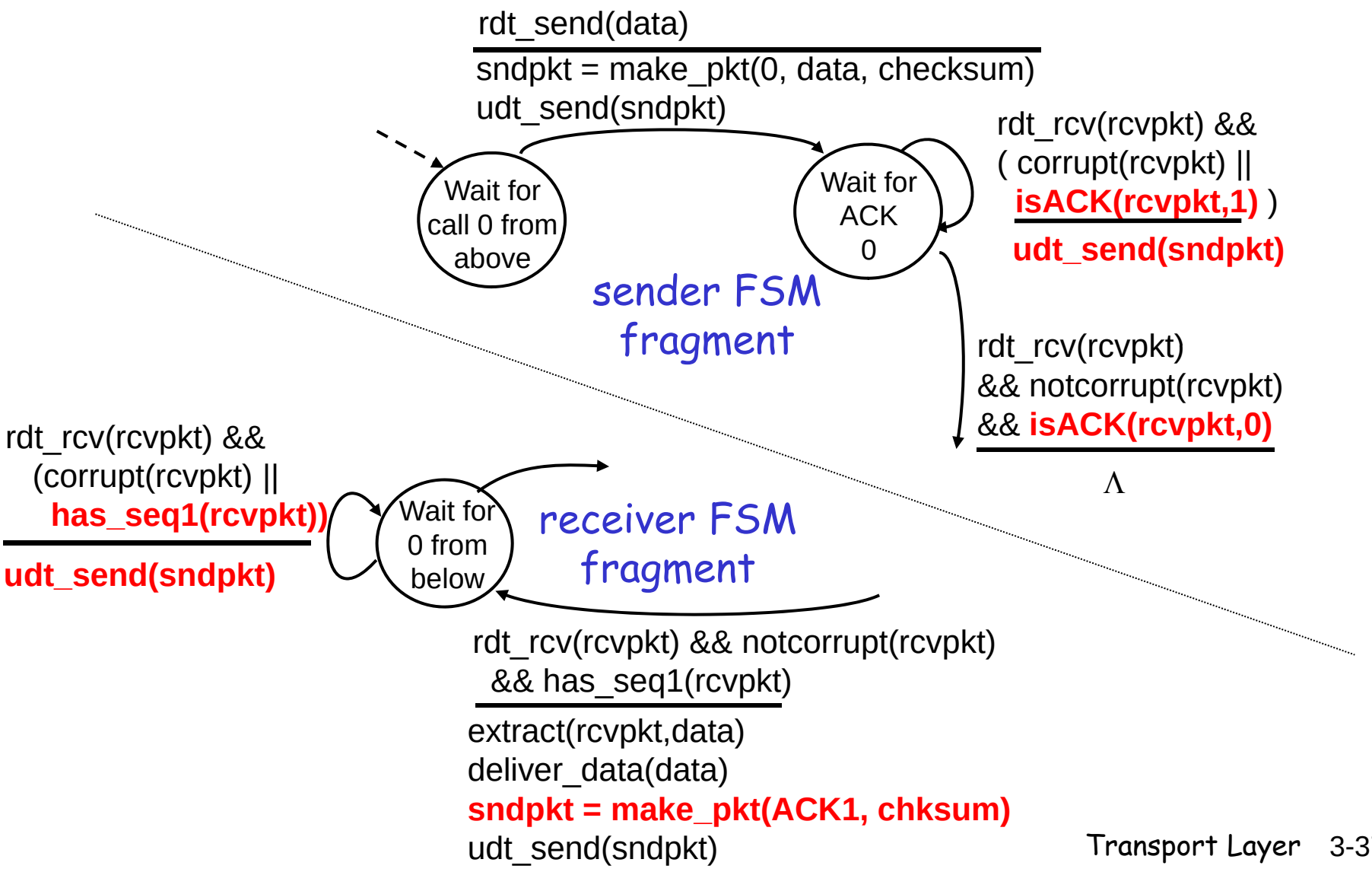
rdt2.2: a NAK-free protocol
不用NAK的協定

- same functionality as rdt2.1, using ACKs only
  只用ACK，不用NAK
- instead of NAK, receiver sends ACK for last pkt received OK
  - receiver must *explicitly* include seq # of pkt being ACKed

  告訴傳送端，最後一個成功接收的封包序號
- duplicate ACK at sender results in same action as NAK: *retransmit current pkt*

  重覆收到相同的ACK，表示有封包錯誤，因此重新傳送封包

rdt2.2: sender, receiver fragments



rdt3.0: channels with errors *and* loss

位元錯誤及封包遺失

New assumption:

underlying channel can also lose packets (data or ACKs)

