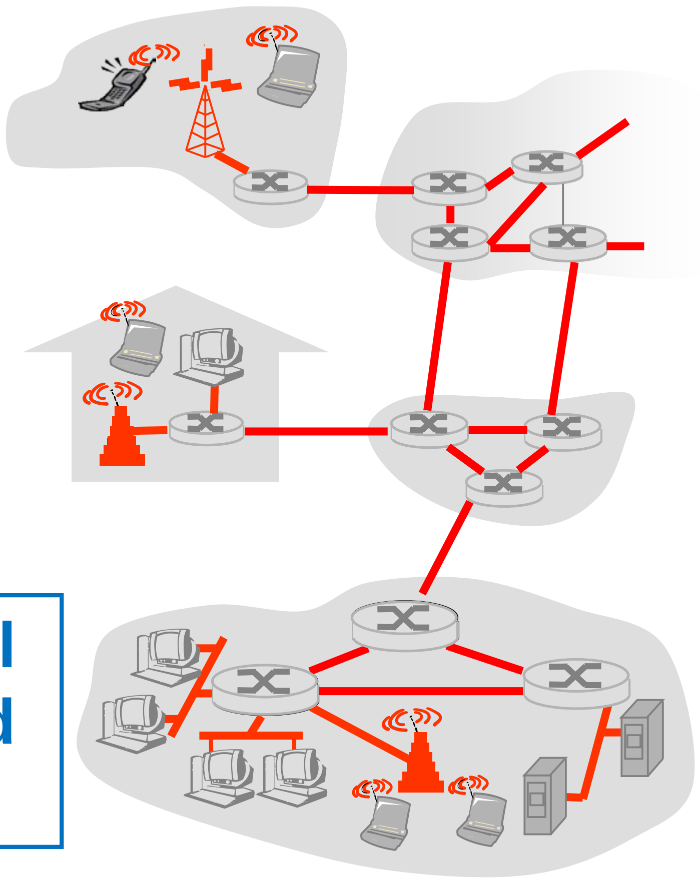
Direct Connection Networks

Acknowledgements
These Slides have been adapted from the originals made available by J. Kurose and K. Ross

Overview

Some terminology:

- hosts and routers are **nodes**
- communication channels that connect adjacent nodes along communication path are **links**
  - wired/wireless links
  - Point-to-point/shared links



In this part of the course we will look at how data are transferred between adjacent nodes

Goals

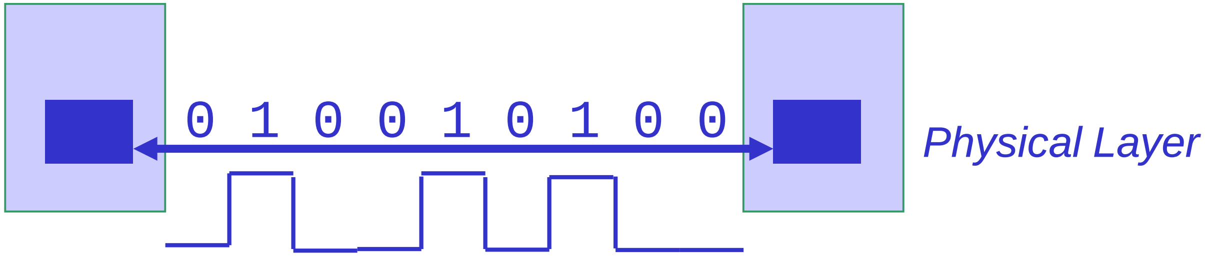
- Introducing direct connection networks
- understanding principles behind Data Link layer services:
  - reliable data transfer
    - error detection, correction
    - Acknowledgement, timeout, and re-transmission
  - flow control
  - sharing a broadcast channel
    - multiple access
    - link layer addressing
- instantiation and implementation of various link layer technologies

Direct Connection Networks

- Introduction
- Error detection and correction
- Reliable Data Transfer
- PPP
- Multiple access protocols
- Local Area Networks (LAN)
- Ethernet

Introduction

Physical Link

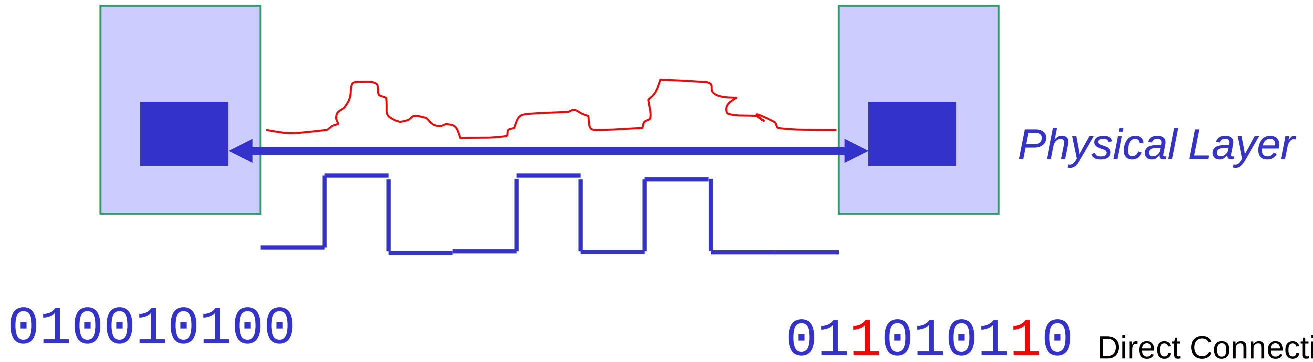


Encoding

Bits are coded through an electric/electro-magnetic/light signal and send over the physical link

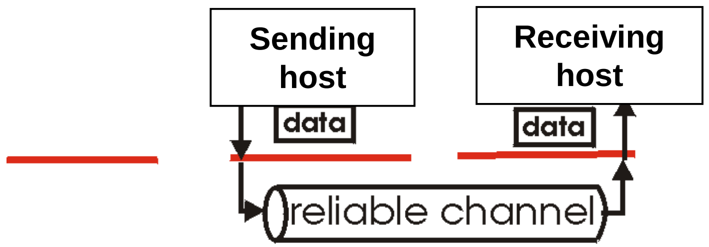
Real-life Problems

- The transmission channel is not ideal
  - Signal attenuation
  - Noise
    - Interferences, fading, …
- The received data sequence may be different from the transmitted one



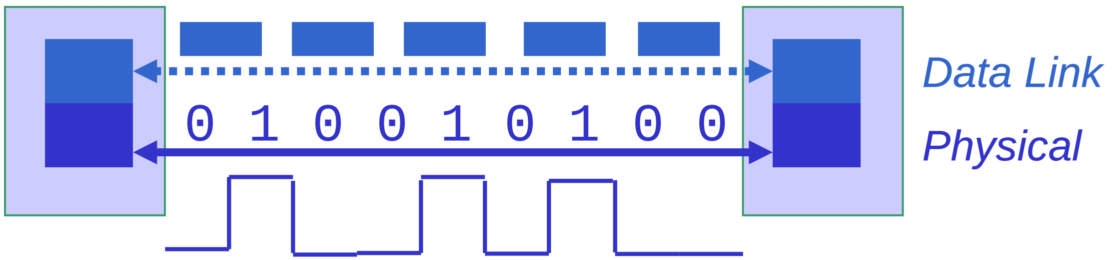
Data Link Layer

- Reliable delivery between adjacent nodes



Data-Link Layer Services

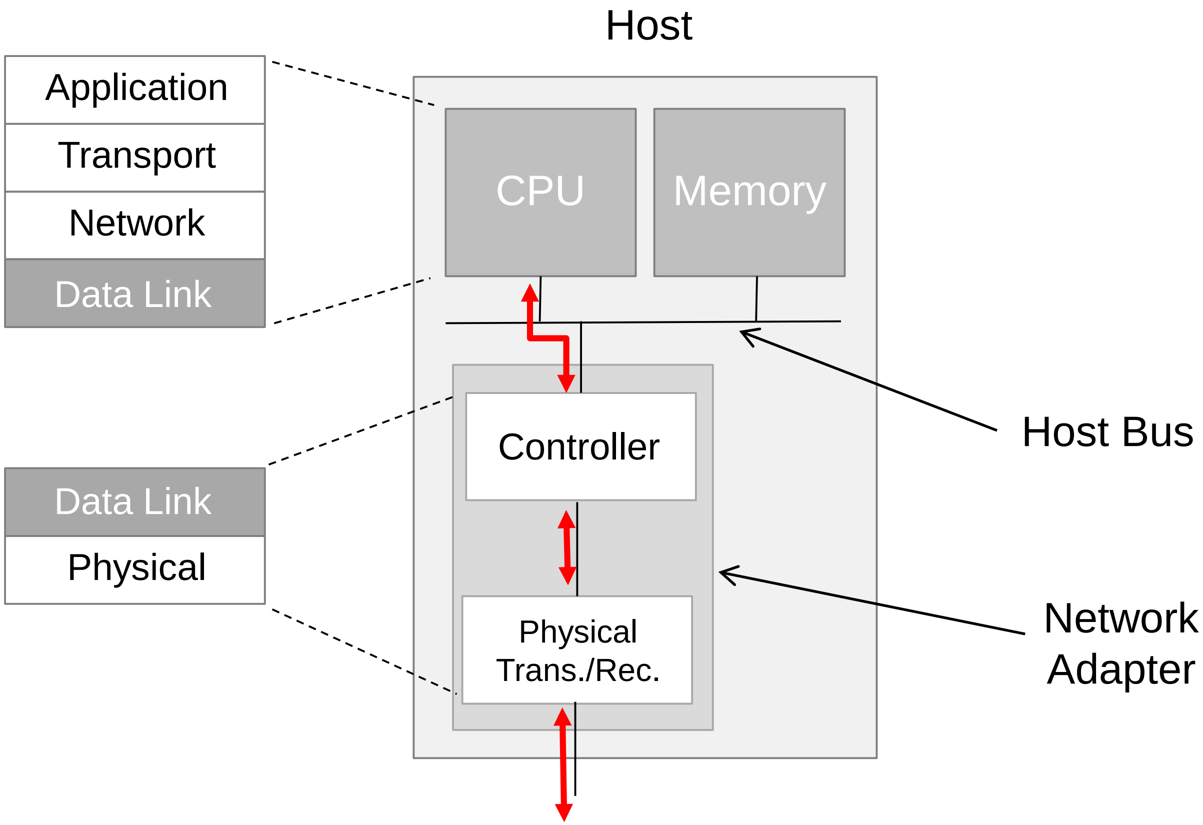
- Framing
  - encapsulate datagram into frame, adding header, trailer



Data-Link Layer Services

- Error detection:
  - receiver detects presence of errors
- Error correction:
  - receiver identifies *and corrects* bit error(s)
- Reliable Data Transfer
  - Through acknowledgements and retransmissions
- Flow control:
  - pacing between adjacent sending and receiving nodes
- Half-duplex and full-duplex
  - with half duplex, nodes at both ends of link can transmit, but not at same time

Where is the link layer implemented?



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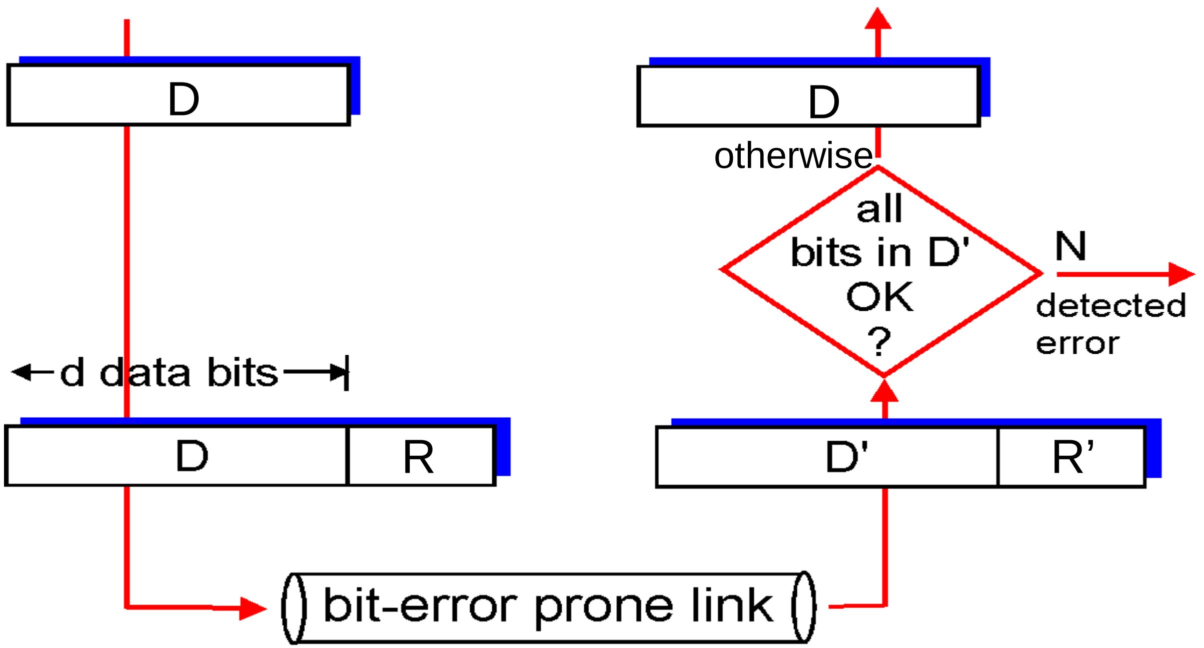
Error Detection

R= Redundancy bits
D= Data protected by error checking, may include header fields

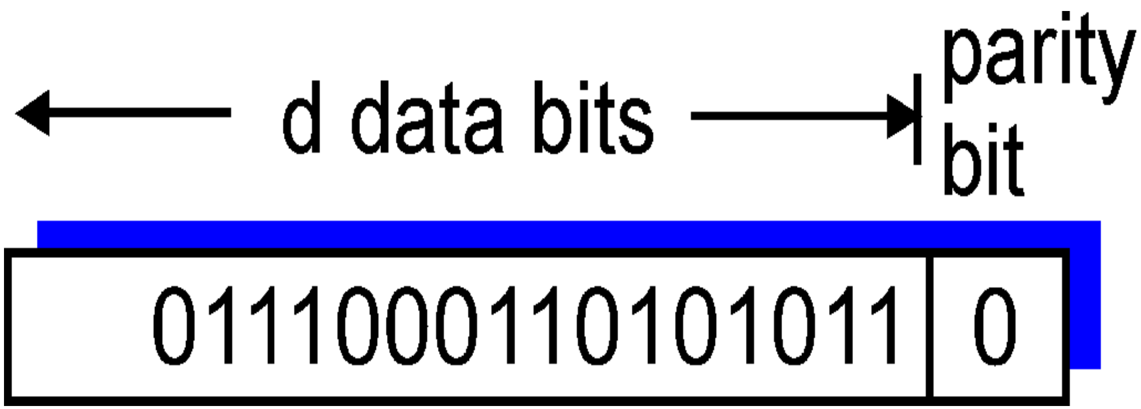
- Error detection not 100% reliable!
  - protocol may miss some errors, but rarely
  - larger R field yields better detection



Parity Checking

Single Bit Parity:

Detect single bit errors



Checksum

<u>Goal:</u> detect "errors" (e.g., flipped bits) in transmitted packet (note: used at transport layer *only*)

<u>Sender:</u>

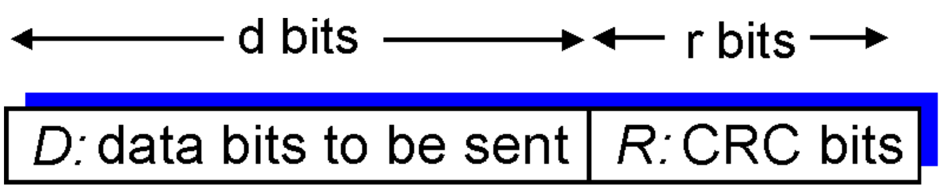
- treat segment contents as sequence of 16-bit integers
- checksum: addition (1's complement sum) of segment contents
- sender puts checksum value into packet checksum field

<u>Receiver:</u>

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

Cyclic Redundancy Check (CRC)

- widely used in practice (Ethernet, 802.11 WiFi, …)
- view data bits, D, as a binary number
- choose r+1 bit pattern G (Generator)
- goal: choose r CRC bits, R, such that
  - <D,R> exactly divisible by G (using modulo-2 arithmetic)
  - receiver knows G, divides <D,R> by G.
  - If non-zero remainder: error detected!
- can detect all burst errors less than r+1 bits



bit pattern

$$D * 2^r \quad \text{XOR} \quad R$$

mathematical formula

CRC – How to derive R?