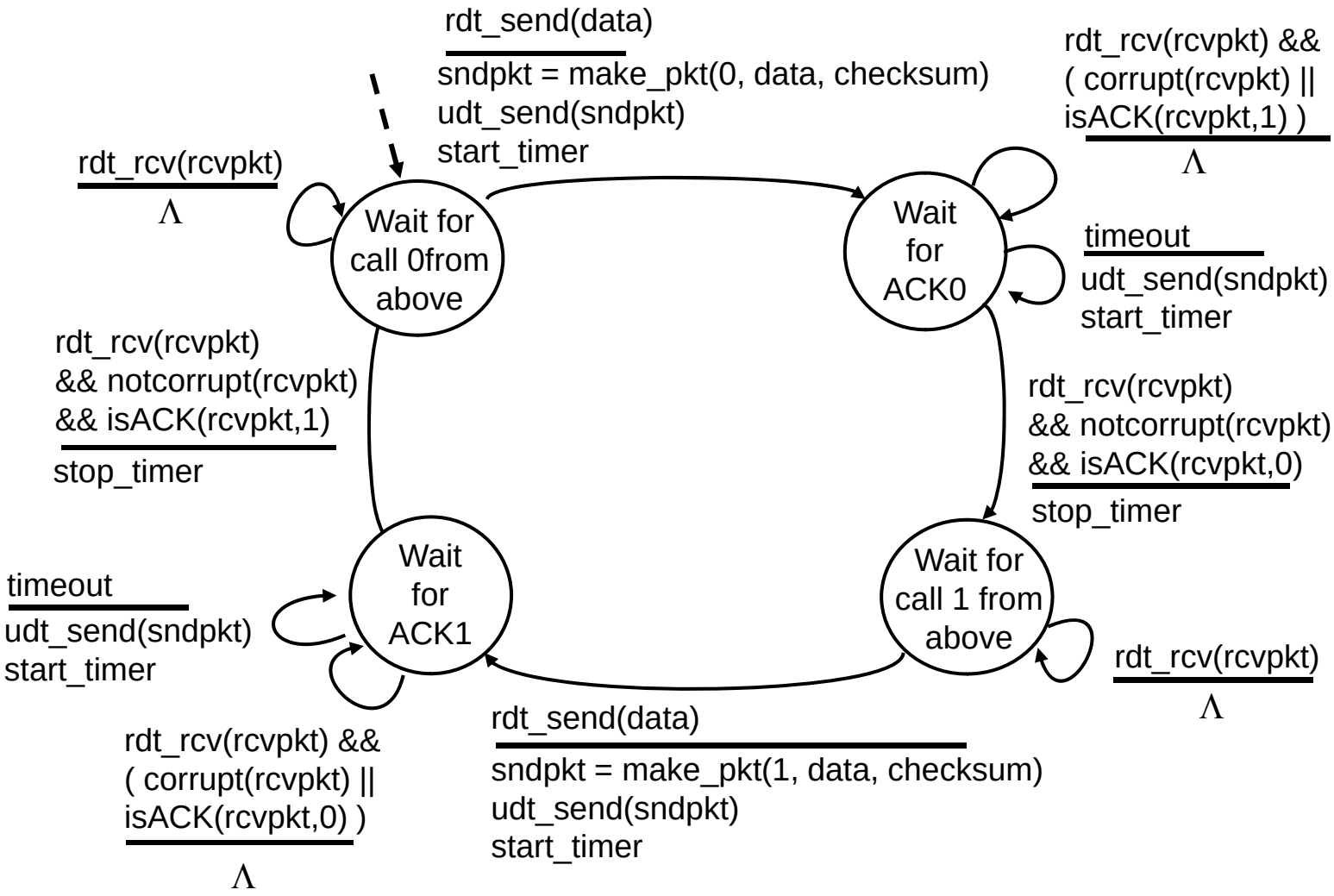
- checksum, seq. #, ACKs, retransmissions will be of help, but not enough

Approach: sender waits "reasonable" amount of time for ACK (timeout)

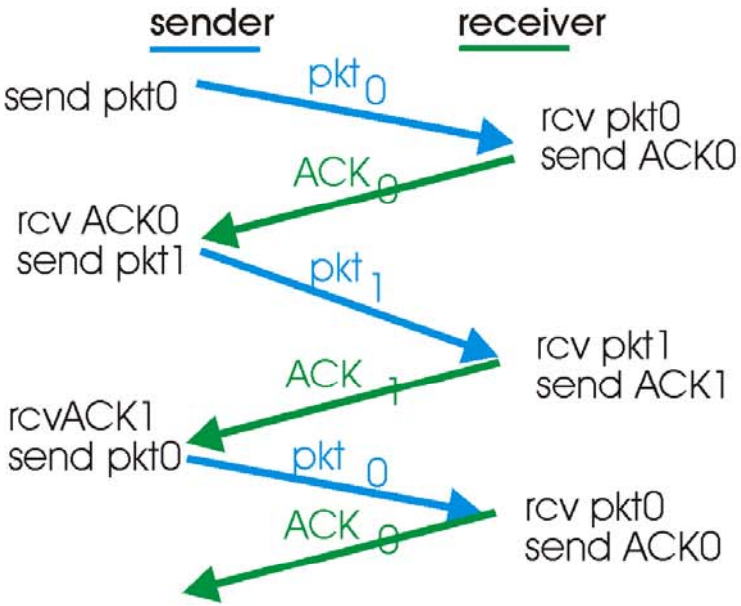
等待一段"合理的"時間

- 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

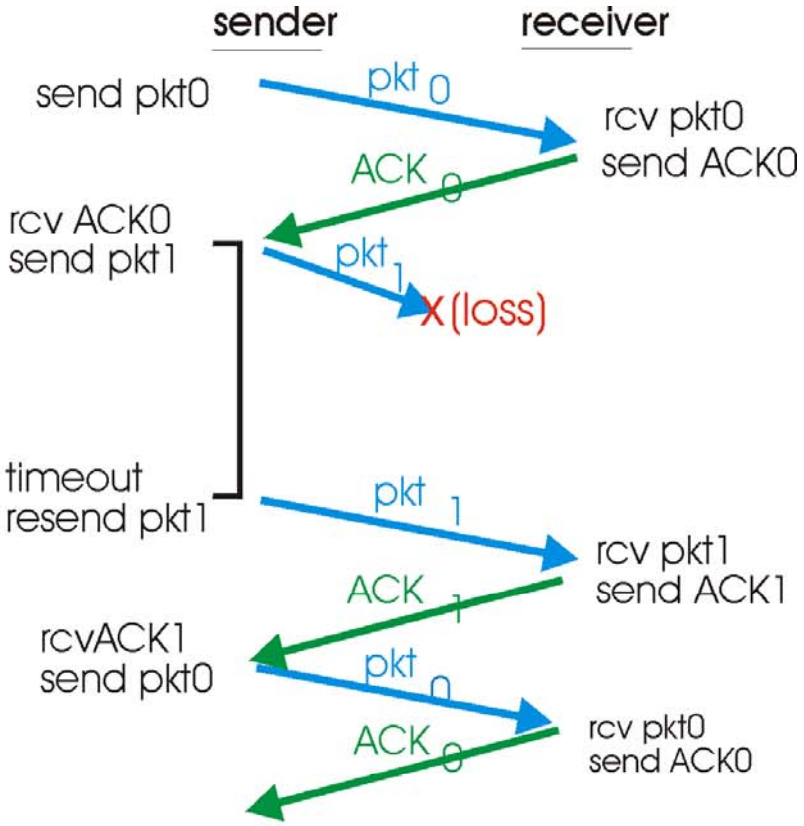
rdt3.0 sender



rdt3.0 in action

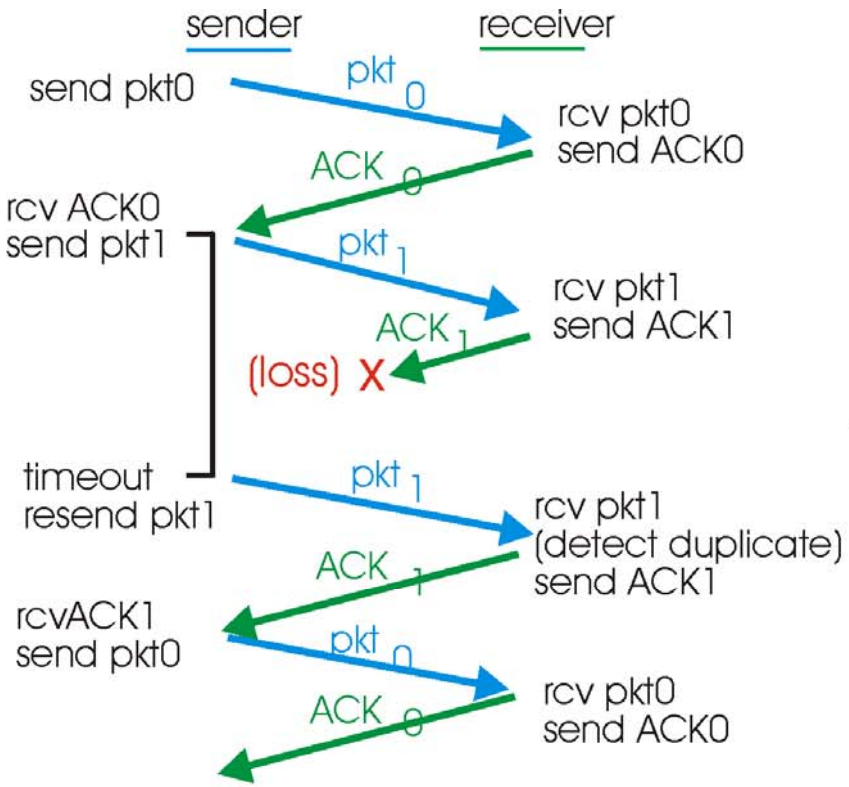


(a) operation with no loss

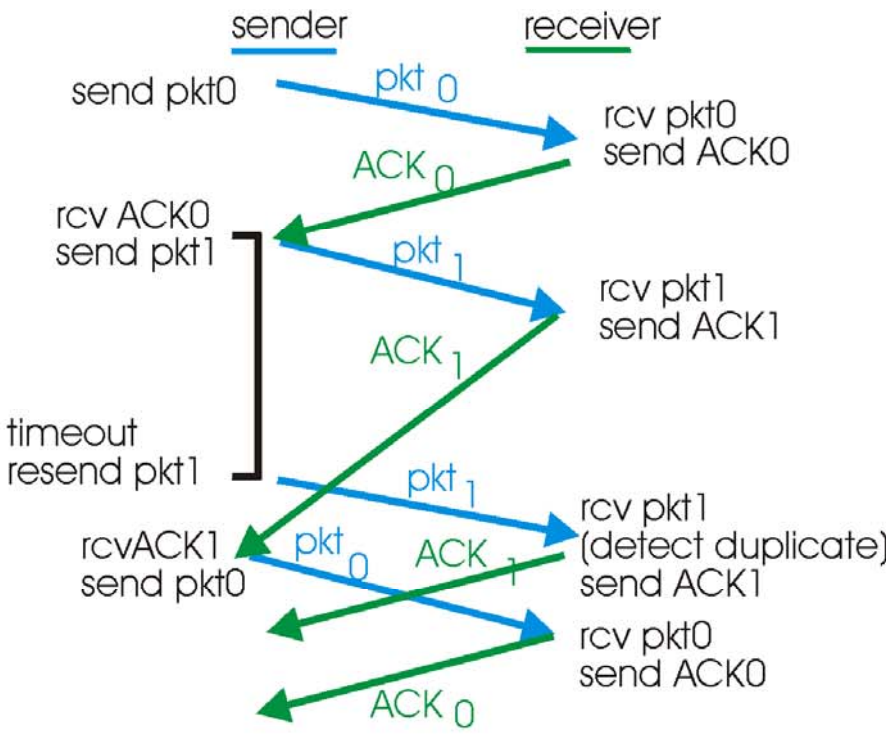


(b) lost packet

rdt3.0 in action



(c) lost ACK



(d) premature timeout

Performance of rdt3.0

- rdt3.0 works, but performance stinks
- example: 1 Gbps link, 15 ms e-e prop. delay, 1KB packet:

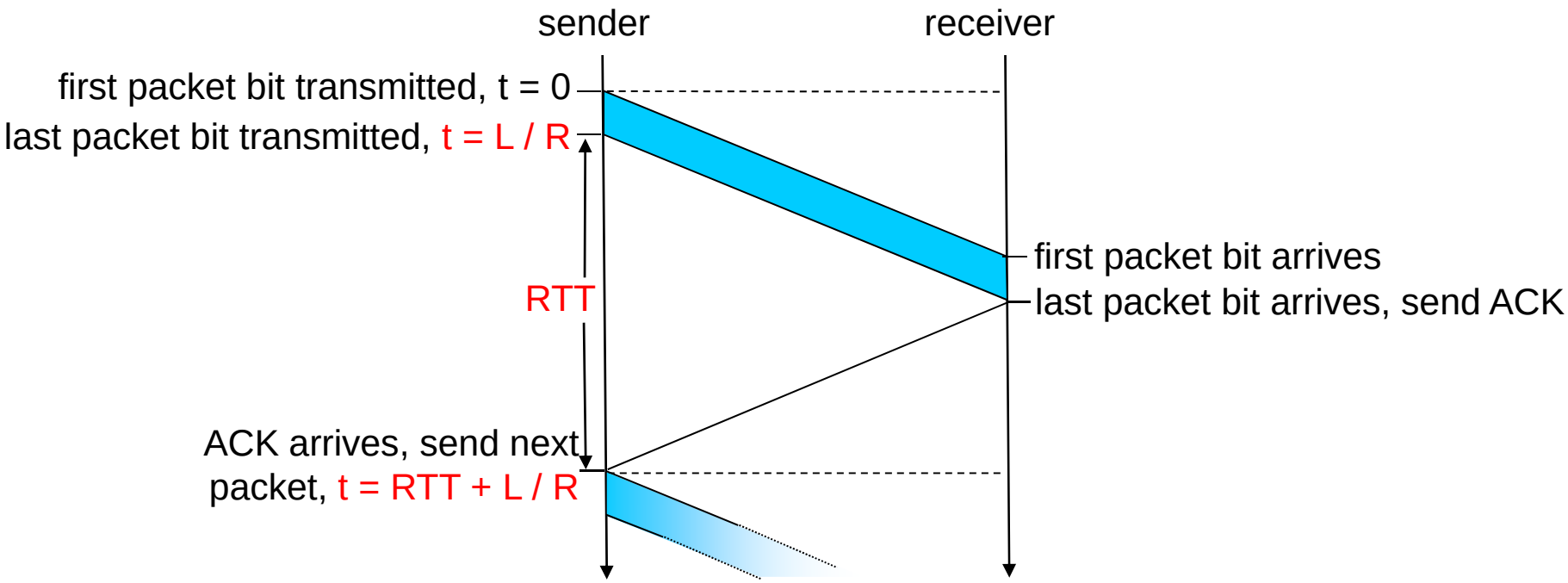
$$T_{transmit} = \frac{L \text{ (packet length in bits)}}{R \text{ (transmission rate, bps)}} = \frac{8kb/pkt}{10**9 \text{ b/sec}} = 8 \text{ microsec}$$