Want R such that:

$D \cdot 2^r$ XOR R = nG

equivalently:

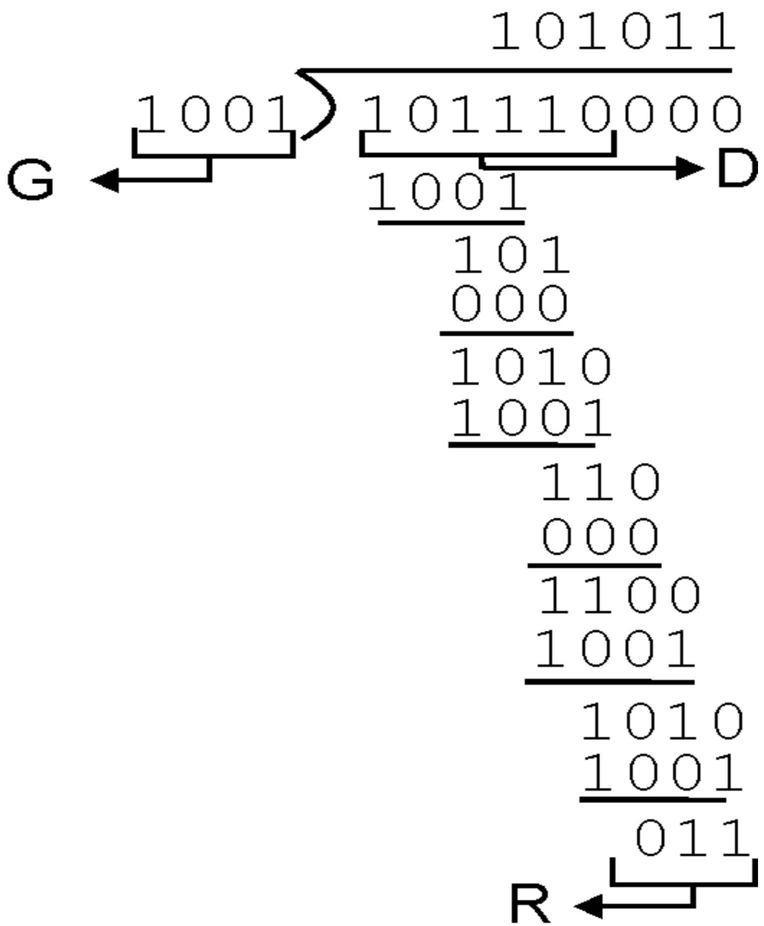
$D \cdot 2^r$ = nG XOR R

equivalently:

if we divide $D \cdot 2^r$ by G, the remainder is equal to R

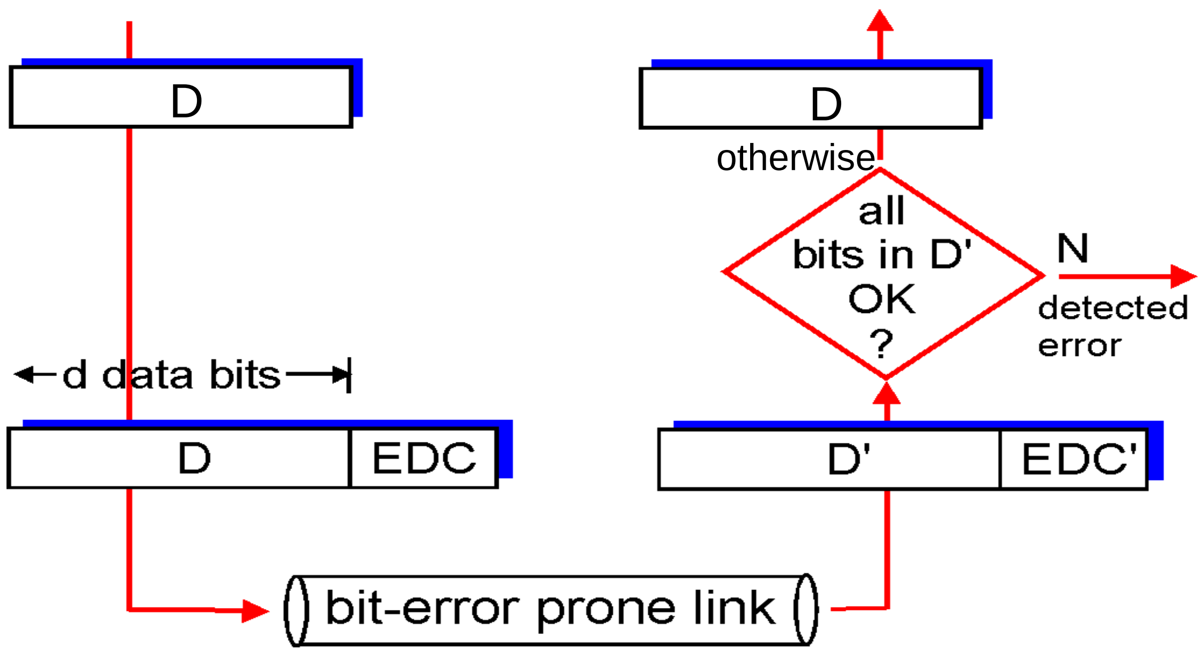
$$R = \text{remainder}\left[\frac{D \cdot 2^r}{G}\right]$$

Example



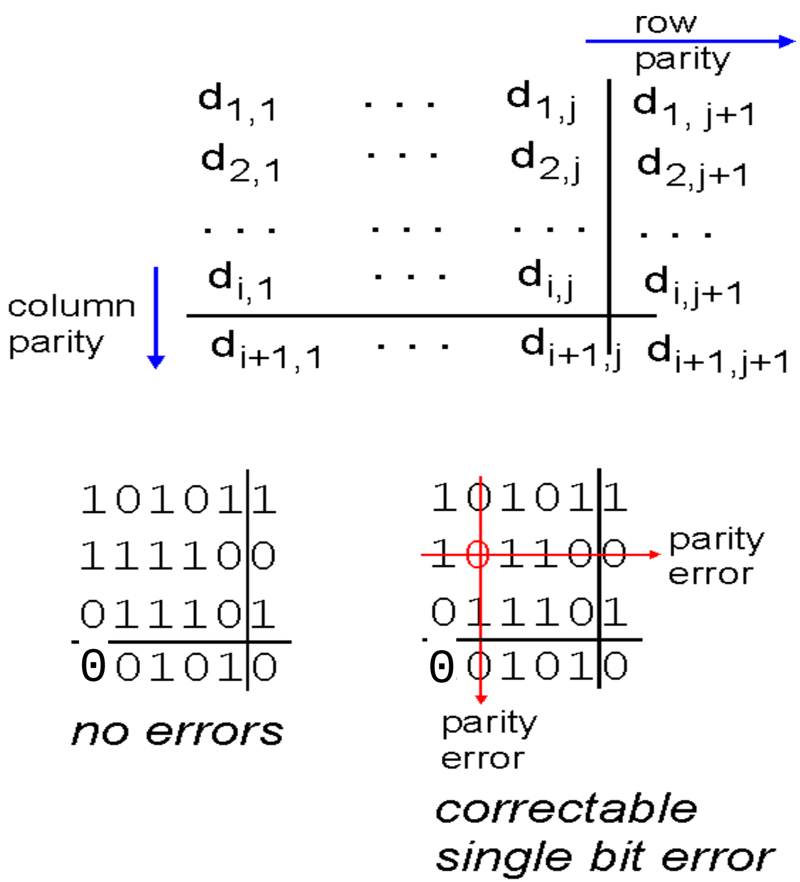
Forward Error Correction (FEC)

EDC	Error Detection and Correction (redundancy bits)
D	Data protected by error checking, may include header fields



Two-Dimensional Bit Parity

- Detects and correct single bit values



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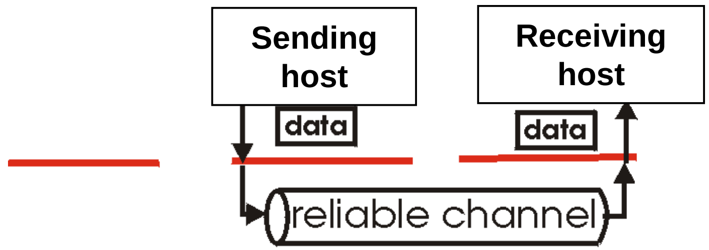
Principle of Reliable Data Transfer

- Important in Data Link, Transport, and Application layers
- Top-10 list of important networking topics!

What we would like to get



Reliable data transfer: overview

rdt_send(): *called from above, (e.g., by network). 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*

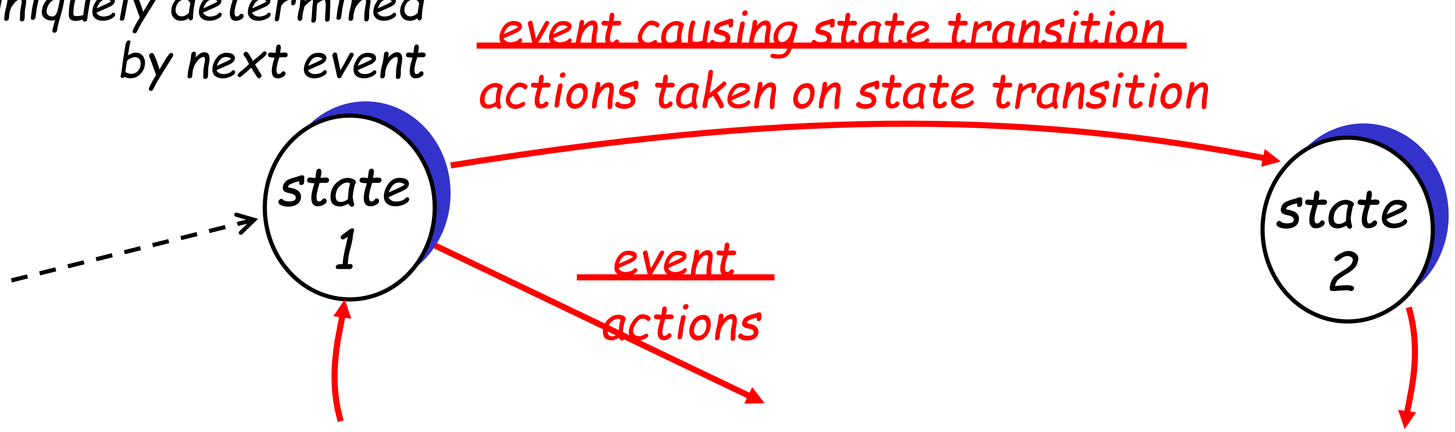
rdt_rcv(): *called when packet arrives on rcv-side of channel*

Reliable data transfer: getting started

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

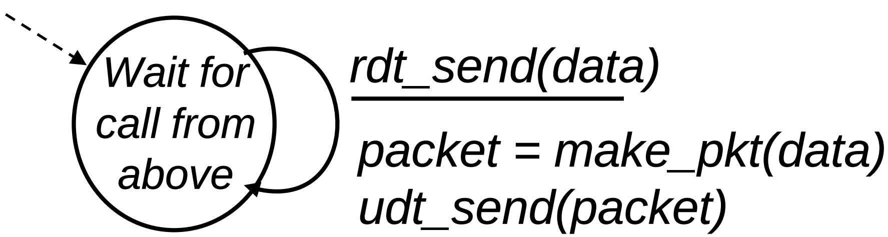
Reliable Data Transfer: FSM

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

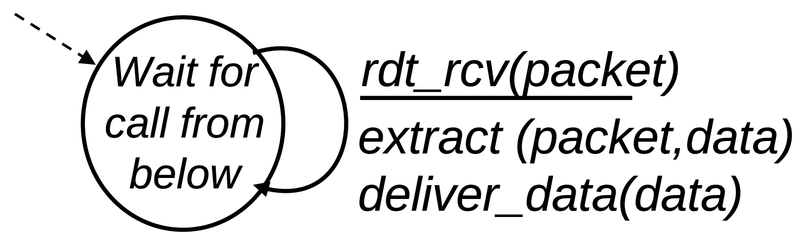


Rdt1.0: reliable transfer over a reliable channel

- 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

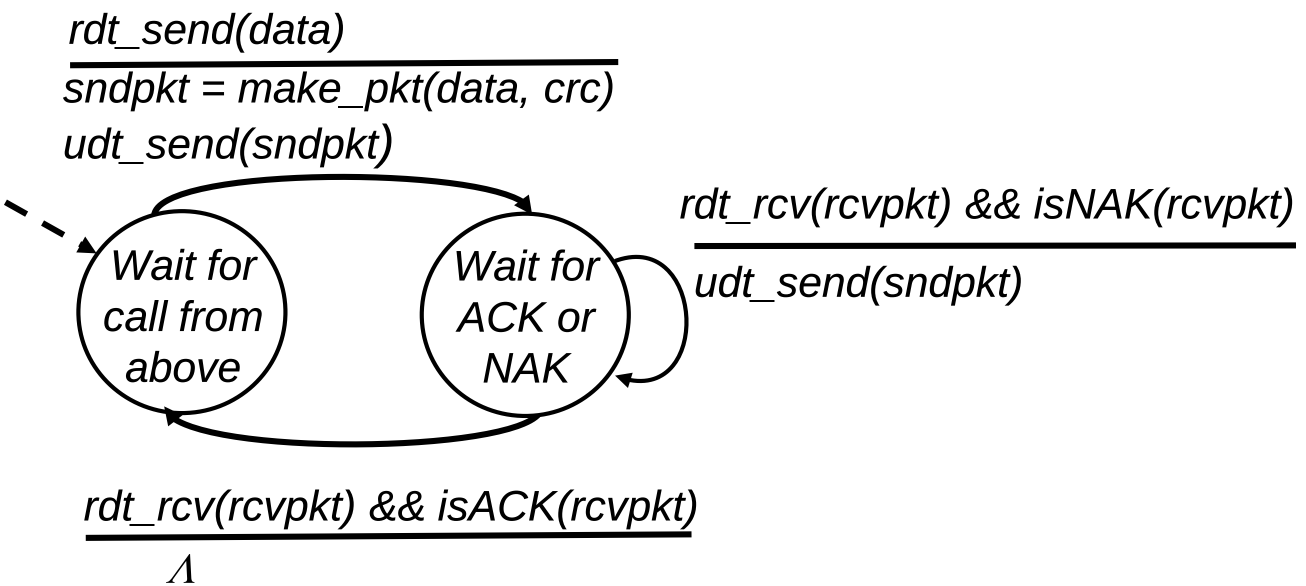


receiver

Rdt2.0: channel with bit errors

- Underlying channel may flip bits in packet
  - Error detection to detect bit errors
    - CRC (Data Link layer)
    - Checksum (Transport layer)
- 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
- Automatic Repeat reQuest (ARQ) protocol
  - error detection (receiver side)
  - receiver feedback
    - control msgs (ACK,NAK) sent from receiver to sender
  - retransmission (sender side)

rdt2.0: FSM specification



sender

stop and wait
Sender sends one packet, then waits for receiver response

rdt2.0 has a fatal flaw!

What happens if ACK/NAK corrupted?

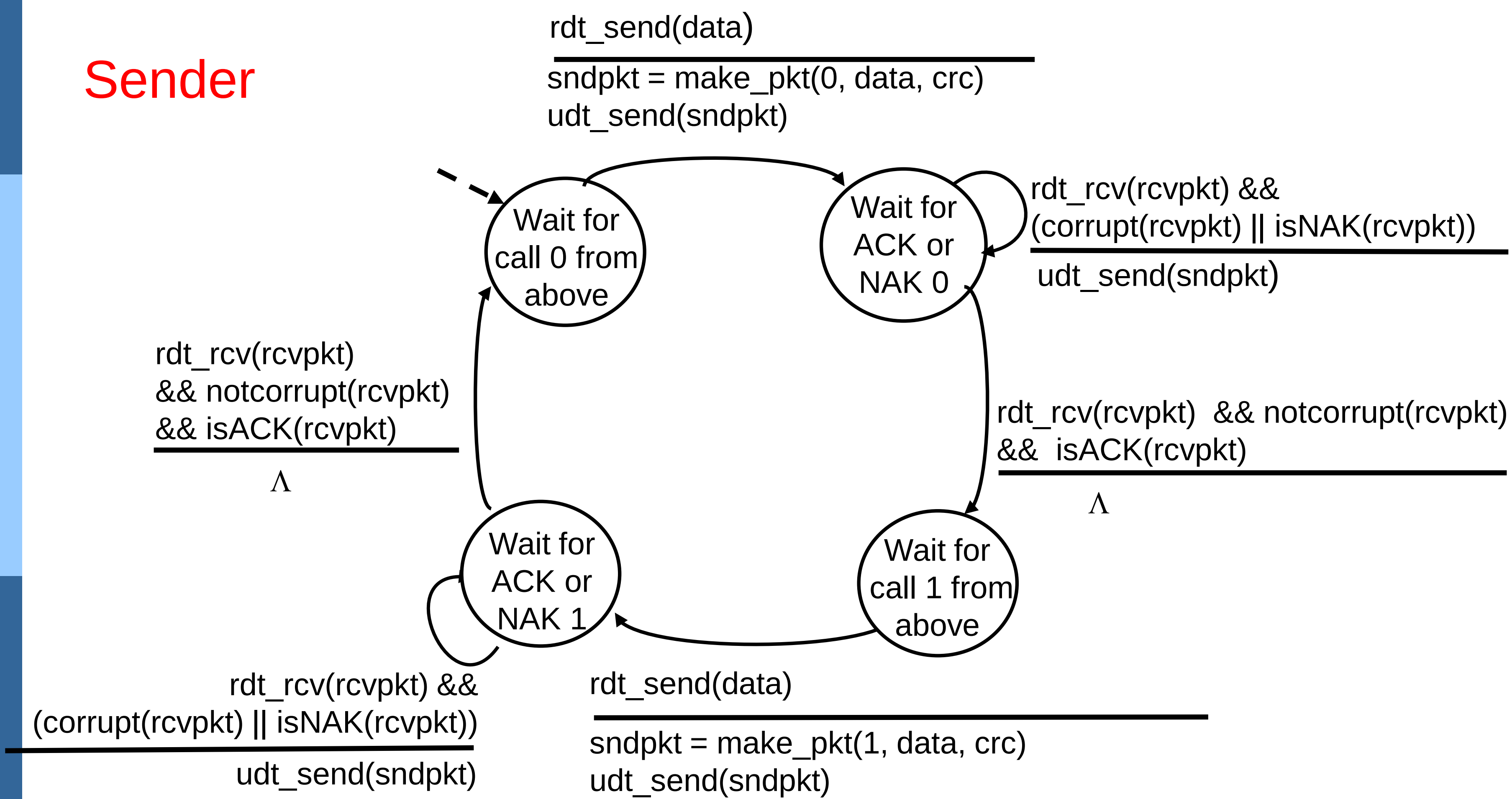
- sender doesn't know what happened at receiver!
- Error correction on ACKs/NAKs
  - Makes the channel error-free
  - Does not work on lossy channels where packets may get lost
- Re-transmission
  - sender retransmits current packet if ACK/NAK garbled
  - Possible Duplicates

Handling duplicates:

- sender adds *sequence number* to each packet
- receiver discards (doesn't deliver up) duplicate packets
- For a Stop-and-Wait protocol a 1-bit sequence number is enough

rdt2.1: Handling of garbled ACK/NAKs

Sender



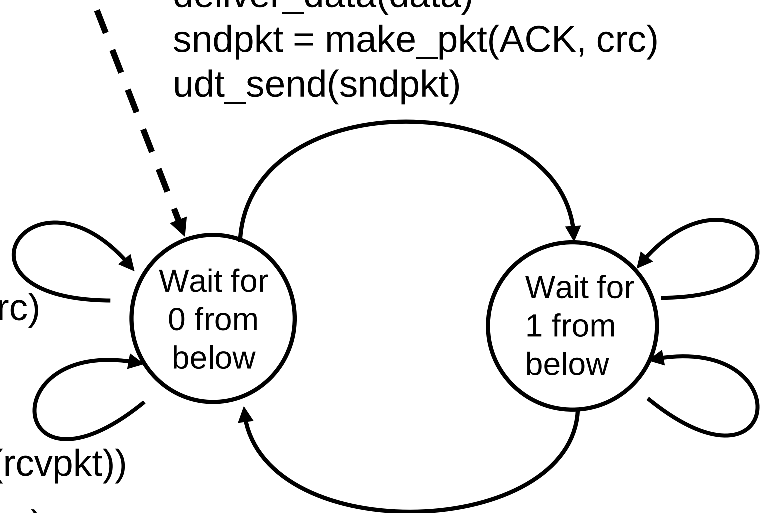
rdt2.1: Handling of garbled ACK/NAKs

Receiver

rdt_rcv(rcvpkt) && notcorrupt(rcvpkt) && has_seq0(rcvpkt)
___
extract(rcvpkt,data)
deliver_data(data)
sndpkt = make_pkt(ACK, crc)
udt_send(sndpkt)

rdt_rcv(rcvpkt)
&& not corrupt(rcvpkt)
&& has_seq1(rcvpkt)
___
sndpkt = make_pkt(ACK, crc)
udt_send(sndpkt)

rdt_rcv(rcvpkt) && (corrupt(rcvpkt))
___
sndpkt = make_pkt(NAK, crc)
udt_send(sndpkt)



rdt_rcv(rcvpkt) && (corrupt(rcvpkt)
___
sndpkt = make_pkt(NAK, crc)
udt_send(sndpkt)