  - 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

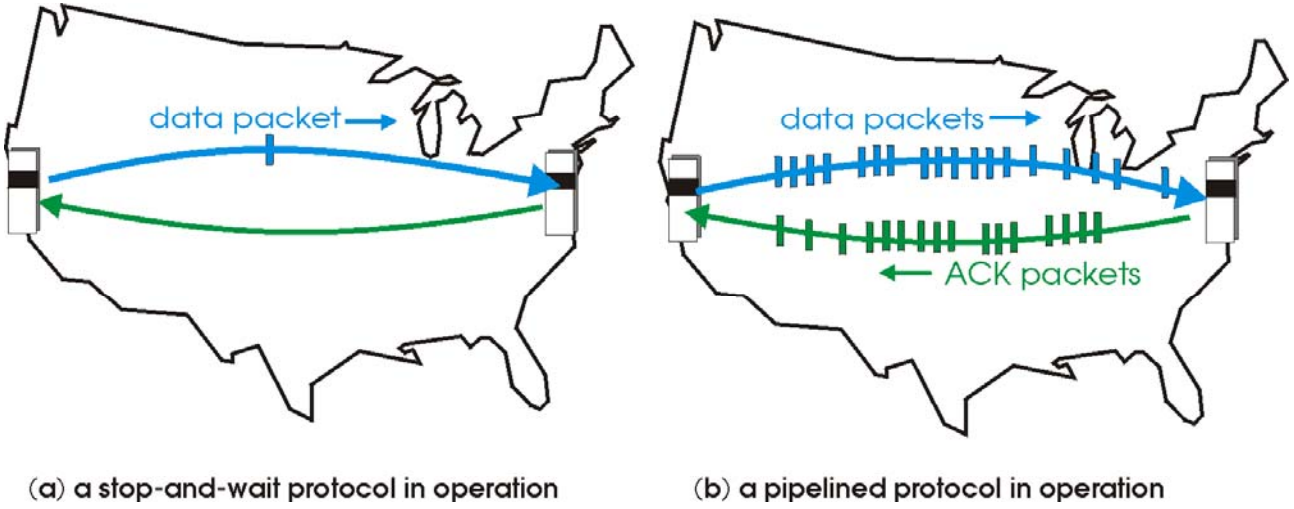


$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

Pipelined protocols (平行)管線傳送

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

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

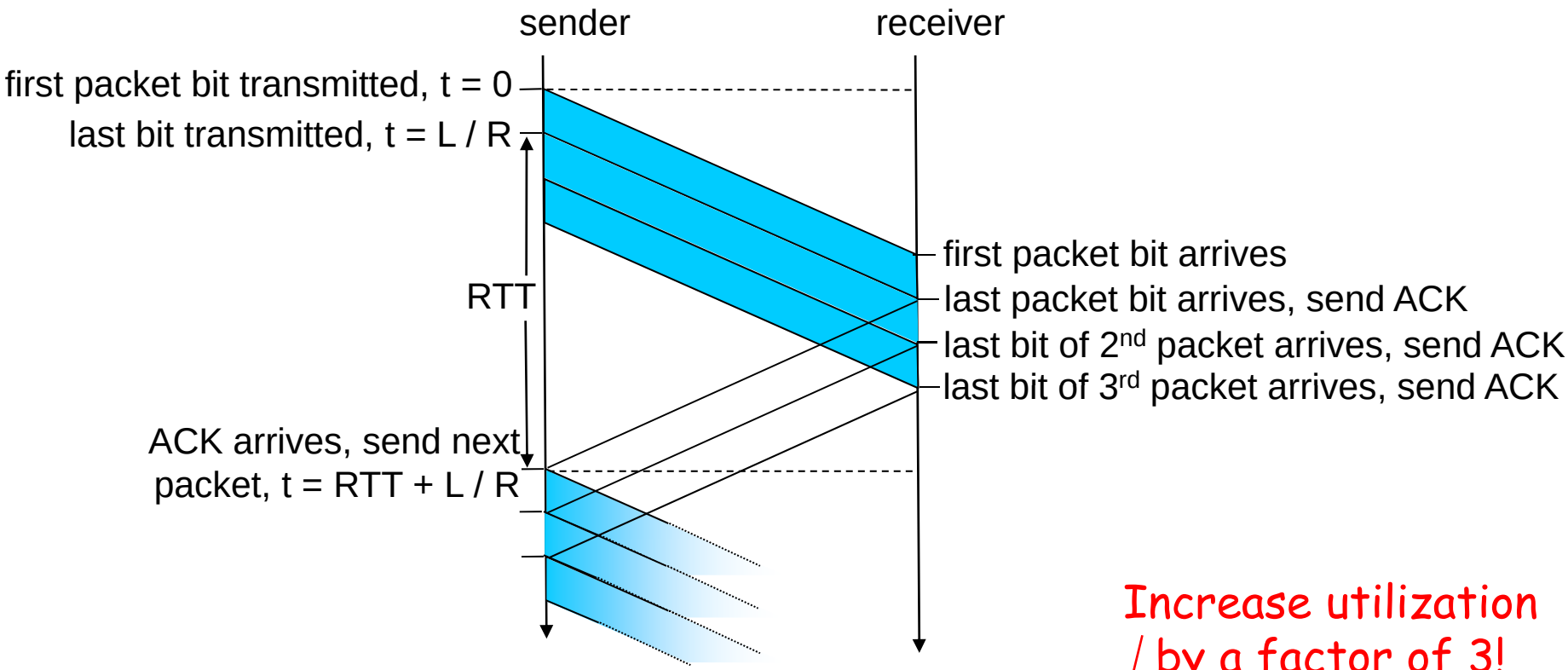


(a) a stop-and-wait protocol in operation

(b) a pipelined protocol in operation

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

Pipelining: increased utilization



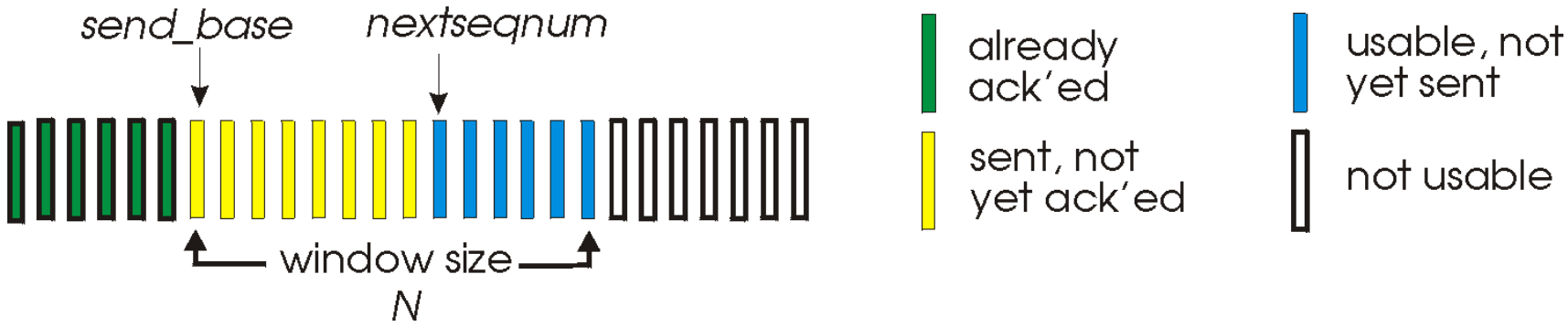
Increase utilization by a factor of 3!