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sndpkt = make_pkt(ACK, crc)
udt_send(sndpkt)

rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)
&& has_seq1(rcvpkt)
___
extract(rcvpkt,data)
deliver_data(data)
sndpkt = make_pkt(ACK, crc)
udt_send(sndpkt)

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. #

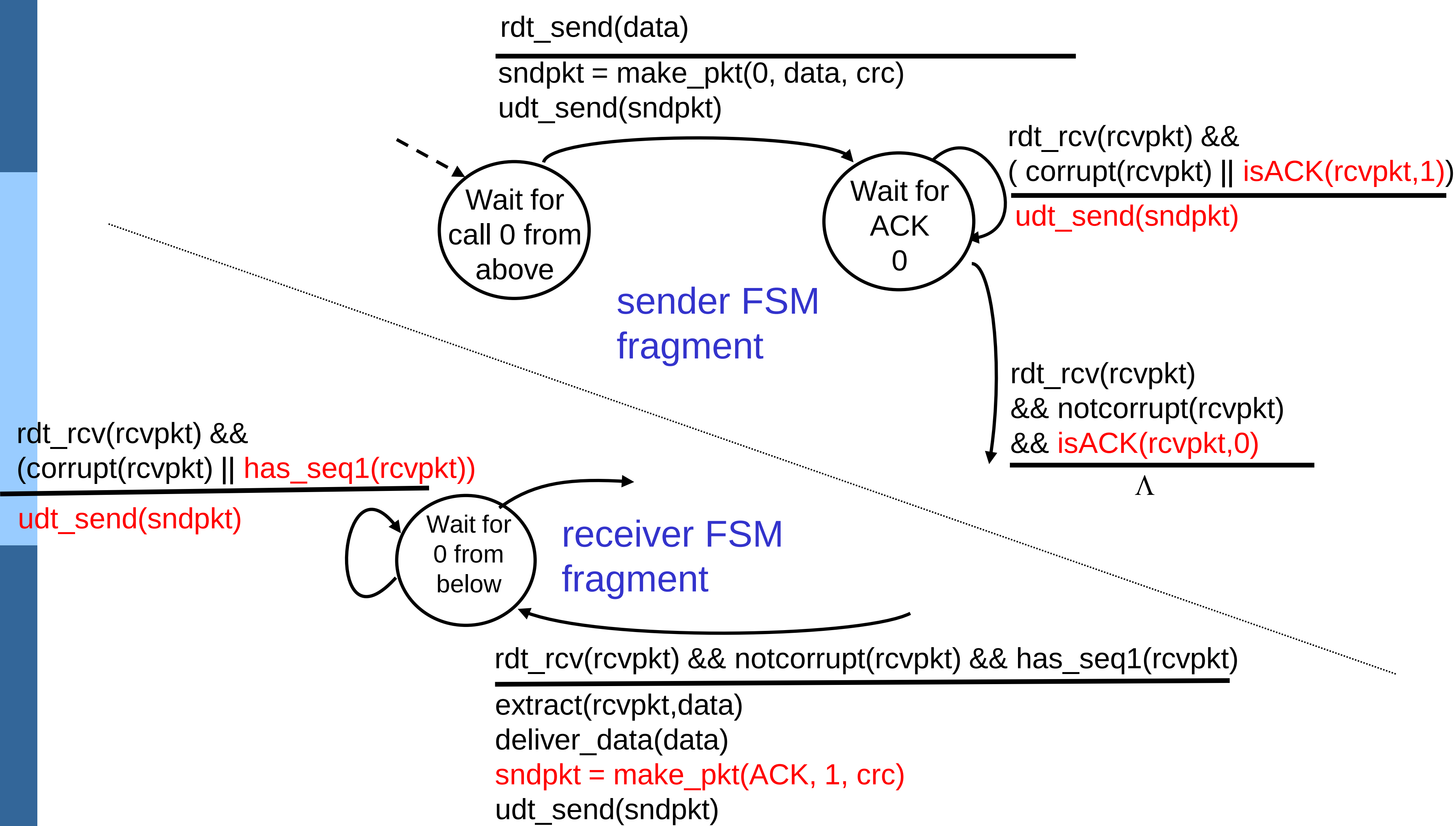
Receiver:

- 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 the last packet received OK
  - receiver must *explicitly* include the seq # of the packet being ACKed
- duplicate ACK at sender results in same action as NAK: *retransmit the current packet*

rdt2.2: sender, receiver FSMs



rdt3.0: channels with errors *and* loss

New assumption:

underlying channel can also lose packets (data or ACKs)

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

New Problem:

How to detect a packet loss?

Approach:

sender waits "reasonable" amount of time (time-out) for ACK

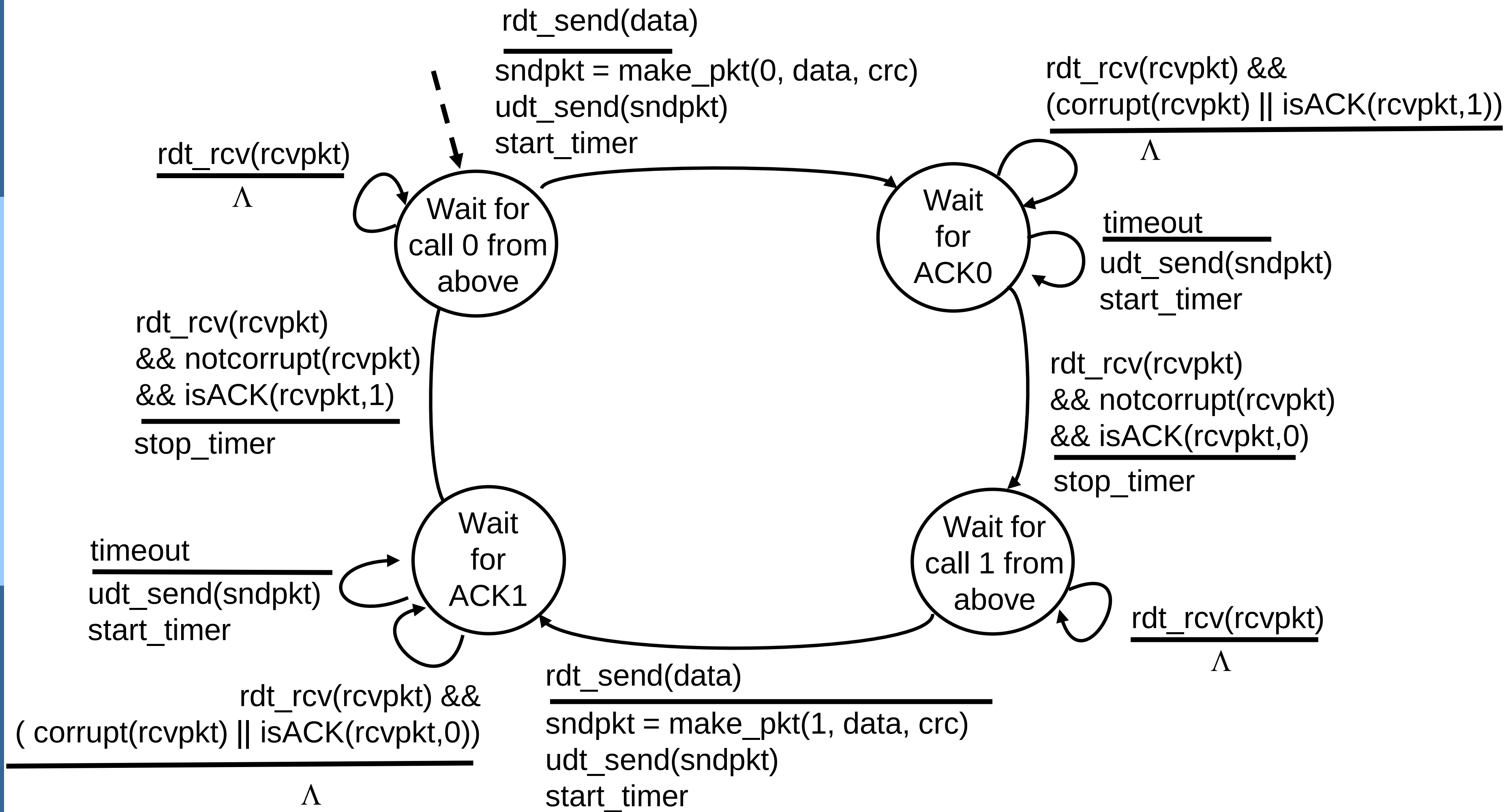
- requires a countdown timer at sender
- Sender retransmits if no ACK received in this time
- receiver must specify seq # of pkt being ACKed

rdt3.0: channels with errors *and* loss

How long to wait?

- If the time-out is too long
  - The data transfer process is made slower
- If the time-out is too short
  - if pkt (or ACK) just delayed (not lost), retransmission will produce duplicates at the receiver
    - but use of seq. #'s already handles this
- The time-out should be tailored to the Round Trip Time (RTT)

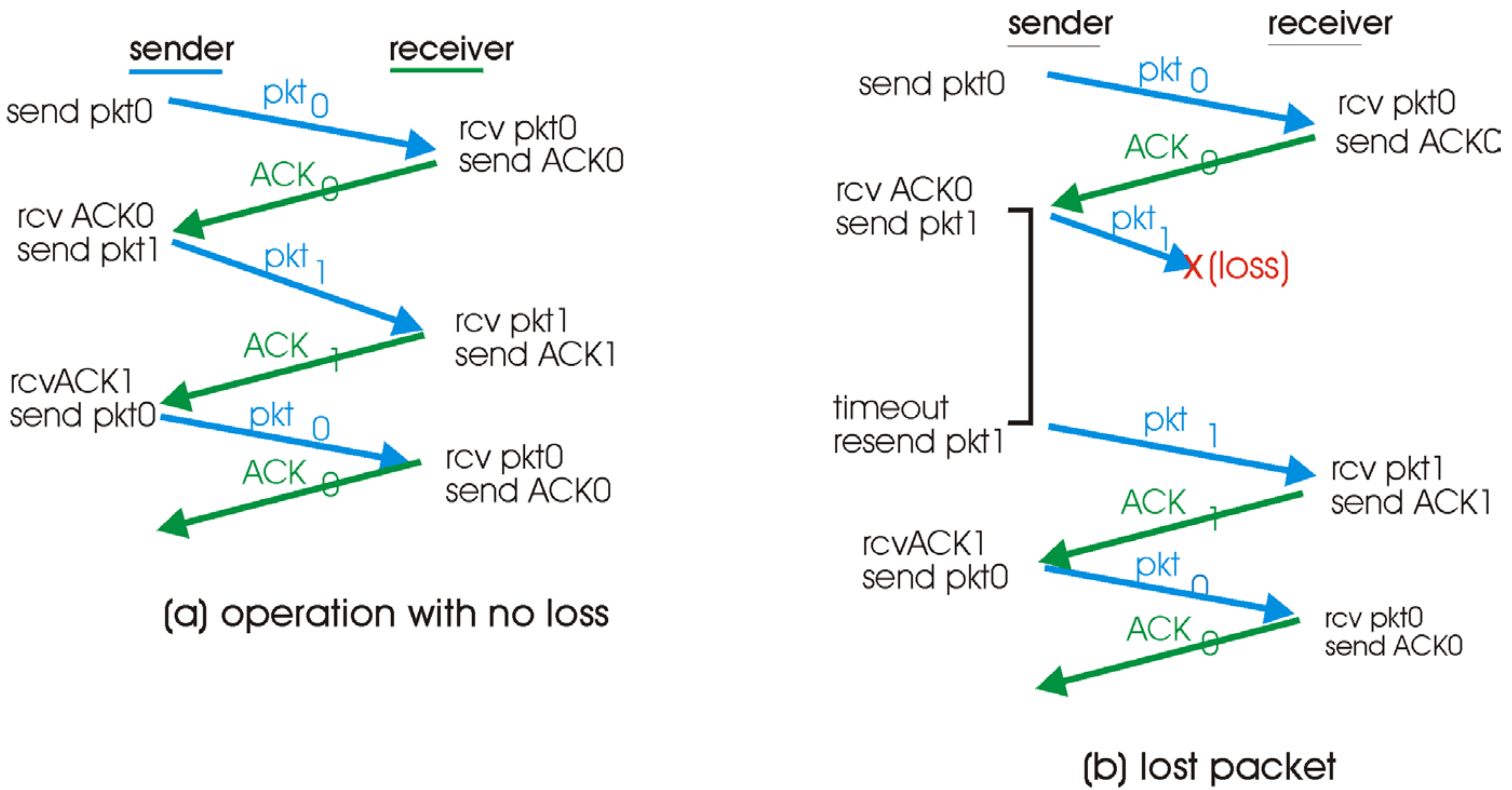
rdt3.0 sender



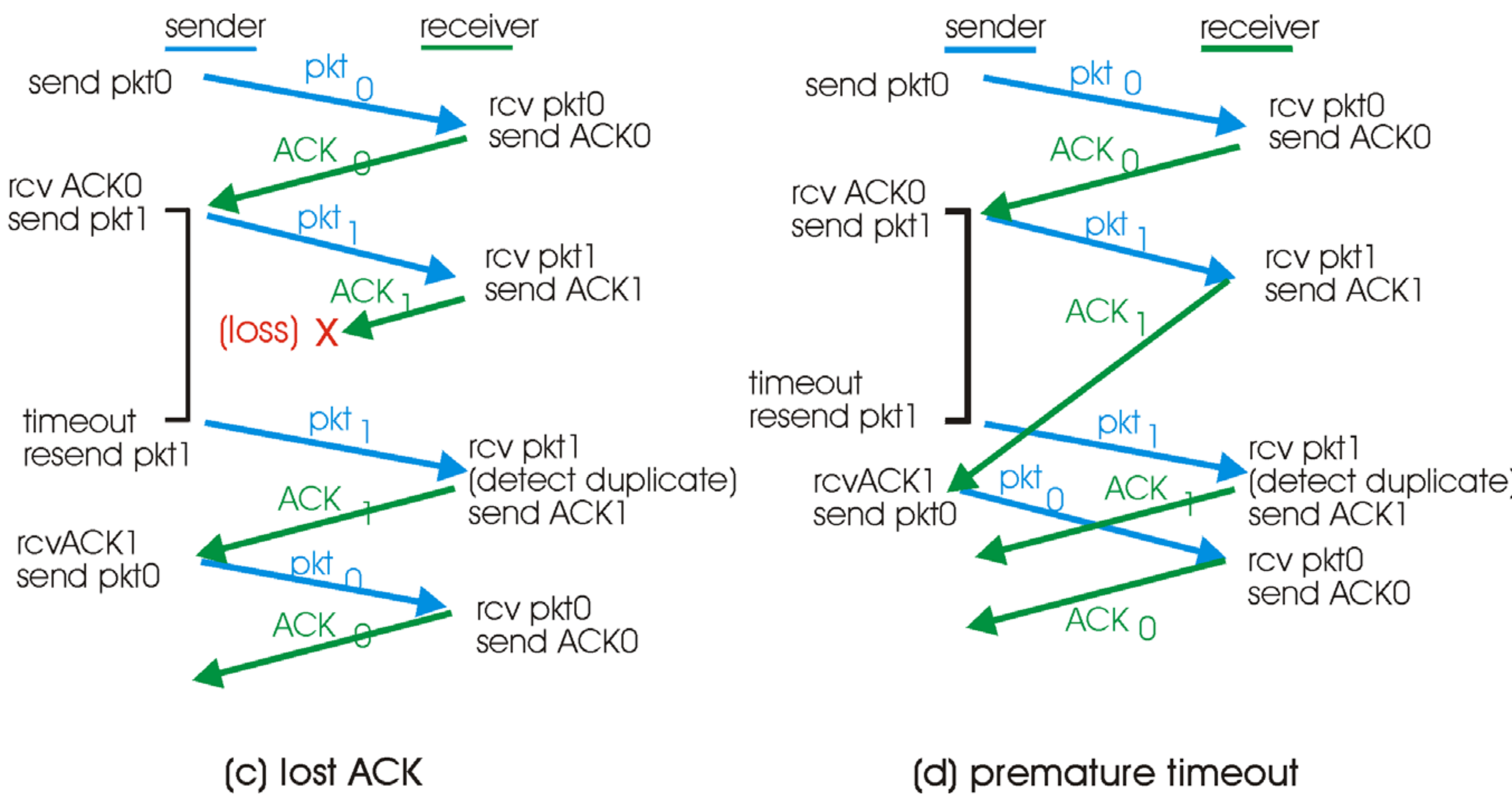
rdt3.0 receiver

- Left to students as a homework
- Define the receiver FSM
  - Like the sender FSM shown in the previous slide

rdt3.0 in action



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:

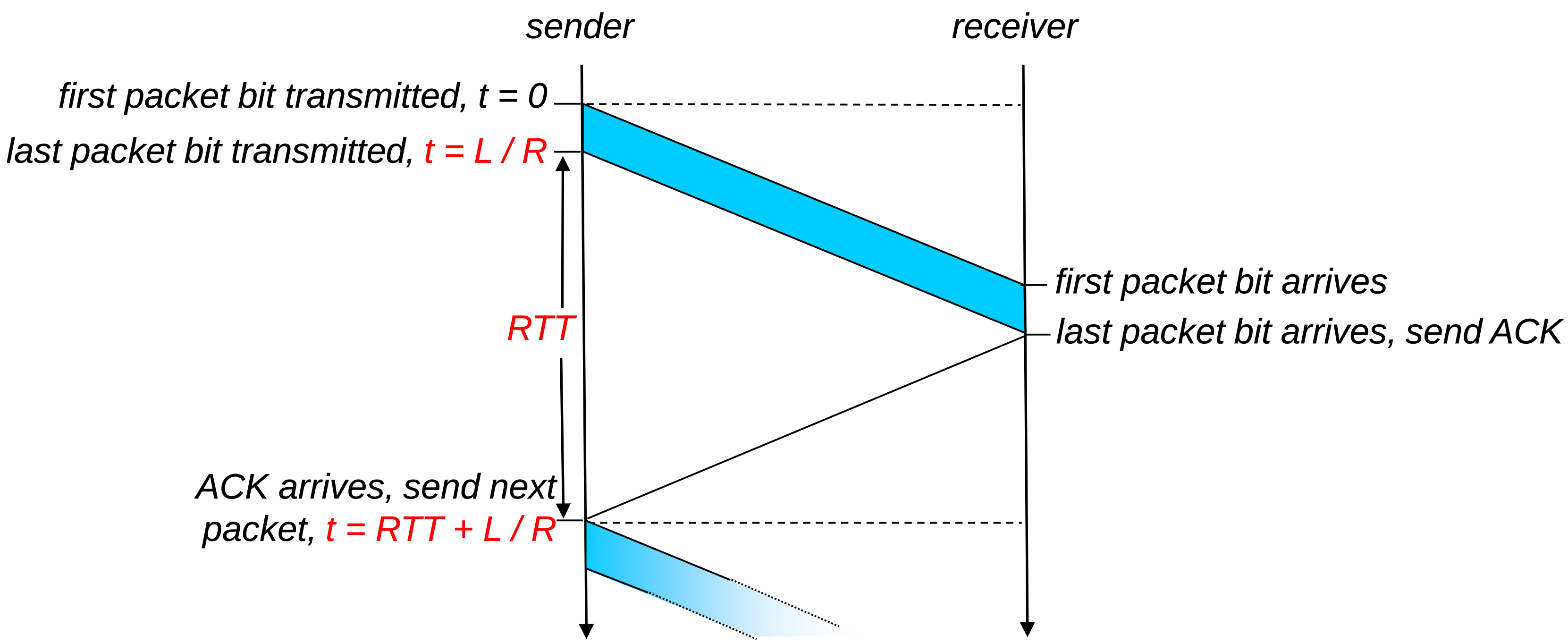
$$d_{trans} = \frac{L}{R} = \frac{8000\text{bits}}{10^9\,\text{bps}} = 8\,\text{microseconds}$$

  - *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 -> 267 Kbps throughput over 1 Gbps link*
  - *network protocol limits use of physical resources!*

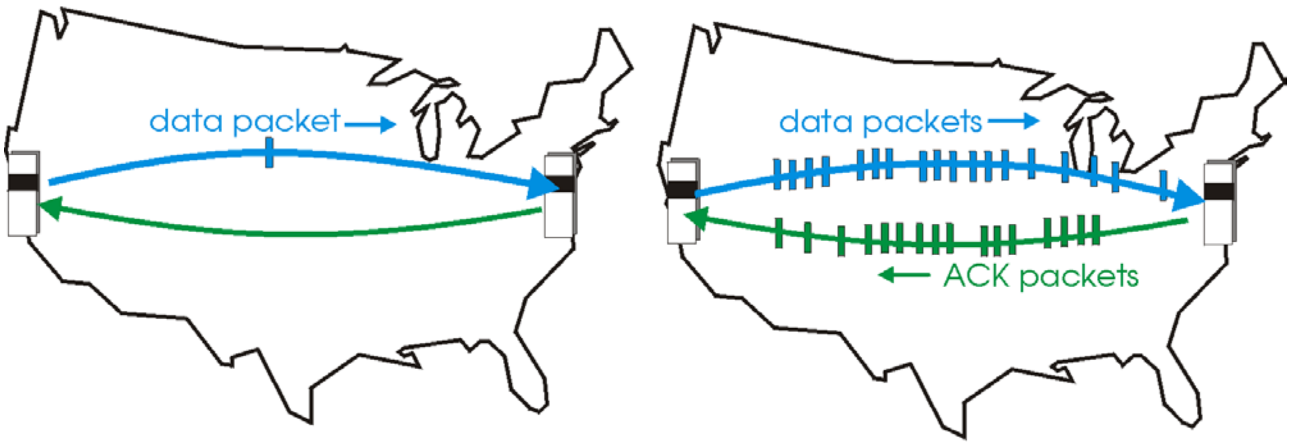
rdt3.0: stop-and-wait operation



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

Pipelining

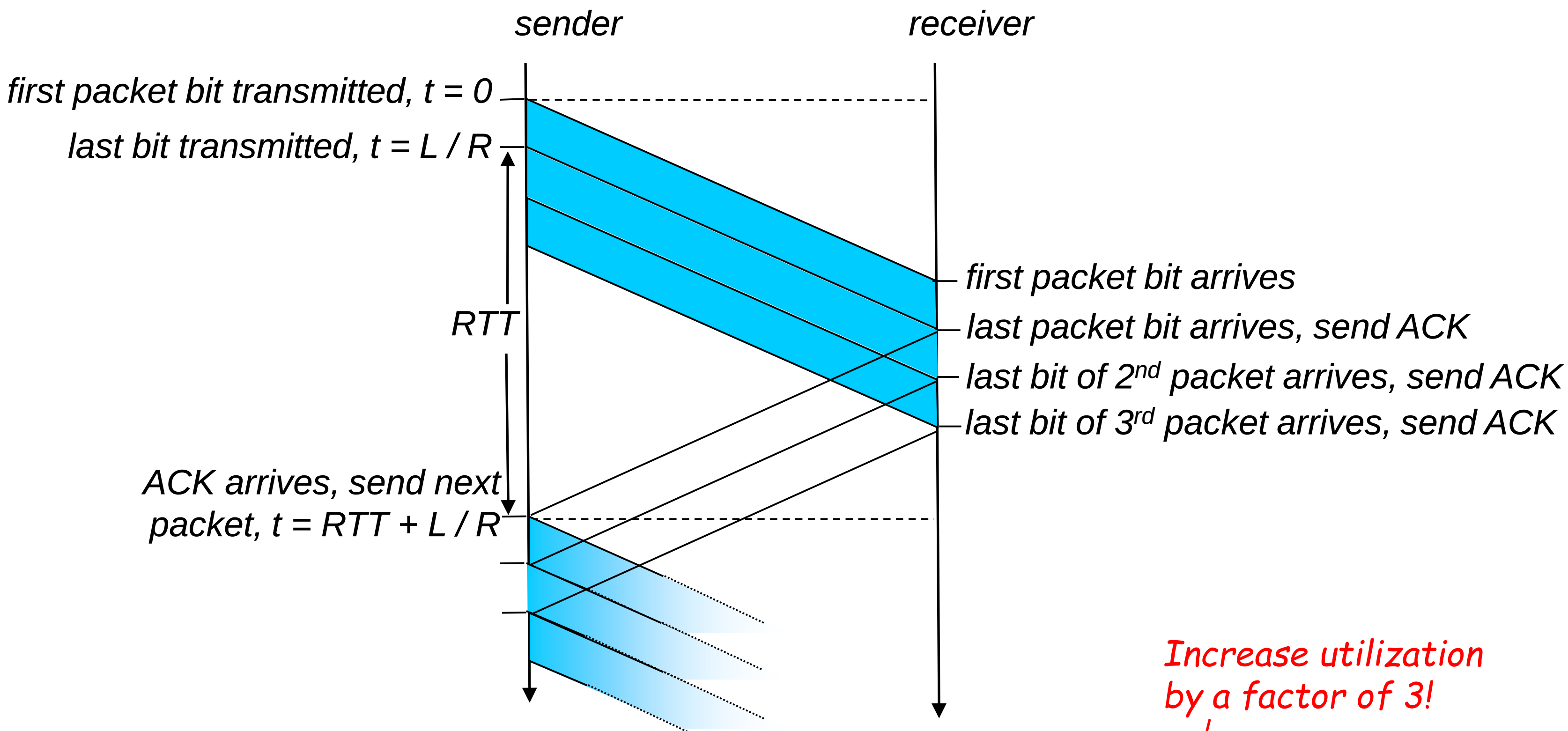
Pipelining: sender allows multiple, "in-flight", yet-to-be-acknowledged pkts



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

(b) a pipelined protocol in operation

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: Additional Mechanisms

- Range of sequence numbers must be increased
- Buffering at sender and/or receiver
  - The sender must buffer all packets not yet acknowledged
  - The receiver may buffer out-of-order packets
- The range of seq numbers and buffer size depend on how the protocol manages lost, corrupted, and delayed packets
- Error recovery strategies
  - Go-back-N
  - Selective Repeat

Pipelining Protocols

Go-back-N

- *sender:* up to N unACKed pkts in pipeline
- *receiver:* only sends cumulative ACKs
  - doesn't ACK pkt if there's a gap
- *sender:* has timer for oldest unACKed pkt
  - if timer expires: retransmit all unACKed packets

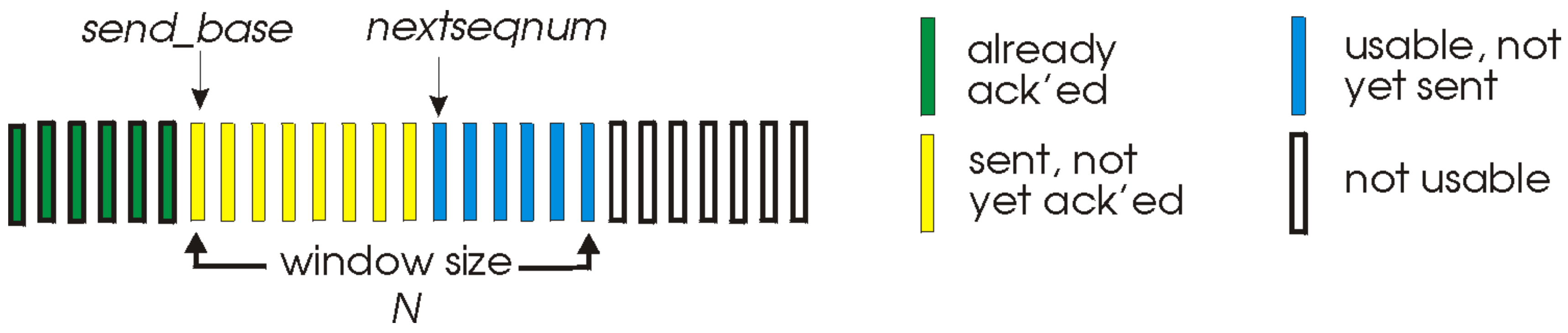
Selective Repeat

- *sender:* up to N unACKed packets in pipeline
- *receiver:* ACKs individual pkts
- *sender:* maintains timer for each unACKed pkt
  - if timer expires: retransmit only unACKed packets

Go-Back-N

Sender:

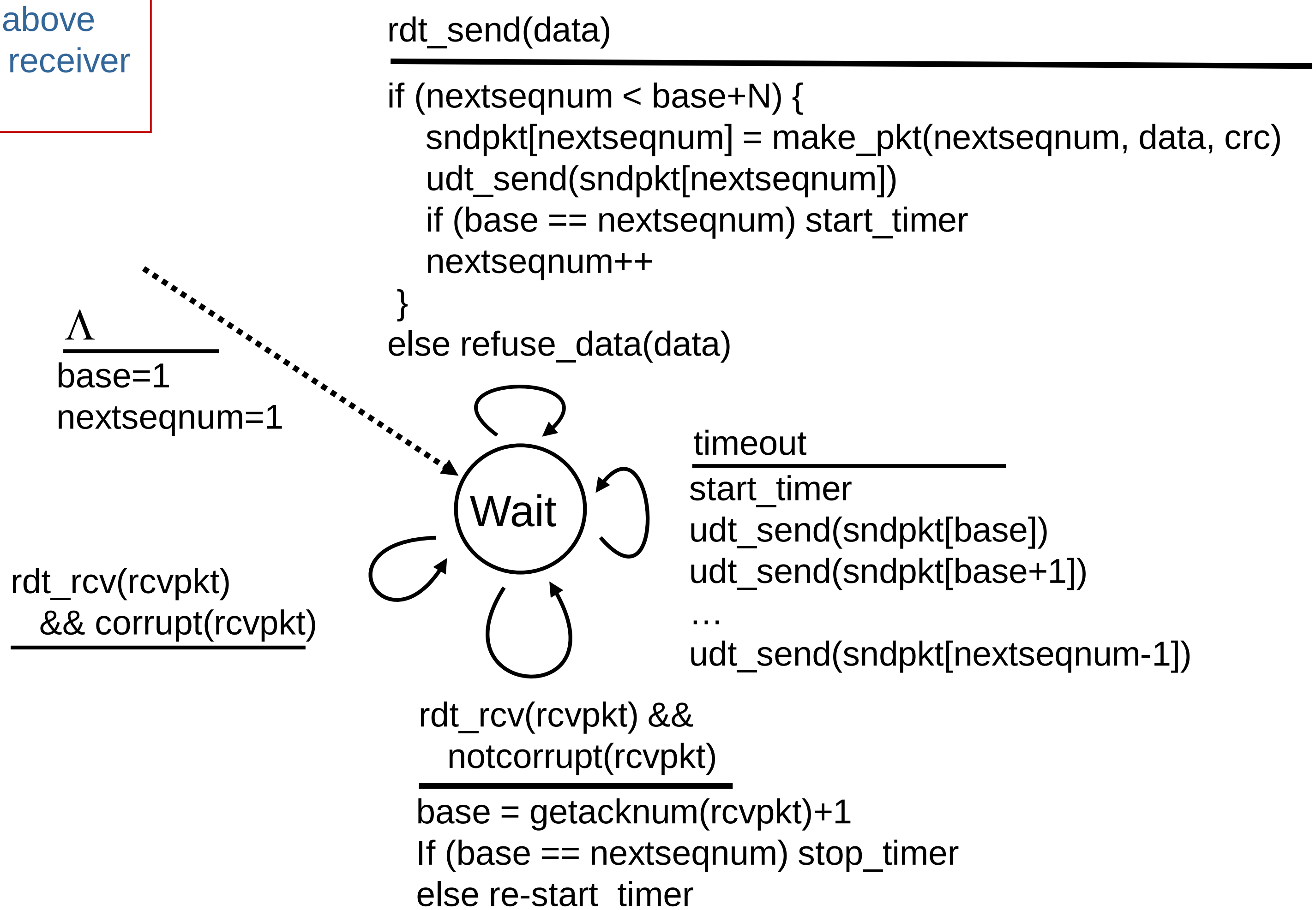
- k-bit seq # in pkt header
- "window" of up to N, consecutive unACKed pkts allowed



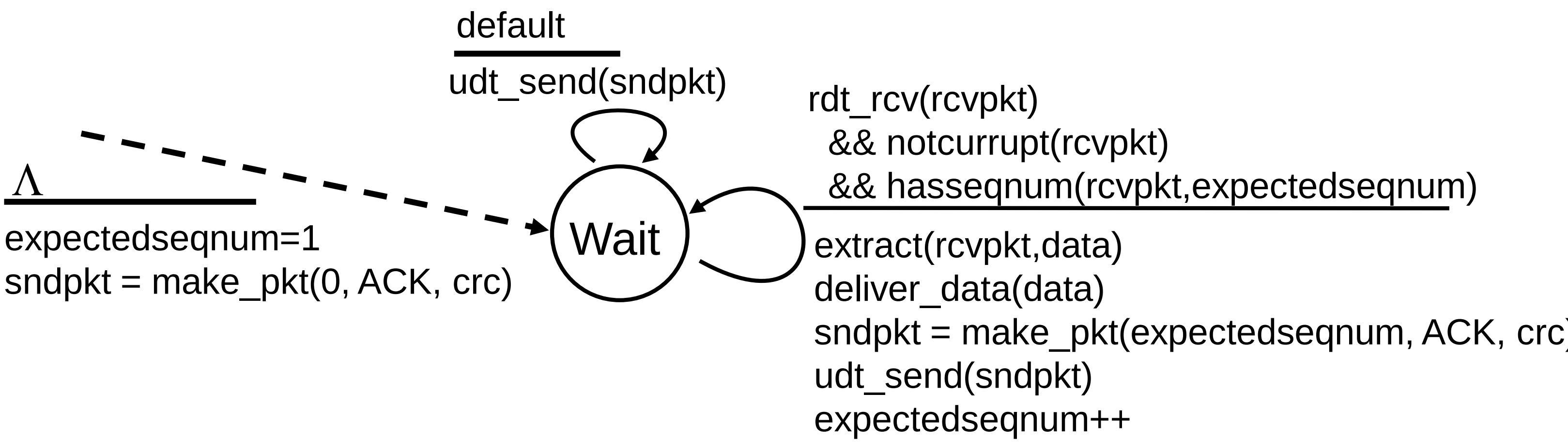
- ACK(n): ACKs all pkts up to, including seq # n ("cumulative ACK")
  - may receive duplicate ACKs (see receiver)
- timer for the oldest packet only (send base)
- timeout: retransmit pkt sendbase and all higher seq # pkts in window

GBN: sender extended FSM

Events:
- Data from above
- ACK from receiver
- Timeout



GBN: receiver extended FSM

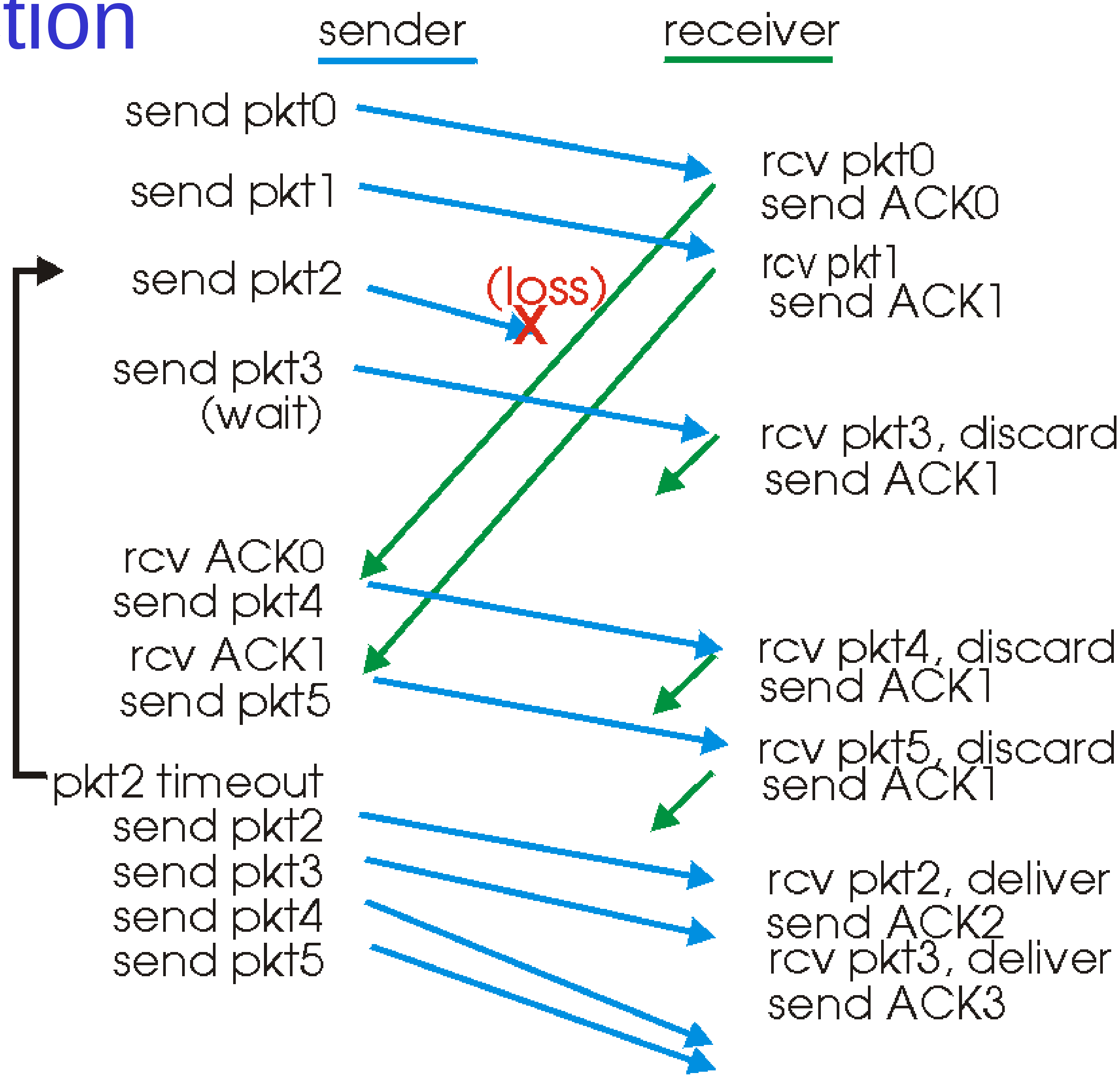


ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq #

- 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



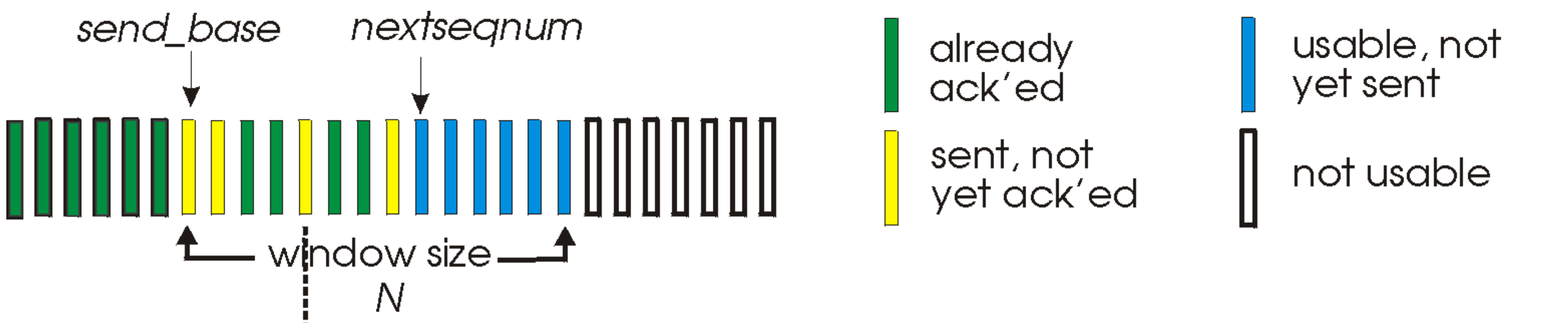
Limits of Go-back-N

- Packets are acked on a *cumulative* base
- Upon experiencing a time-out the sender retransmits *all* packets since the last received in order
  - Un-necessary re-transmissions
    - Consume bandwidth
    - Consume energy
  - The receiver does not need to buffer out-of-order packets
- Complexity is shifted at the sender side
  - The receiver only needs to know **`expectedseqnum`**