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

Go-Back-N

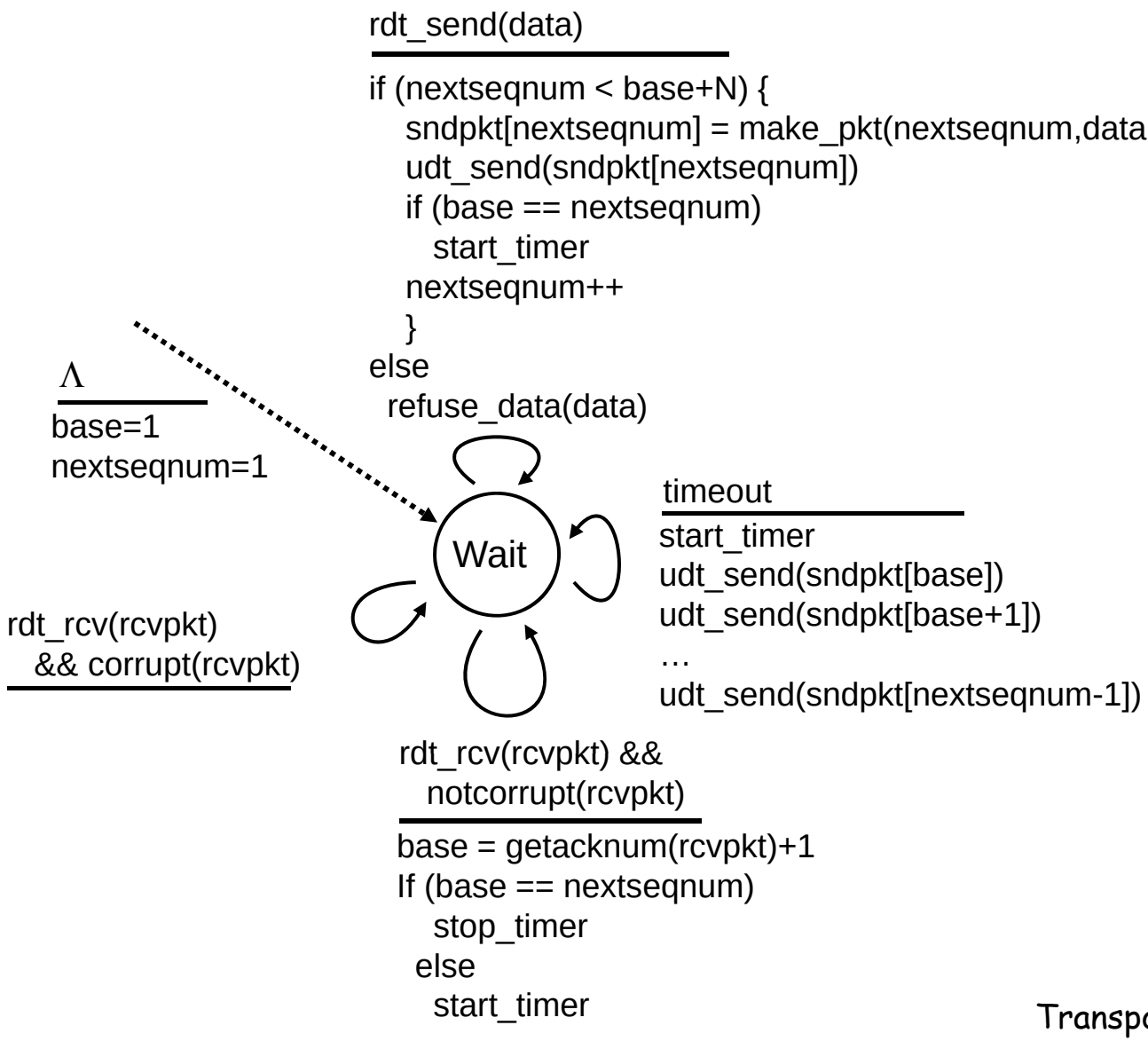
Sender:

- k-bit seq # in pkt header (用k-bit來表示序號)
- "window" of up to N, consecutive unack'ed pkts allowed

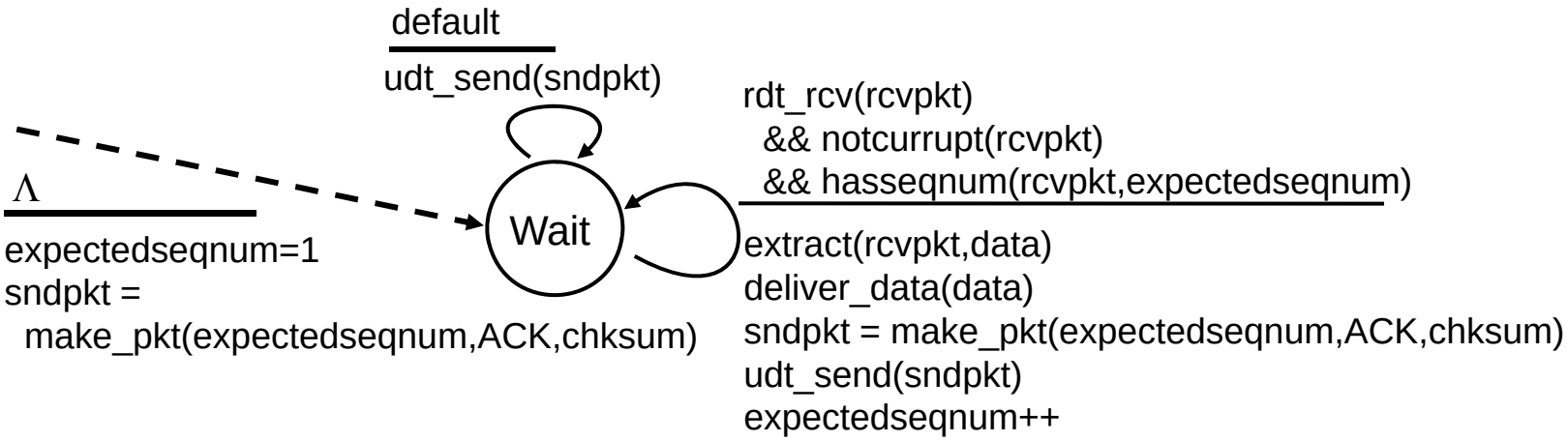


- ACK(n): ACKs all pkts up to, including seq # n - "cumulative ACK"
  - may receive duplicate ACKs (see receiver)
- timer for in-flight pkt 設重傳時間
- *timeout(n):* retransmit pkt n and all higher seq # pkts in window
  重傳沒收到ACK的封包，以及seq#較大的封包