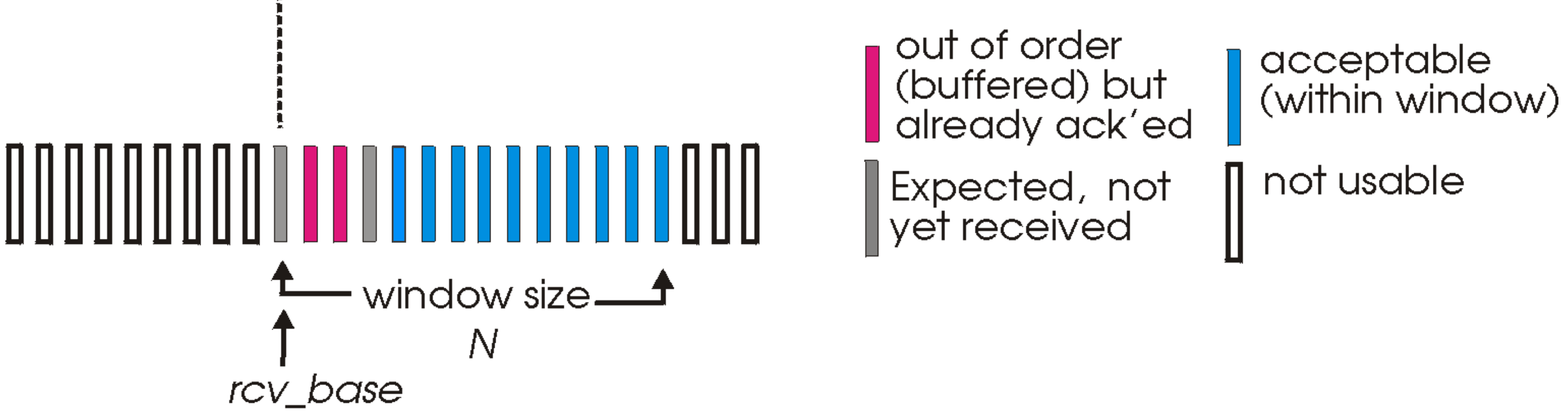
Selective Repeat (SR)

- 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

SR: sender, receiver windows



(a) sender view of sequence numbers



(b) receiver view of sequence numbers

Selective Repeat

sender

data from above :

- if next available seq # in window, send pkt

timeout(n):

- resend pkt n, restart timer(n)

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)
- If out-of-order: buffer pkt n
- If in-order: deliver pkt n

  also deliver buffered, in-order pkts,

  advance rcv_base to next not-yet-received pkt

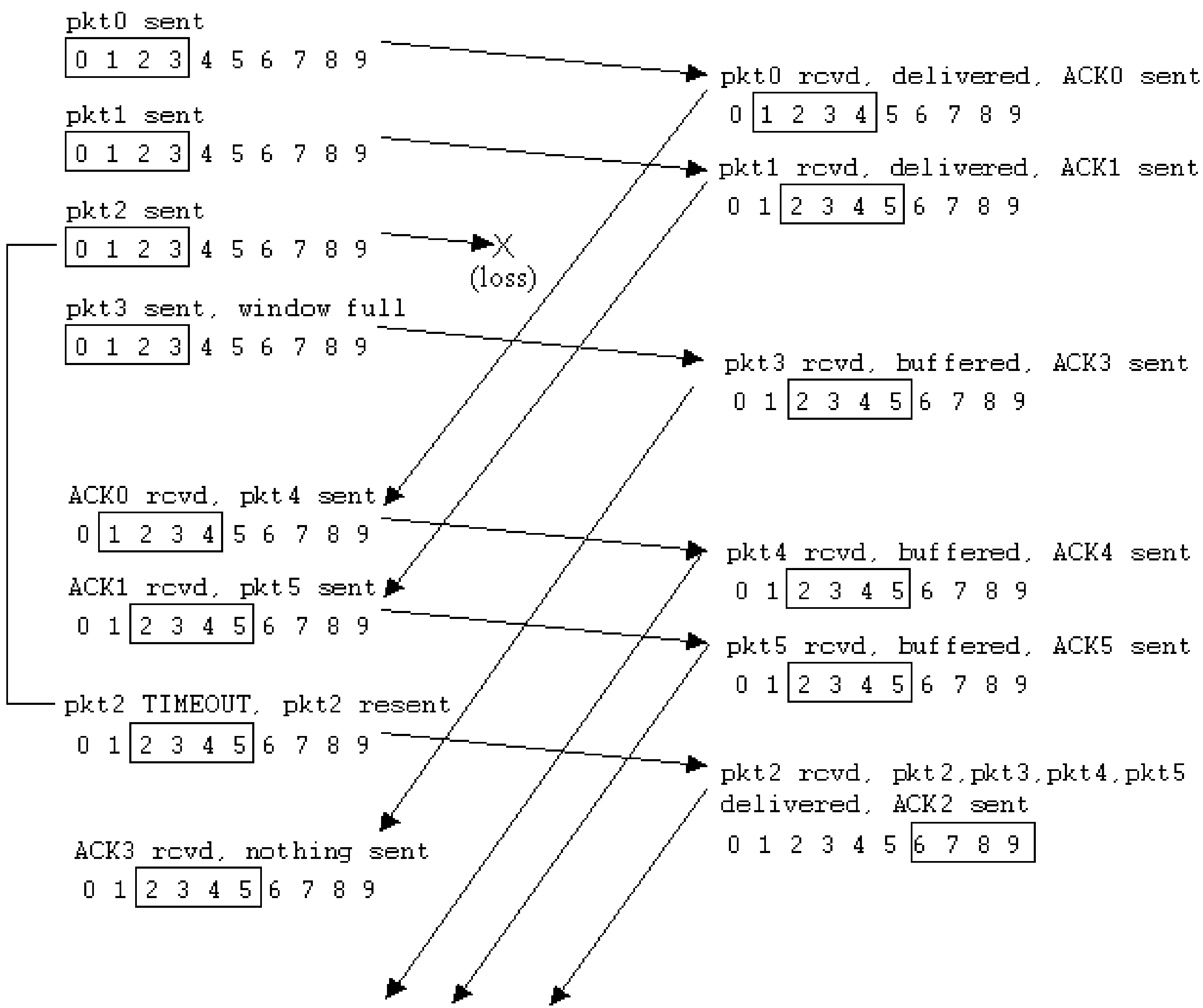
pkt n in [rcvbase-N,rcvbase-1]

- ACK(n)

otherwise:

- ignore

Selective Repeat in action

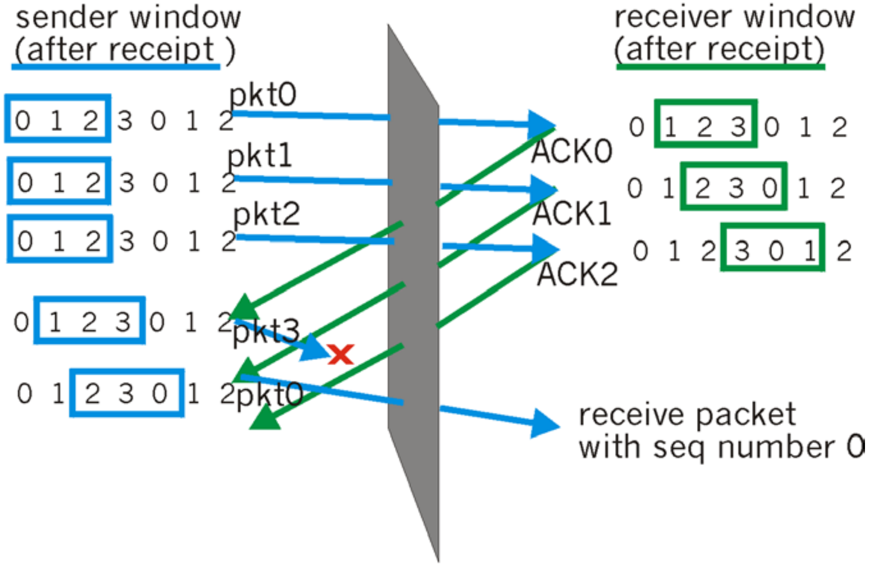


SR: dilemma

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)



(b)

Window Sizing

- Question
- What relationship between window size and sequence number space?

- The window size must be less than or equal to half of the sequence number space

Window Sizing

- Performance
  - The window size should allow the sender to fill the pipe
- Flow Control
  - The window size should also avoid buffer overflow at the receiver
  - In a Point-to-Point link the window size can be defined based on
    - Round Trip Time (RTT)
    - Receiver Buffer Size

Reliable Data Transfer: Summary

- Error detection (e.g., CRC)
- Acnowledgements (ACKs)
- Negative Acnowledgements (NAKs)
- Retransmission
- Sequence Number
- Retransmission Timer (Timeout)
- Pipelining (window)

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Point to Point Data-Link Protocols

- one sender, one receiver, one link
  - e.g., dialup link, ISDN line, ADSL, …
- Popular point-to-point DLC protocols:
  - SLIP (Serial Link IP)
  - PPP (Point-to-Point Protocol)
  - HDLC: High level Data Link Control
    - Data Link used to be considered "high layer" in protocol stack!

SLIP

- Ideato nel 1984 (RFC 1055)
  - Per interconnettere SUN ws a Internet tramite rete telefonica
- Nessuna gestione degli errori
  - I livelli superiori devono farsene carico
- Supporta solo IP
- Assegnazione statica di indirizzi IP
  - Data la limitatezza degli indirizzi IP è un grosso limite
- Nessuna autenticazione
  - Va bene per linee dedicate ma non per collegamenti telefonici
- Molte versioni (spesso incompatibili)
  - Non è uno standard Internet approvato

Point to Point Protocol (PPP) [RFC 1547]

- packet framing: encapsulation of network-layer datagram in Data Link frame
  - carry network layer data of any network layer protocol (not just IP) *at same time*
  - ability to demultiplex upwards
- bit transparency: must carry any bit pattern in the data field
- error detection (no correction)
- connection liveness: detect and signal link failure to network layer
- network layer address negotiation: endpoint can learn/configure each other's network address

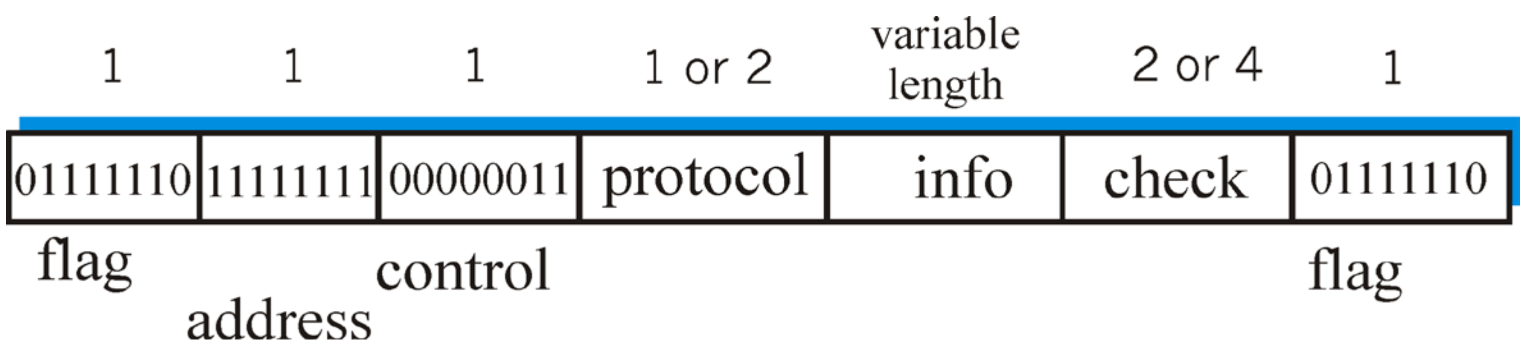
PPP non-requirements

- No error correction/recovery
- No flow control
- Possible out of order delivery
- No support for point-to-multi-point communication
  - Other DL protocols supports this feature (e.g., HDLC)

Error recovery, flow control, data re-ordering
all delegated to higher layers (e.g., TCP)!

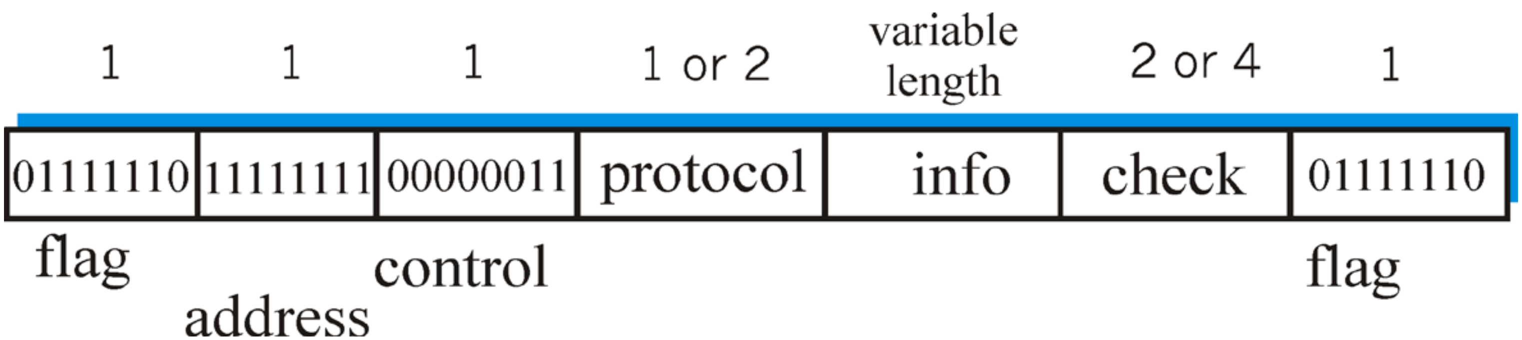
PPP Data Frame

- Flag: delimiter (framing)
- Address: does nothing (only one option)
- Control: does nothing; in the future possible multiple control fields
- Protocol: upper layer protocol to which frame delivered (eg, PPP-LCP, IP, IPCP, etc)



PPP Data Frame

- info: upper-layer data being carried
- check: cyclic redundancy check for error detection

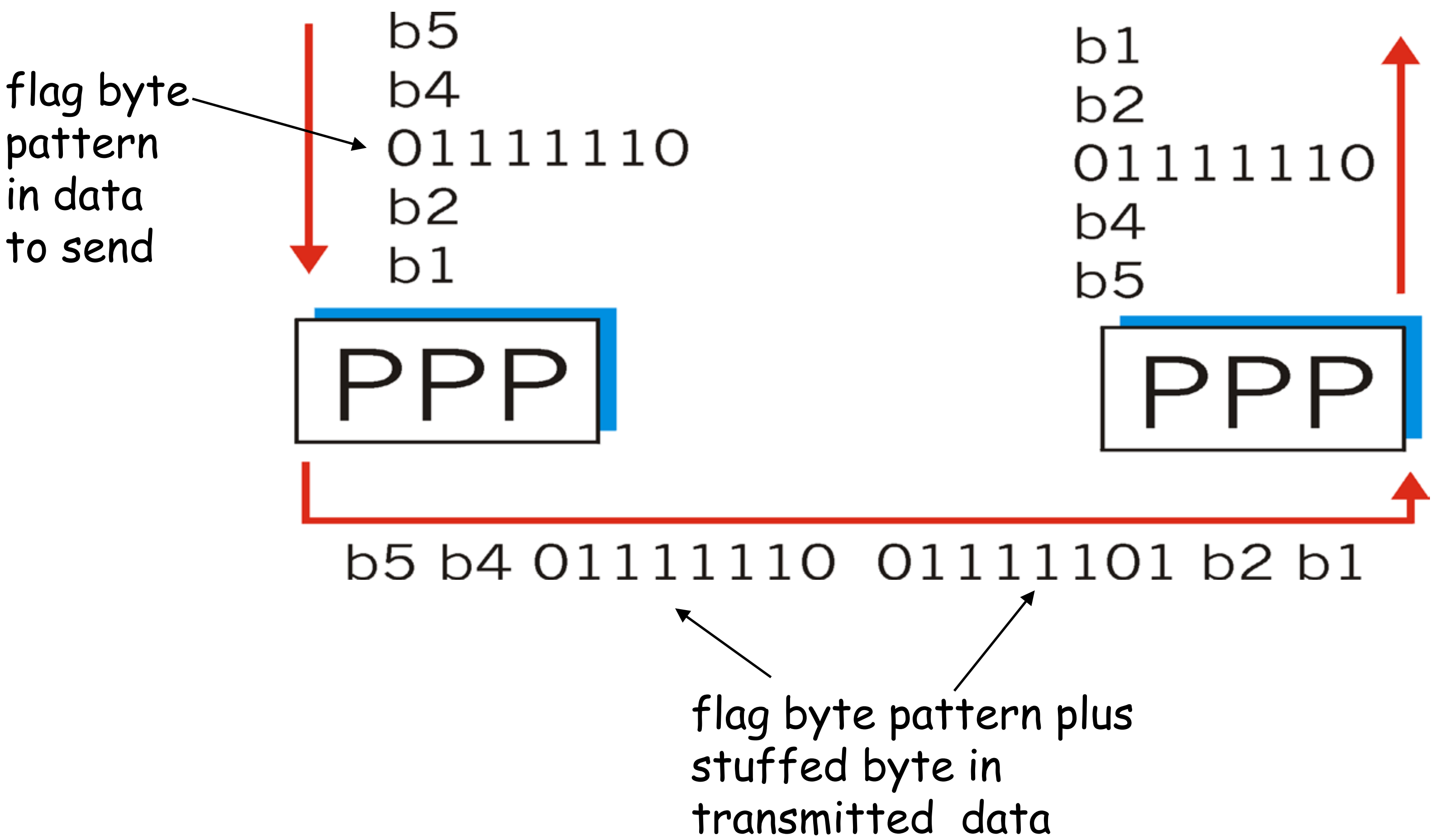


Byte Stuffing

- "data transparency" requirement: data field must be allowed to include flag pattern <01111110>
  - Q: is received <01111110> data or flag?