GBN: sender extended FSM



GBN: receiver extended FSM

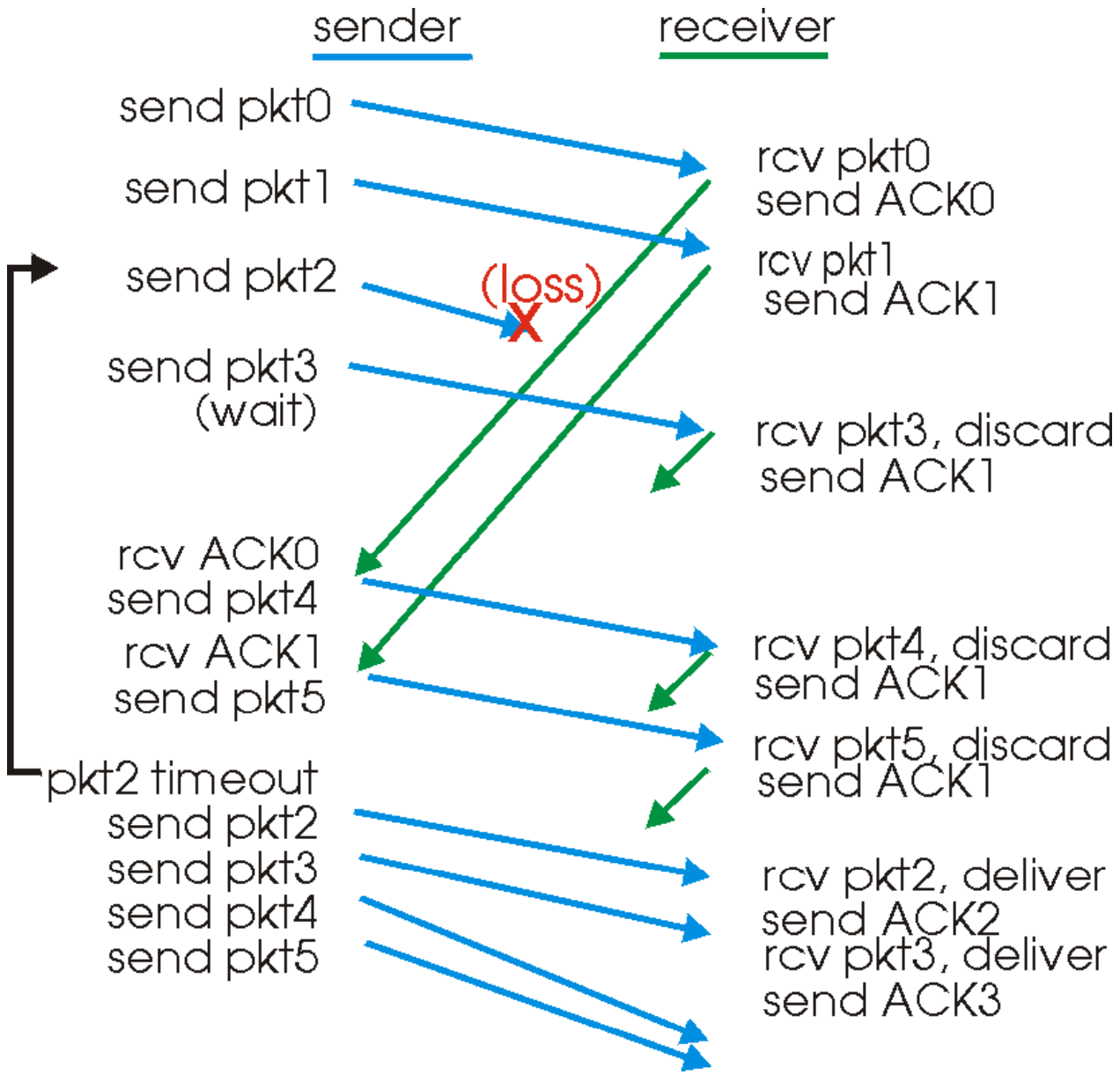


ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq # (ACK最大且依序到達的封包)

- may generate duplicate ACKs
- need only remember **expectedseqnum**
  只要記得下一個要收的封包序號即可

- out-of-order pkt: 不照順序抵達的封包處理方法
  - discard (don't buffer) -> no receiver buffering! 捨棄
  - Re-ACK pkt with highest in-order seq #