- Sender:
  - adds ("stuffs") extra < 01111101> byte after each <01111110> *data* byte
    - < 01111101> byte = escape byte
- Receiver:
  - Whenever receives 01111101 01111110 discards the escape byte

Byte Stuffing/Unstuffing



Byte Stuffing/Unstuffing (More)

- Sender (byte stuffing)
  - 01111110 → 01111101 01111110
  - 01111101 → 01111101 01111101
- Receiver (byte unstuffing)
  - 01111101 01111110 → 01111110
  - 01111101 01111101 → 01111101

PPP Link Control

Before exchanging network-layer data, Data Link peers must

- configure PPP link (max. frame length, authentication)
  - Through Link Control Protocol (LCP)
- learn/configure network layer information
  - for IP: carry IP Control Protocol (IPCP) msgs (protocol field: 8021) to configure/learn IP address

Esempio: Attivazione di una connessione PPP via modem

1. Il PC chiama il router del provider via modem
2. Il modem del provider risponde
   - Si stabilisce un collegamento fisico tra PC e router del provider
3. Negoziazione dei parametri di link (protocollo LCP)
   - Utilizzo dei campi Address e Control, Lunghezza max frame, Protocollo di autenticazione,
4. Negoziazione parametri di rete
   - Compressione pacchetti IP?, …
   - Viene effettutata tramite una serie di pacchetti IPCP (inviati mediante frame PPP)
5. Viene assegnato un indirizzo IP al PC
6. Il PC è ora collegato a Internet

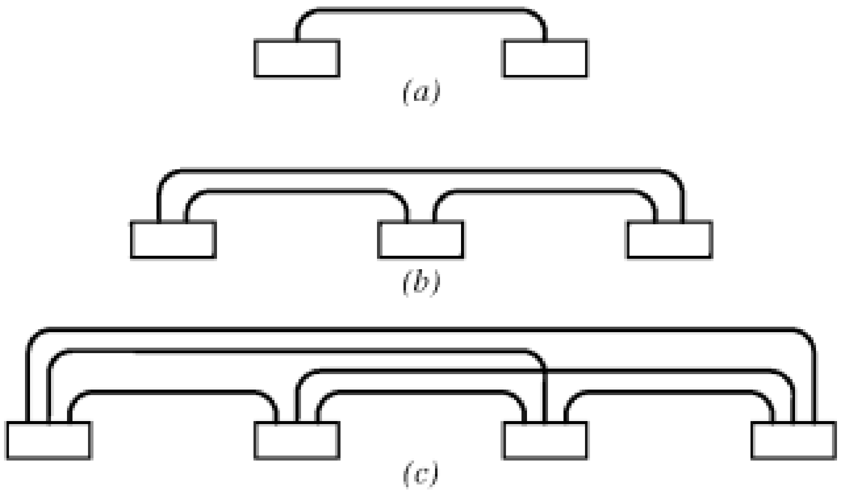
Esempio: Chiusura di una connessione PPP

1. Protocollo IPCP
   - Rilascio dell'indirizzo IP
   - Rilascio della connessione di livello rete
2. Protocollo LCP
   - Rilascio della connessione di livello Direct Connection Networks
3. Viene rilasciato il collegamento telefonico

Direct Connection Networks

- Introduction
- Error detection and correction
- Reliable Data Transfer
- PPP
- Multiple access protocols
- Local Area Networks (LAN)
- Ethernet

Limits of Point-to-Point Links



N: Number of Nodes

Required number of links

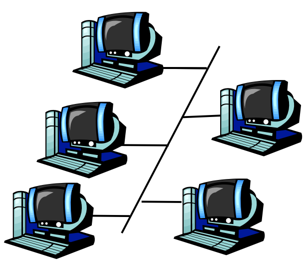
$$\frac{N(N-1)}{2}$$

Doesn't scale!!

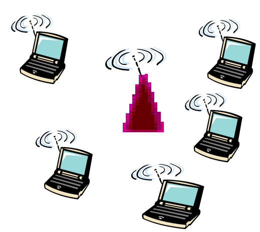
Multiple Access Links and Protocols

Two types of "links":

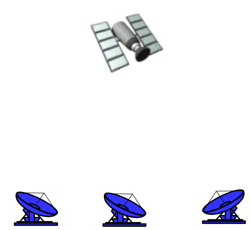
- point-to-point links
  - PPP protocol
  - HDLC protocol
- broadcast (shared wire or medium)
  - old-fashioned Ethernet
  - upstream HFC
  - 802.11 wireless LAN



shared wire (e.g., cabled Ethernet)



shared RF (e.g., 802.11 WiFi)



shared RF (satellite)

humans at a cocktail party (shared air, acoustical)

Multiple Access protocols

- single shared broadcast channel
- two or more simultaneous transmissions by nodes: interference
  - collision if node receives two or more signals at the same time

Multiple Access Protocol

- distributed algorithm that determines how nodes share channel, i.e., determine when nodes can transmit
- communication about channel sharing must use channel itself!
  - no out-of-band channel for coordination

Ideal Multiple Access Protocol

Broadcast channel of rate R bps

1. Fully Utilization
   - when one node wants to transmit, it should send at rate R
2. Fairness
   - when M nodes want to transmit, each should send at average rate R/M
3. Fully decentralization
   - no special node to coordinate transmissions
   - no synchronization of clocks, slot assignment, …
4. Simplicity

MAC Protocols: a taxonomy

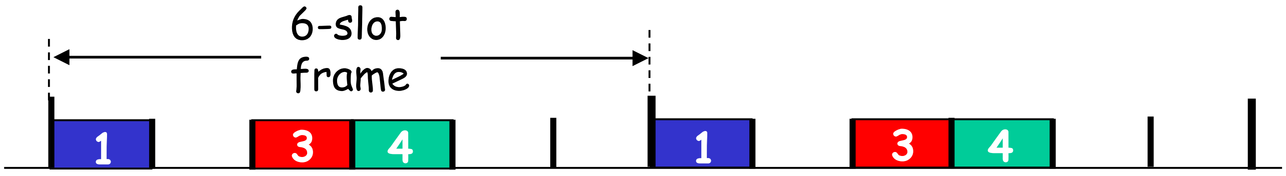
Three broad classes:

- Channel Partitioning
  - divide channel into smaller "pieces" (time slots, frequency, code)
  - allocate piece to node for exclusive use
- Random Access
  - channel not divided, allow collisions
  - "recover" from collisions
- "Taking turns"
  - nodes take turns, but nodes with more to send can take longer turns

Channel Partitioning MAC protocols: TDMA

TDMA: Time Division Multiple Access

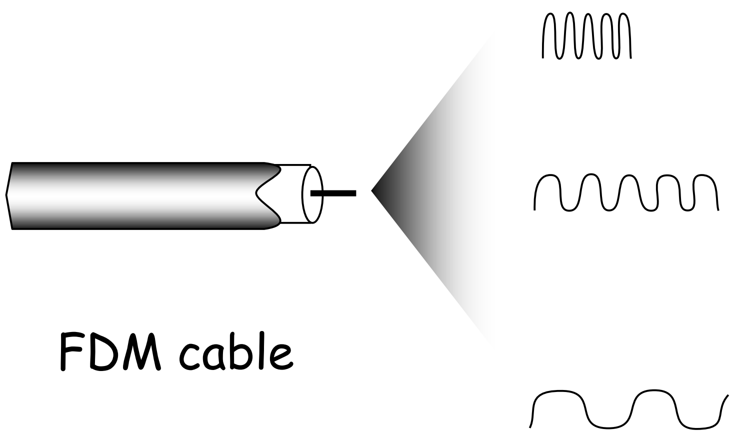
- access to channel in "rounds"
- each station gets fixed length slot (length = pkt trans time) in each round
- unused slots go idle
- example: 6-station LAN, 1,3,4 have pkt, slots 2,5,6 idle

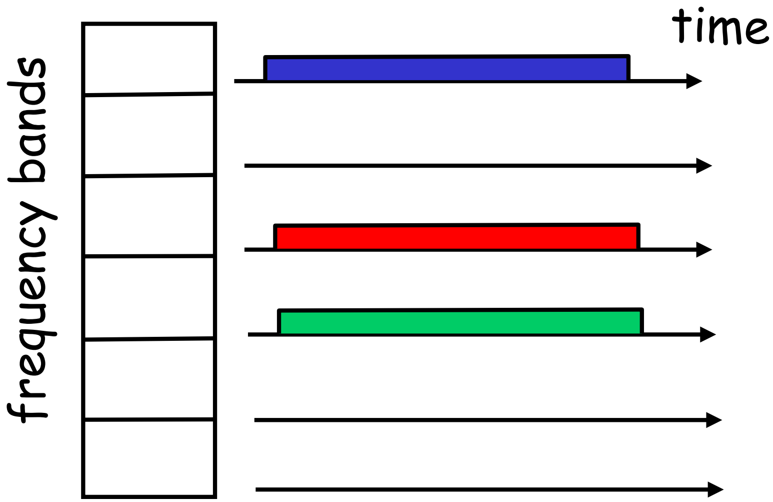


Channel Partitioning MAC protocols: FDMA

FDMA: Frequency Division Multiple Access

- channel spectrum divided into frequency bands
- each station assigned fixed frequency band
- unused transmission time in frequency bands go idle
- example: 6-station LAN, 1,3,4 have pkt, frequency bands 2,5,6 idle





Channel Partitioning MAC protocols: CDMA

CDMA: Code Division multiple access

- Each pair of nodes assigned with different code
  - Unique code used to encode transmitted data
- Simultaneous transmissions
- Each receiver can correctly decode
  - In spite of interferences from other nodes
- Mainly used in military applications and cellular telephony

Random Access Protocols

- When node has packet to send
  - transmit at full channel data rate R.
  - no *a priori* coordination among nodes
- two or more transmitting nodes ➜ "collision",
- random access MAC protocol specifies:
  - how to avoid/detect collisions
  - how to recover from collisions (e.g., via delayed retransmissions)
- Examples of random access MAC protocols:
  - slotted ALOHA
  - ALOHA
  - CSMA, CSMA/CD, CSMA/CA

Slotted ALOHA

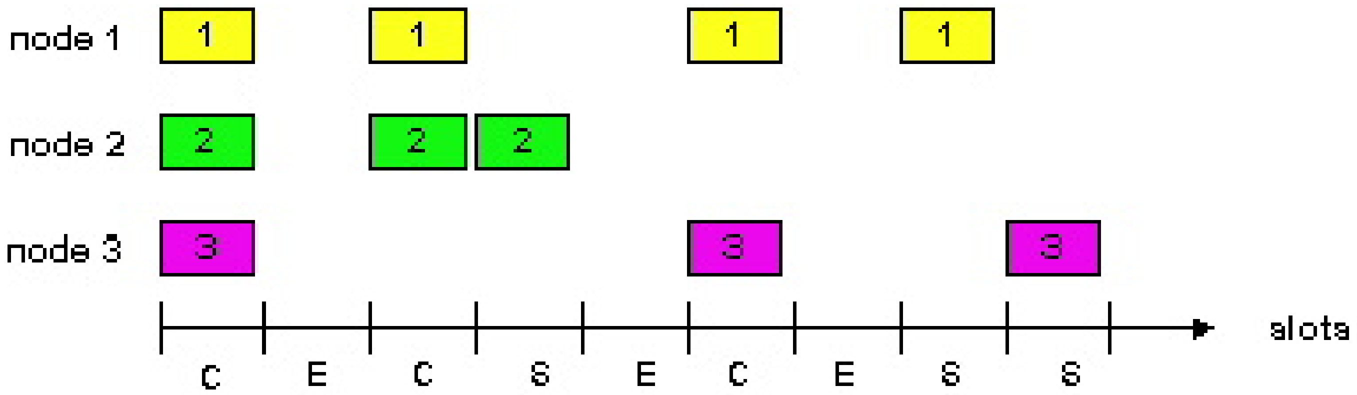
Assumptions:

- all frames same size
- time divided into equal size slots (time to transmit 1 frame)
- nodes start to transmit only at slot beginning
- nodes are synchronized
- if 2 or more nodes transmit in the same slot, all nodes detect collision

Operation:

- when node obtains fresh data, transmits in next slot
  - *if no collision:* node can send new frame in next slot
  - *if collision:* node retransmits frame in each subsequent slot *with probability p* until success

Slotted ALOHA



Pros

- single active node can continuously transmit at full rate of channel
- highly decentralized: only nodes need to be in sync
- simple

Cons

- collisions, wasting slots
- idle slots
- nodes may be able to detect collision in less than time to transmit packet
- clock synchronization

Slotted Aloha efficiency

Efficiency : long-run fraction of successful slots (many nodes, all with many frames to send)

- *suppose:* N nodes with many frames to send, each transmits in slot with probability *p*
- prob that given node has success in a slot = $p(1-p)^{N-1}$
- prob that *any* node has a success = $Np(1-p)^{N-1}$
- max efficiency: find p* that maximizes $Np(1-p)^{N-1}$
- for many nodes, take limit of $Np^*(1-p^*)^{N-1}$ as N goes to infinity, gives:

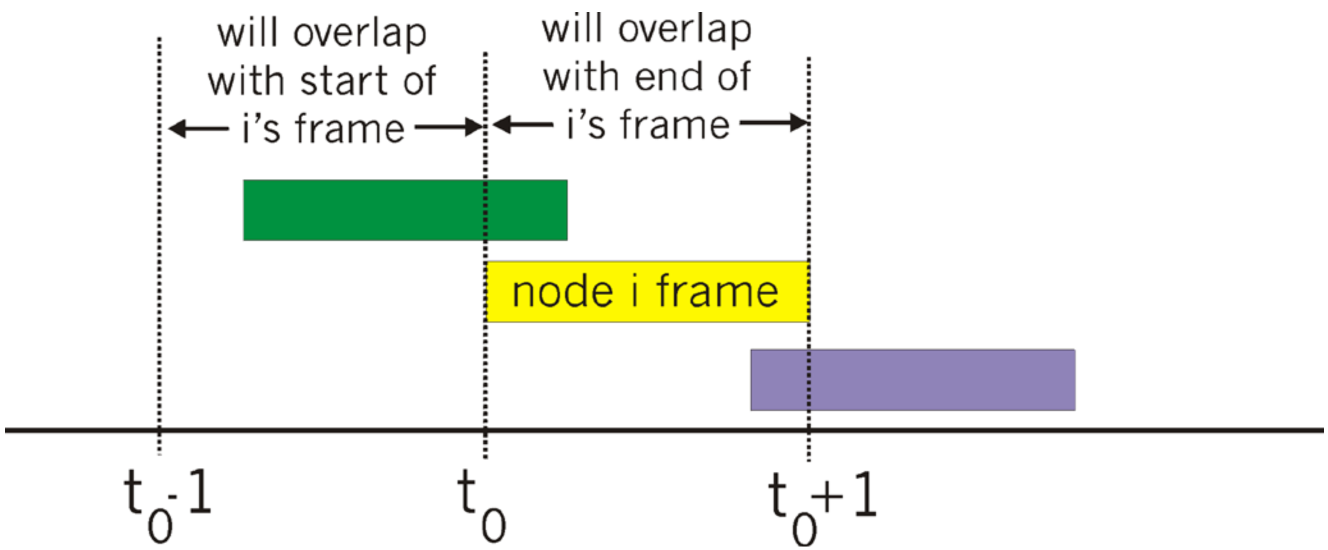
Max efficiency = 1/e = .37

At best: channel used for useful transmissions 37% of time!

!

Pure (unslotted) ALOHA

- unslotted Aloha: simpler, no synchronization
- when frame first arrives transmit immediately
  - After a collision transmit with probability p
- collision probability increases:
  - frame sent at t_0 collides with other frames sent in $[t_0-1, t_0+1]$



Pure Aloha efficiency

P(success by given node) = P(node transmits) ·

P(no other node transmits in $[t_0-1,t_0]$ ·

P(no other node transmits in $[t_0, t_0+1]$

$= p \cdot (1-p)^{N-1} \cdot (1-p)^{N-1}$

$= p \cdot (1-p)^{2(N-1)}$

... choosing optimum p and then letting n -> infty ...

= 1/(2e) = .18

even worse than slotted Aloha!

CSMA (Carrier Sense Multiple Access)

CSMA: listen before transmit:

If channel sensed idle: transmit entire frame

- If channel sensed busy, defer transmission

- human analogy: don't interrupt others!