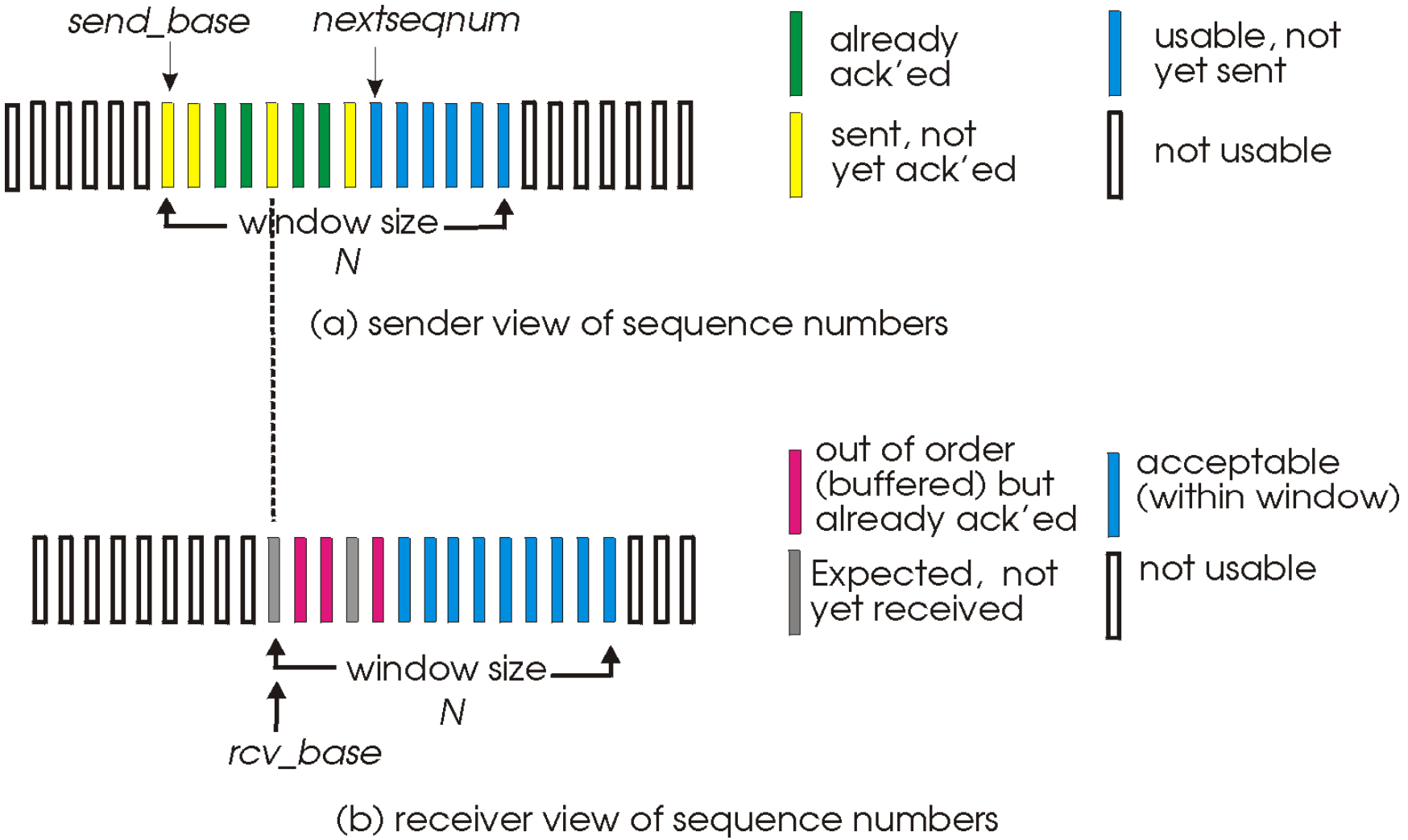
GBN in action



Selective Repeat

- receiver *individually* acknowledges all correctly received pkts 各自ACK正確收到的封包
  - buffers pkts, as needed, for eventual in-order delivery to upper layer 有buffer可用
- sender only resends pkts for which ACK not received 只要送沒有ACK的封包
  - 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
  如果下一個可送的序號在window內，就送出

timeout(n):

- resend pkt n, restart timer
  重新傳送packet, 重設計時器

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
  移動window

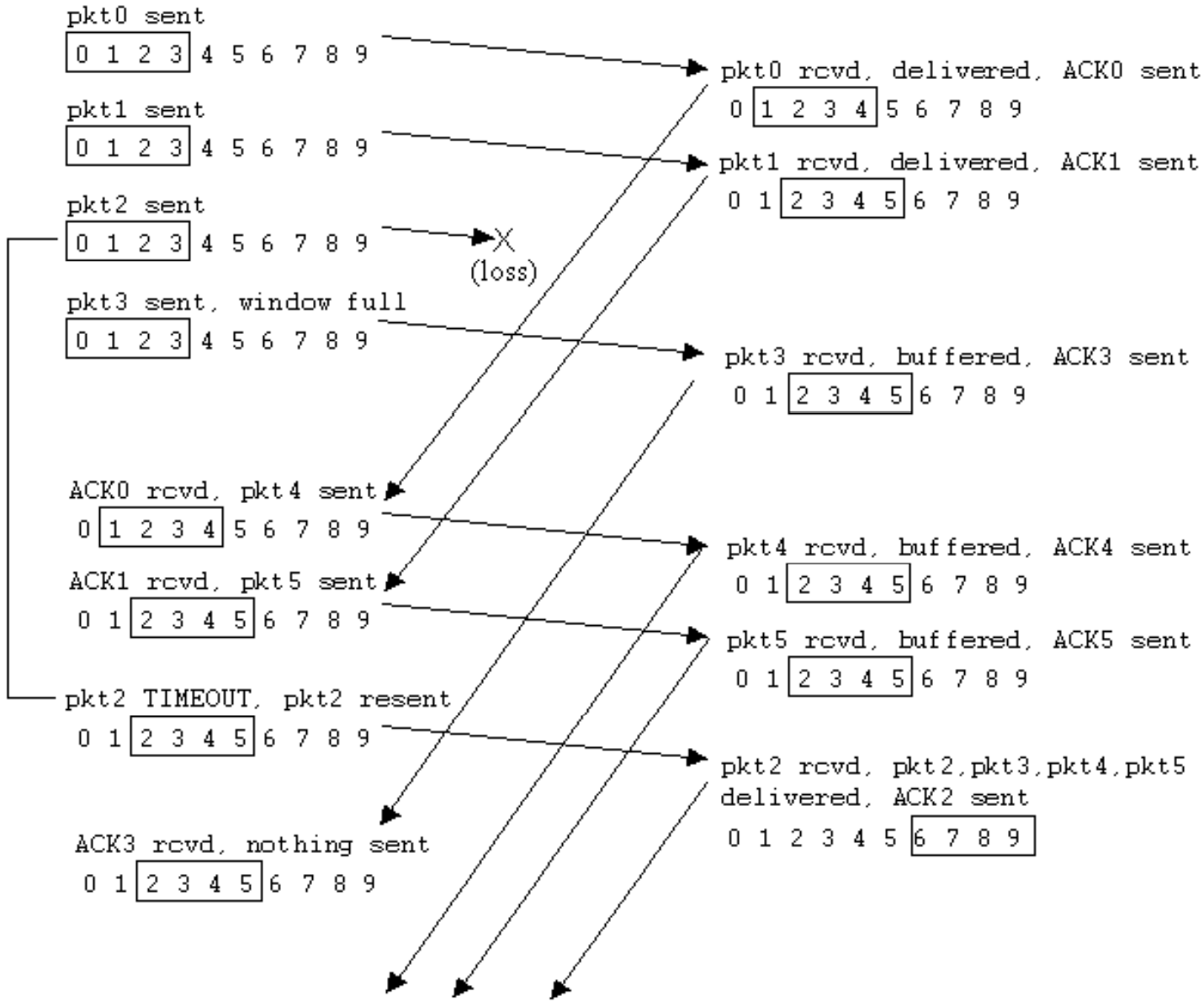
pkt n in [rcvbase-N,rcvbase-1]

- ACK(n)

otherwise:

- ignore

Selective repeat in action



Selective repeat: dilemma 問題來了

- 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? (Homework)