CSMA collisions

collisions *can* still occur:

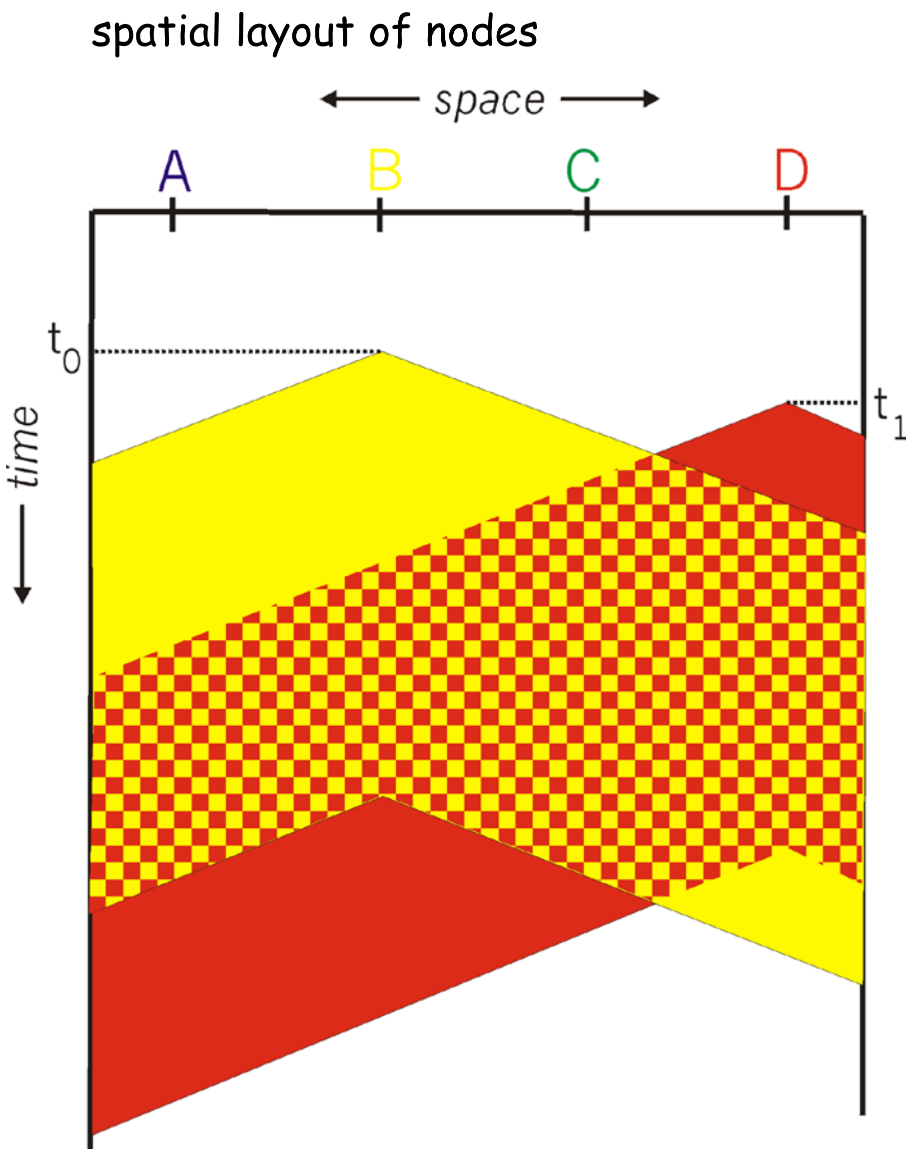
propagation delay means two nodes may not hear each other's transmission

collision:

entire packet transmission time wasted

note:

role of distance & propagation delay in determining collision probability



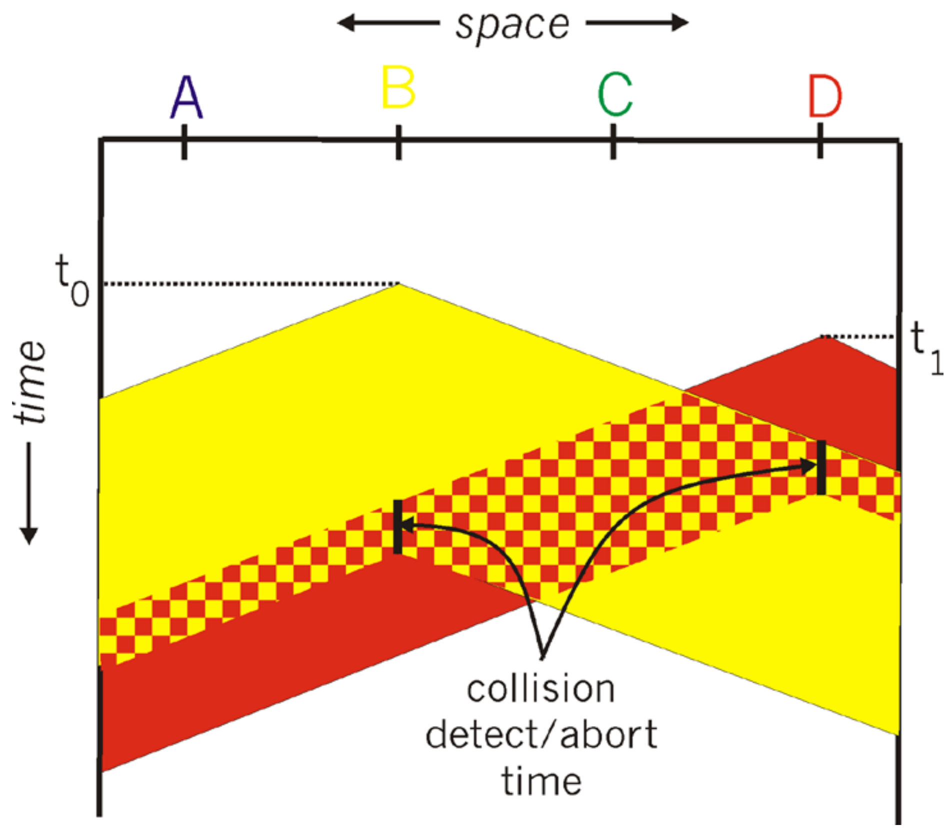
CSMA/CD (Collision Detection)

CSMA/CD: carrier sensing, deferral as in CSMA

- collisions *detected* within short time
- colliding transmissions aborted, reducing channel wastage

- collision detection:
  - easy in wired LANs: measure signal strengths, compare transmitted, received signals
  - difficult in wireless LANs: received signal strength overwhelmed by local transmission strength
- human analogy: the polite conversationalist

CSMA/CD collision detection



"Taking Turns" MAC protocols

channel partitioning MAC protocols:

- share channel efficiently and *fairly* at high load
- *inefficient at low load*: delay in channel access, 1/N bandwidth allocated even if only 1 active node!

Random access MAC protocols

- *efficient at low load*: single node can fully utilize channel
- high load: *collision* overhead

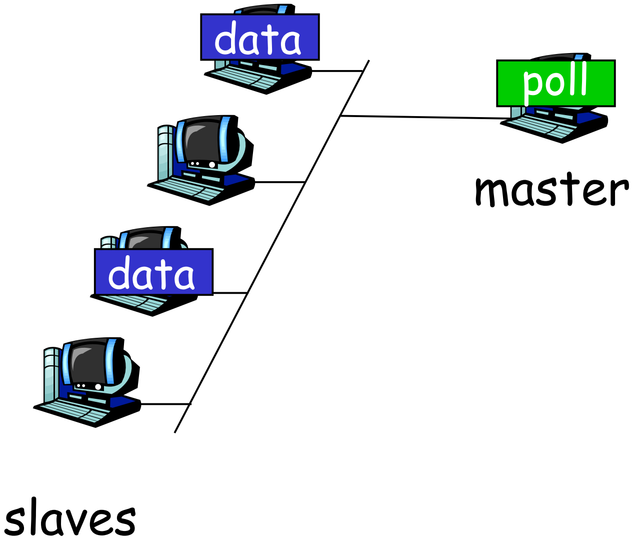
"taking turns" protocols

look for best of both worlds!

“Taking Turns” MAC protocols

Polling:

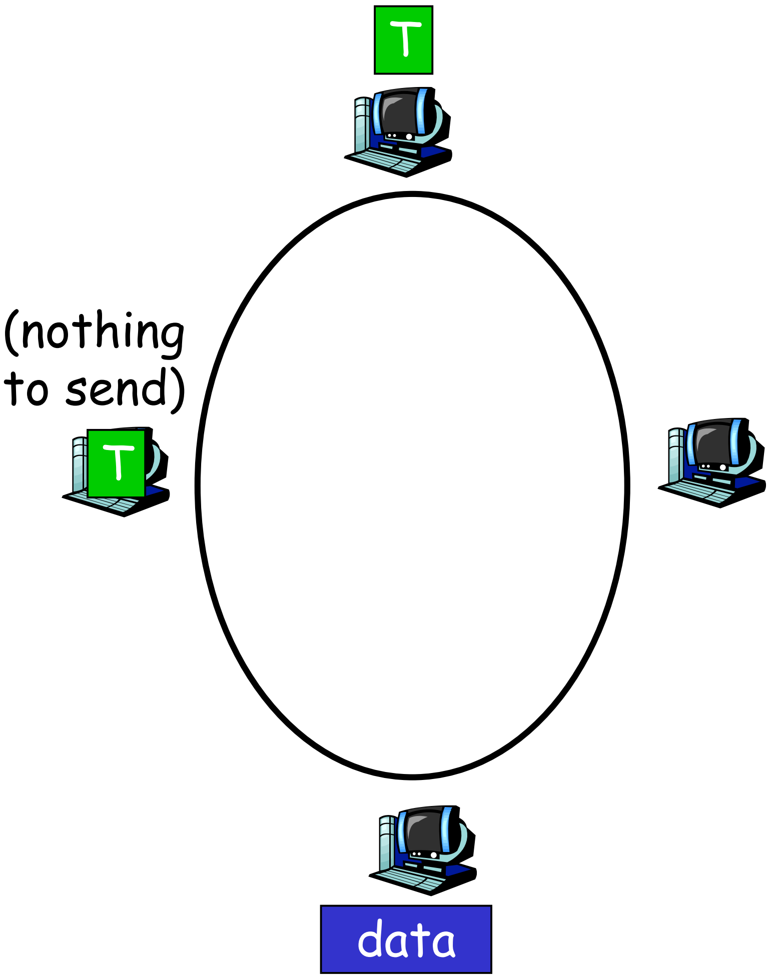
- master node “invites” slave nodes to transmit in turn
- typically used with “dumb” slave devices
- concerns:
  - polling overhead
  - latency
  - single point of failure (master)



"Taking Turns" MAC protocols

Token passing:

- control **token** passed from one node to next sequentially.
- token message
- concerns:
  - token overhead
  - latency
  - single point of failure (token)



Summary of MAC protocols

- *channel partitioning*, by time, frequency or code
  - Time Division, Frequency Division, CDMA
- *random access* (dynamic),
  - ALOHA, S-ALOHA, CSMA, CSMA/CD
  - carrier sensing: easy in some technologies (wire), hard in others (wireless)
  - CSMA/CD used in Ethernet
  - CSMA/CA used in 802.11
- *taking turns*
  - polling from central site, token passing
  - Bluetooth, FDDI, IBM Token Ring

Addressing

- Hardware Address
  - Also called Physical address, link-layer address, or MAC address:
  - function: *get frame from one interface to another physically-connected interface (same network)*

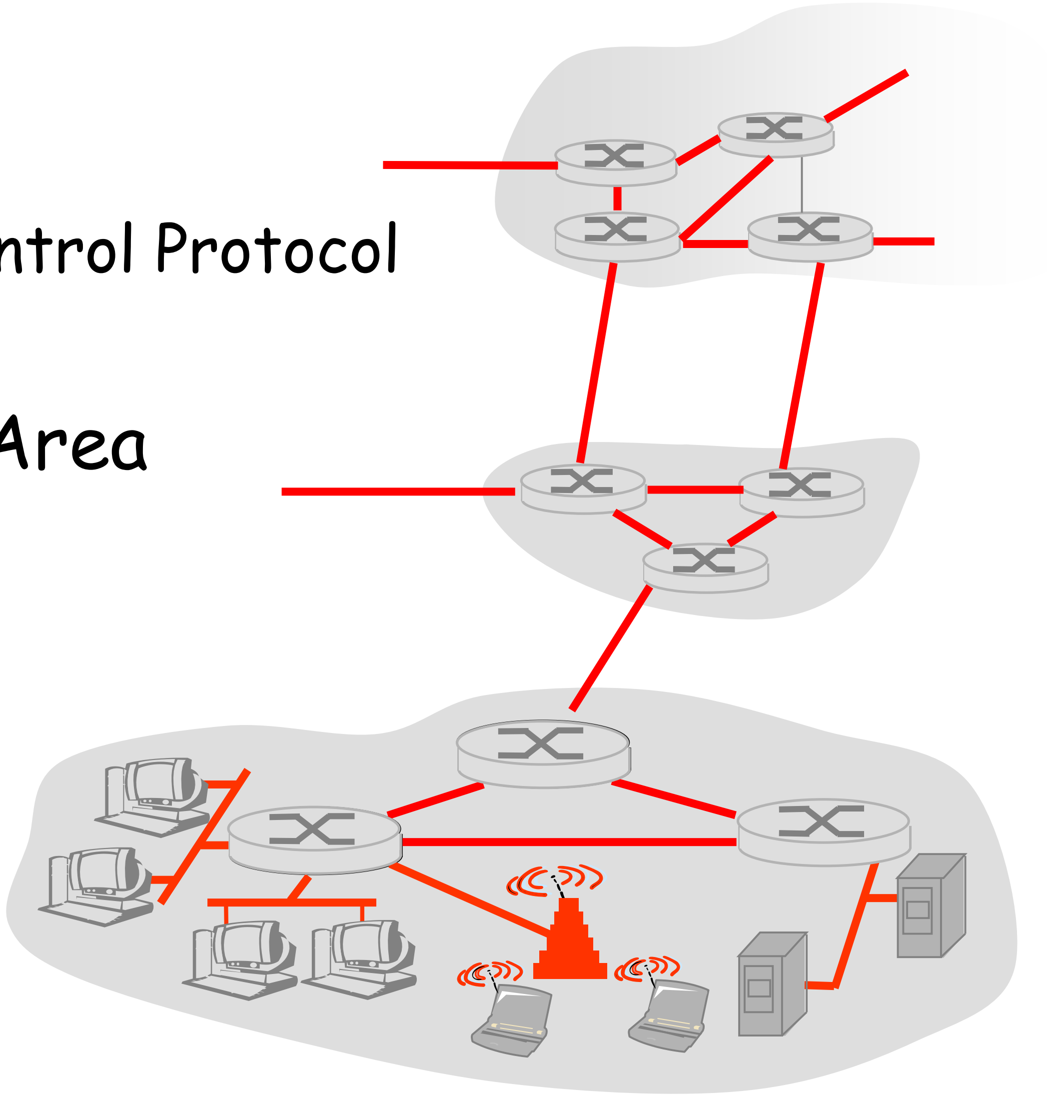
- IEEE Addressing Scheme (LAN)
  - 48 bit link-layer address
    - burned in NIC ROM, also sometimes software settable

Direct Connection Networks

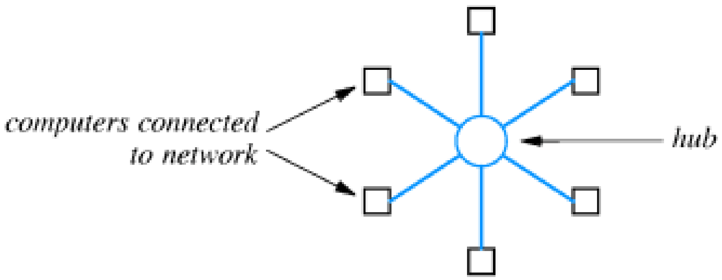
- Introduction
- Error detection and correction
- Reliable Data Transfer
- PPP
- Multiple access protocols
- Local Area Networks (LAN)
- Ethernet

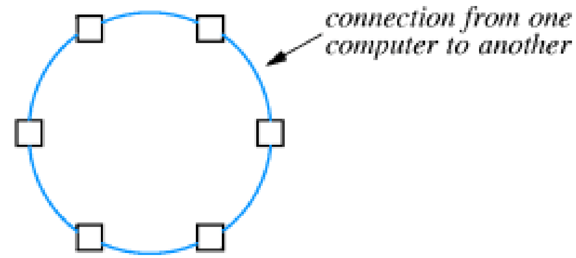
Local Area Networks (LANs)

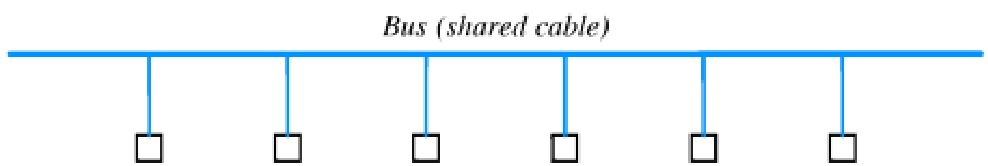
- Broadcast Medium
  - Medium Access Control Protocol
  - MAC addressing
- Limited Coverage Area
  - Building, Campus
- High Bit Rate
  - 10 Mbps – 10 Gbps



Local Area Networks (LANs)



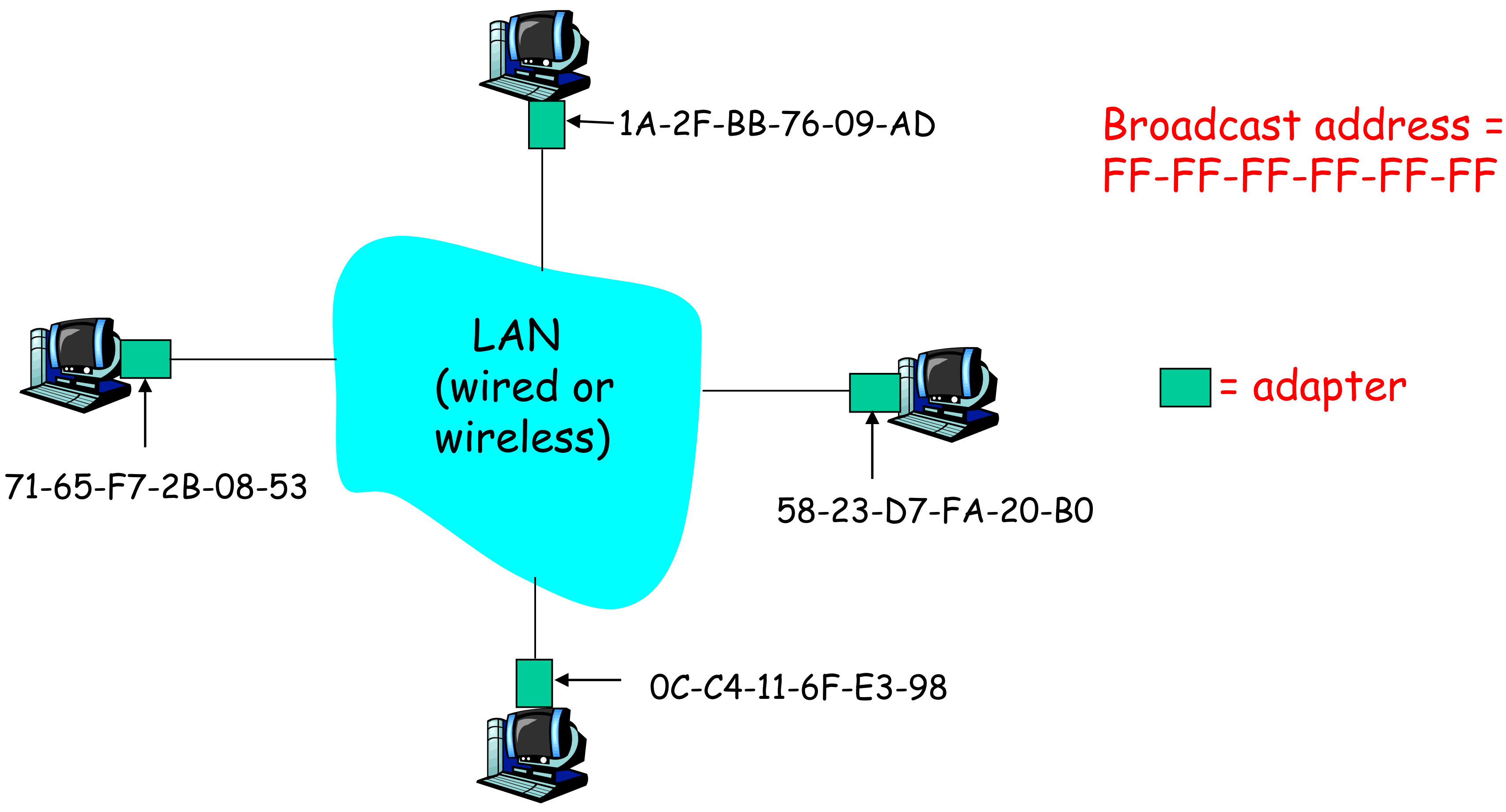




- **Broadcast Medium**
  - Medium Access Control (MAC) Protocol for channel access
  - MAC addressing

MAC Addresses

Each adapter on LAN has unique MAC address



MAC Address (more)

- MAC address allocation administered by IEEE
- manufacturer buys portion of MAC address space (to assure uniqueness)
- MAC flat address ➜ portability
  - can move LAN card from one LAN to another
- IP hierarchical address NOT portable
  - address depends on IP subnet to which node is attached
- analogy:

  (a) MAC address: like Social Security Number

  (b) IP address: like postal address

Direct Connection Networks

- Introduction
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- Multiple access protocols
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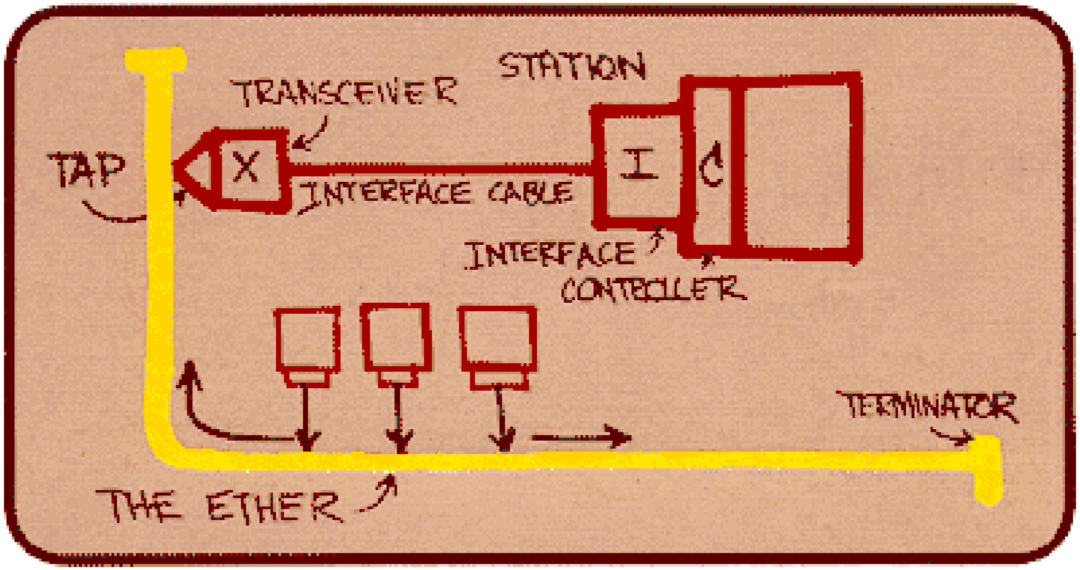
Ethernet

"dominant" wired LAN technology:

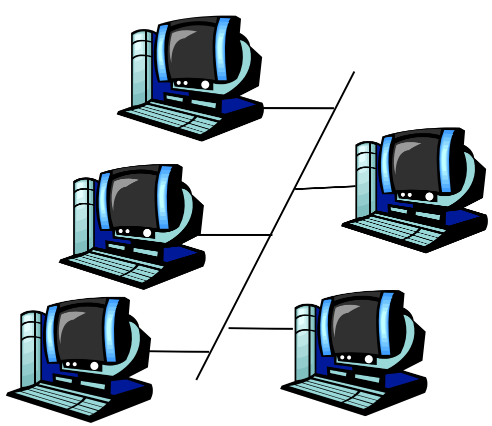
- cheap $20 for NIC
- first widely used LAN technology
- simpler, cheaper than token LANs and ATM
- kept up with speed race: 10 Mbps – 10 Gbps



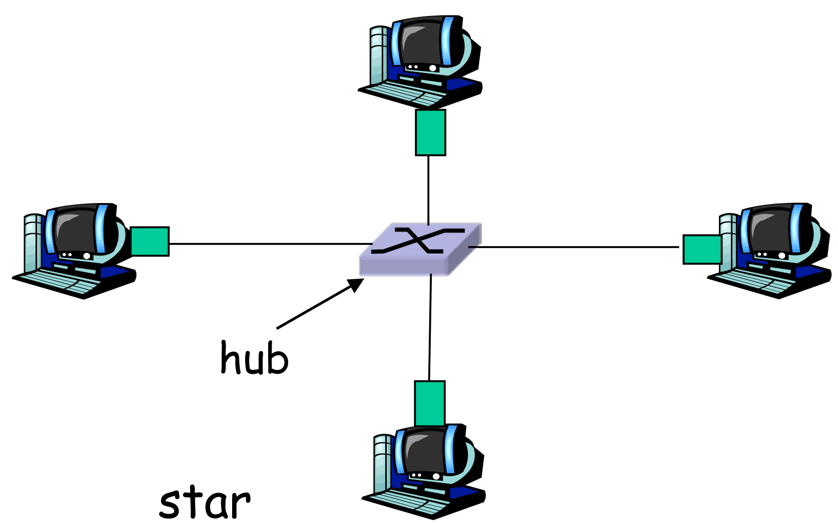
Metcalfe's Ethernet sketch

Topologies

- bus topology
  - Based on a bus
  - all nodes in same collision domain (can collide with each other)
- star topology
  - Based on a central *hub*
  - all nodes in same collision domain (just as in the bus topology)



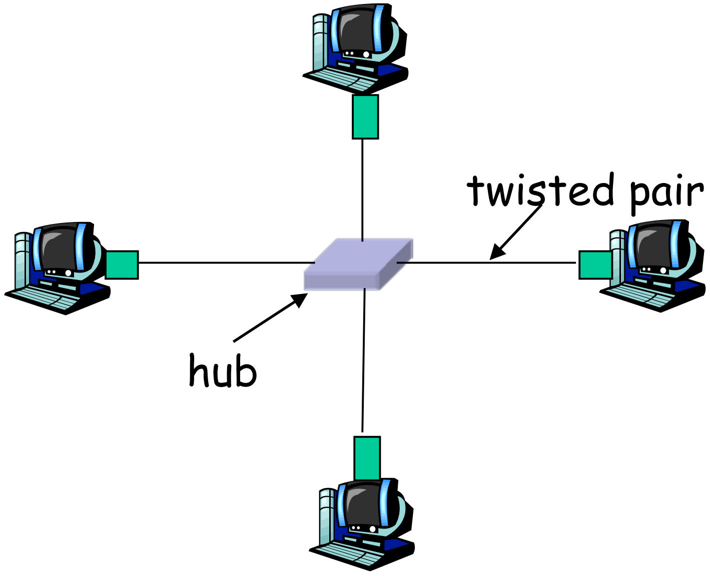
bus: coaxial cable



Hub

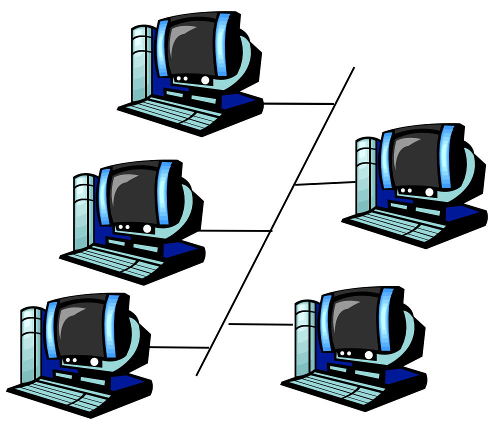
… physical-layer ("dumb") repeater:

- bits coming in one link go out *all* other links at same rate
- all nodes connected to hub can collide with one another
- no frame buffering
- no CSMA/CD at hub: host NICs detect collisions

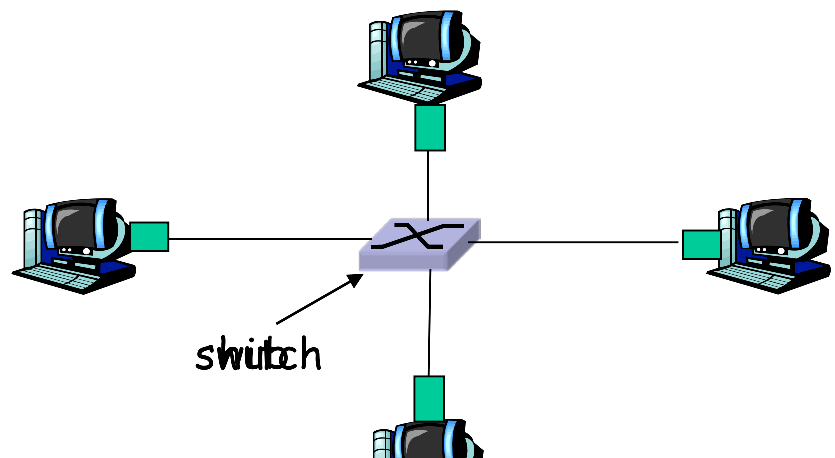


Switched Star topology

- bus topology popular through mid 90s
  - all nodes in same collision domain (can collide with each other)
- today: star topology prevails
  - active *switch* in center
  - each "spoke" runs a (separate) Ethernet protocol (nodes do not collide with each other)



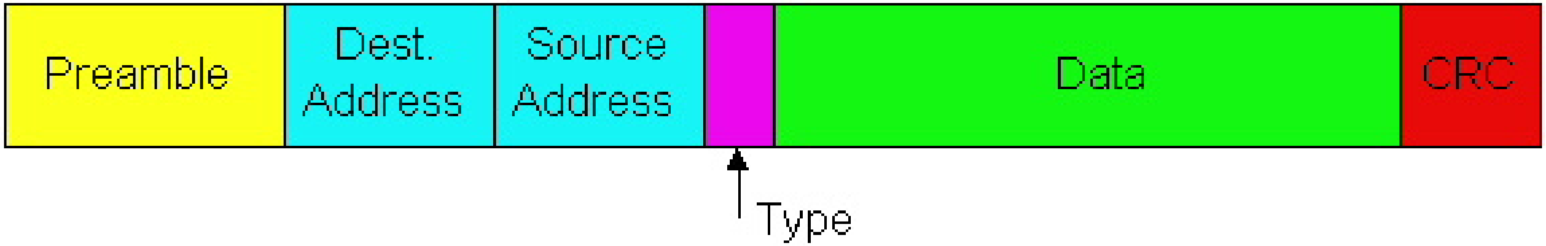
bus: coaxial cable



star

Ethernet Frame Structure

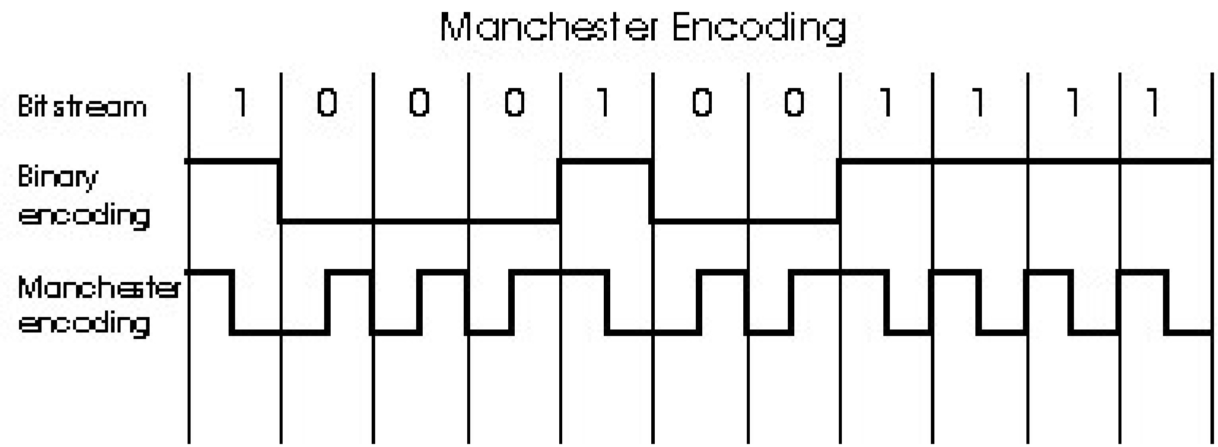
Sending adapter encapsulates IP datagram (or other network layer protocol packet) in Ethernet frame



Preamble (8 bytes):

- 7 bytes with pattern 10101010 followed by one byte with pattern 10101011
- used to synchronize receiver, sender clock rates

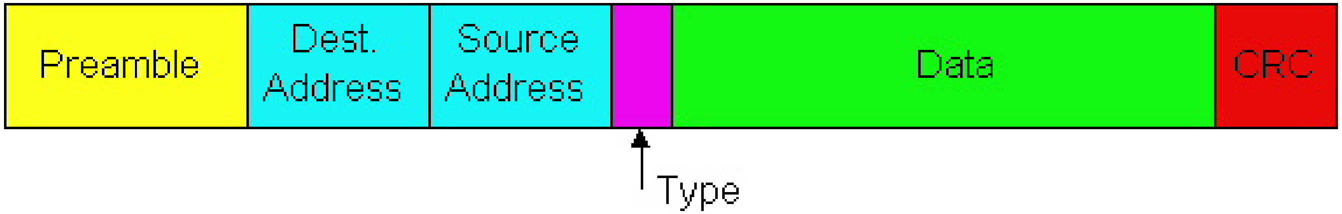
Manchester encoding



- used in 10BaseT
- each bit has a transition
- allows clocks in sending and receiving nodes to synchronize to each other
  - no need for a centralized, global clock among nodes!
- Hey, this is physical-layer stuff!

Ethernet Frame Structure (more)

- Addresses: 6 bytes
  - if adapter receives frame with matching destination address, or with broadcast address (eg ARP packet), it passes data in frame to network layer protocol
  - otherwise, adapter discards frame
- Type: indicates higher layer protocol (mostly IP but others possible, e.g., Novell IPX, AppleTalk)
- CRC: checked at receiver, if error is detected, frame is dropped



Ethernet: Service Type

- connectionless: No handshaking between sending and receiving NICs
- unreliable: receiving NIC doesn't send acks or nacks to sending NIC
  - stream of datagrams passed to network layer can have gaps (missing datagrams)
  - gaps will be filled if app is using TCP
  - otherwise, app will see gaps
- Ethernet's MAC protocol: unslotted CSMA/CD

Ethernet CSMA/CD algorithm

1. NIC receives data from network layer, creates frame
2. If NIC senses channel idle for 96 bits, starts frame transmission

   If NIC senses channel busy, waits until channel idle for 96 bits, then transmits
3. If NIC transmits entire frame without detecting another transmission, NIC is done with frame !
4. If NIC detects another transmission while transmitting, aborts and sends 48-bit jam signal
5. After aborting, NIC enters **exponential backoff**: after the *n*-th collision, NIC chooses *K* at random from $\{0,1,2,\ldots,2^m-1\}$,

   $$m=\min(n,10)$$

   NIC waits K·512 bit times, returns to Step 2

Ethernet's CSMA/CD (more)

Jam Signal: make sure all other transmitters are aware of collision; 48 bits

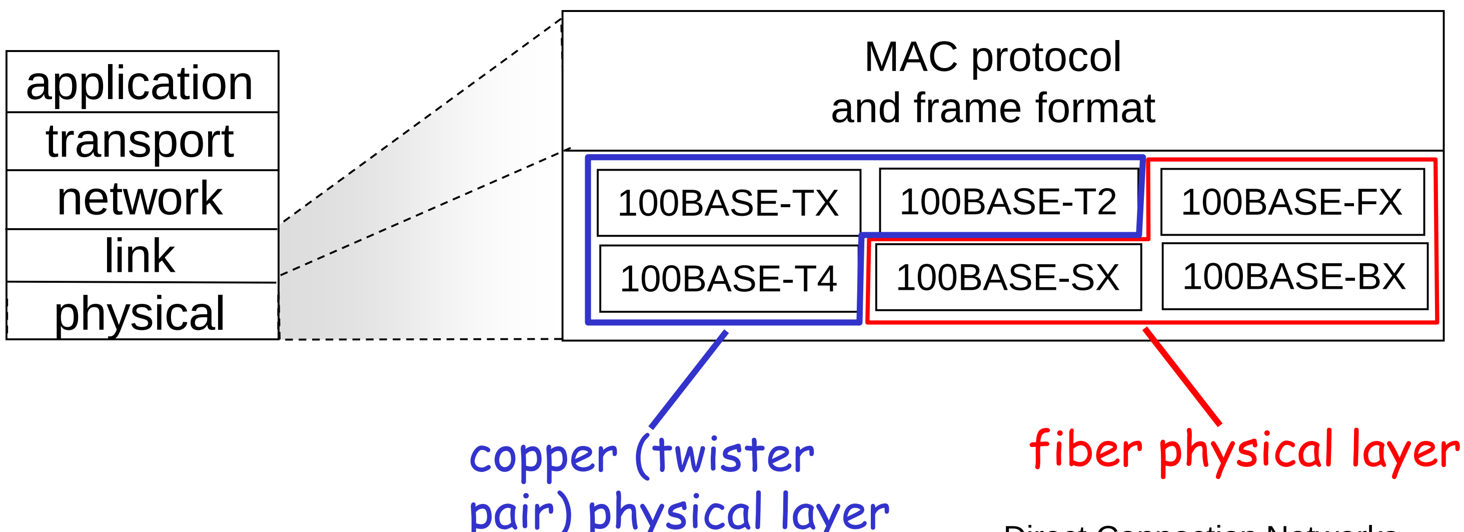
Bit time: .1 microsec for 10 Mbps Ethernet ;
for K=1023, wait time is about 50 msec

Exponential Backoff:

- *Goal*: adapt retransmission attempts to estimated current load
  - heavy load: random wait will be longer
- first collision: choose K from {0,1}; delay is K· 512 bit transmission times
- after second collision: choose K from {0,1,2,3}…
- after ten collisions, choose K from {0,1,2,3,4,…,1023}

802.3 Ethernet Standards: Link & Physical Layers

- *many* different Ethernet standards
  - common MAC protocol and frame format
  - different speeds: 10 Mbps, 100 Mbps, 1Gbps, 10G bps
  - different physical layer media: fiber, cable



Summary

- Direct connection networks
- principles behind Data Link layer services:
  - error detection, correction
  - reliable data transfer
  - sharing a broadcast channel: multiple access
  - link layer addressing
- instantiation and implementation of various link layer technologies
  - PPP
  - Ethernet