Direct Link Communication I: Basic Techniques

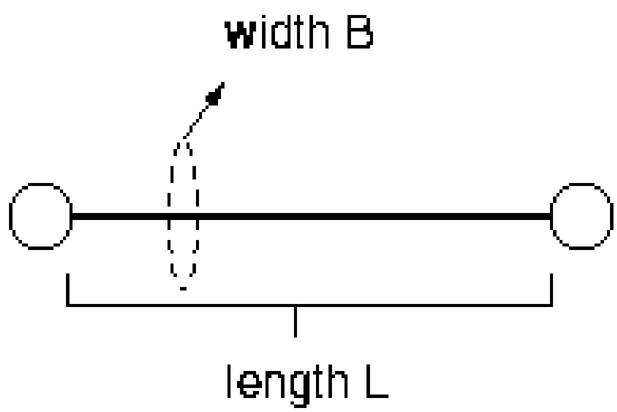
Data Transmission

Link speed unit: bps

⟶ abstraction

⟶ ignore carrier frequency, coding etc.

Point-to-point link:



⟶ wired or wireless

⟶ includes broadcast case

Interested in *completion time*:

$\longrightarrow$ time elapsed between sending/receiving first bit

- Single bit:

  $\rightarrow \approx L/\mathrm{SOL}$ (lower bound)

  $\rightarrow$ latency (or propagation delay)

  $\rightarrow$ optical fiber, wireless: exact

- Multiple, say S, bits:

  $\rightarrow \approx L/\mathrm{SOL} + S/B$

  $\rightarrow$ latency + transmission time

Latency vs. transmission time: which dominates?

$\longrightarrow$ a lot to send, a little to send, ...

$\longrightarrow$ satellite, Zigbee, WLAN, broadband WAN

Reliable Transmission

Principal methodology: ARQ (Automatic Repeat reQuest)

$\longrightarrow$ use retransmission

$\longrightarrow$ used in both wired/wireless

- function duplication

  $\rightarrow$ link layer, transport layer, etc.

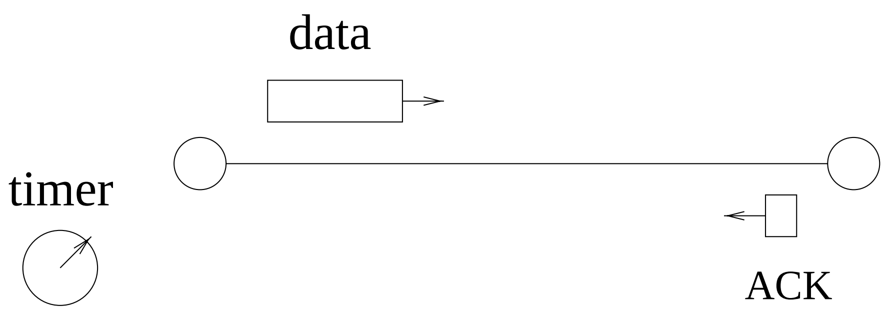
- alternative: FEC

  $\rightarrow$ not assured

  $\rightarrow$ hybrid schemes

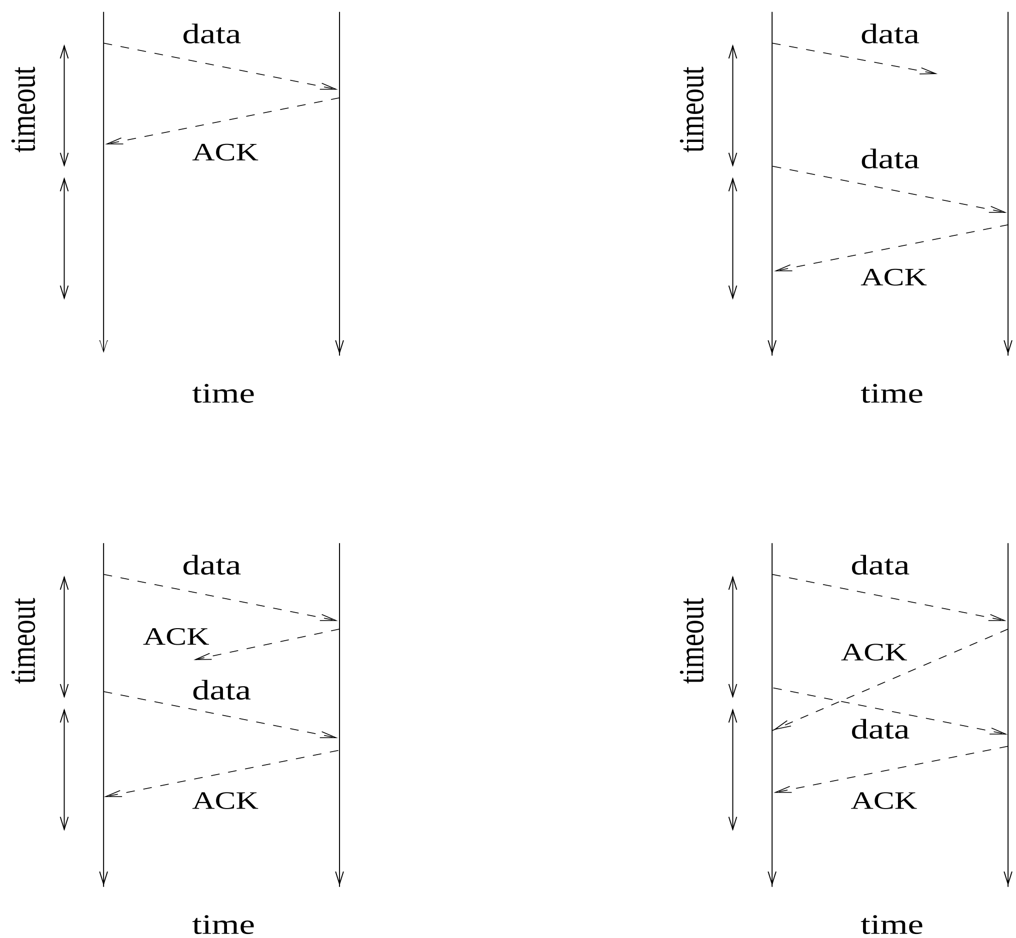
Three components:

- timer
- acknowledgment (ACK)
- retransmit



Stop-and-Wait

Assumption: Frame is "lost" due to corruption; discarded by NIC after error detection.



Issue of RTT (Round-Trip Time) & timer management:

- what is proper value of timer?

  → RTT estimation

- easier for single link

  → RTT is more well-behaved

- more difficult for multi-hop path in internetwork

  → latency + queueing effect

Another key problem: not keeping the pipe full.

$\longrightarrow$ delay-bandwidth product

$\longrightarrow$ volume of data travelling on the link

High throughput: want to keep the pipe full

Stop-and-wait throughput (bps):

- RTT
- frame size (bits)

$\longrightarrow$ throughput = frame size / RTT

Ex.: Link BW 1.5 Mbps, 45 ms RTT

- delay-bandwidth product:

  $\rightarrow$ 1.5 Mbps $\times$ 45 ms $=$ 67.5 kb $\approx$ 8 kB

- if frame size 1 kB, then throughput:

  $\rightarrow$ $1024 \times 8/0.045 = 182$ kbps

  $\rightarrow$ utilization: only 182 kbps/1500 kbps $= 0.121$

Solution: increase frame size

- brute increase of frame size can be problematic

  $\rightarrow$ bully problem

  $\rightarrow$ existing LAN frame standards (legacy compatible)

- send blocks of data, i.e., sequence of frames

Sliding Window Protocol

⟶ send window/block of data

Issues:

- Shield application process from reliability management chore

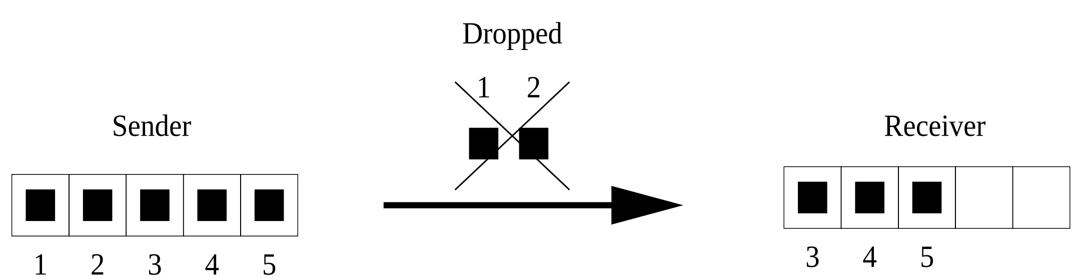
  → exported semantics: continuous byte stream

  → simple app abstraction: e.g., `read` system call

- Both sender and receiver have limited buffer capacity

  → efficiency: space-bounded computation

  → task: "plug holes & flush"



Simple solution when receiver has infinite buffer capacity:

- sender keeps sending at maximum speed
- receiver informs sender of holes

  → i.e., negative ACK
- sender retransmits missing frames

  ⟶ sender's buffer capacity?

  ⟶ need for positive ACK?

With finite buffer:

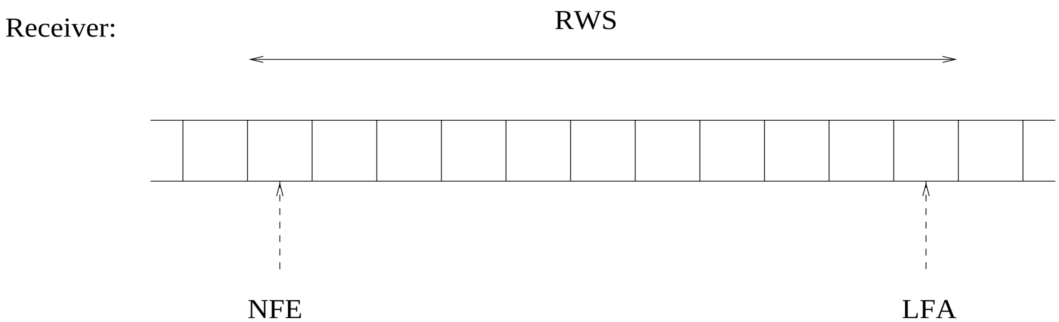
⟶ issue of bookkeeping

Flow control & congestion control:

→ sending too much is counterproductive

→ regulate sending rate

Set-up:



- *SWS*: Sender Window Size (sender buffer size)
- *RWS*: Receiver Window Size (receiver buffer size)
- *LAR*: Last ACK Received
- *LFS*: Last Frame Sent
- *NFE*: Next Frame Expected
- *LFA*: Last Frame Acceptable

Assign sequence numbers to frames.

$\longrightarrow$ IDs

Maintain invariants:

- $\text{LFA} - \text{NFE} + 1 \leq \text{RWS}$
- $\text{LFS} - \text{LAR} + 1 \leq \text{SWS}$

Sender:

- Receive ACK with sequence number X
- Forwind LAR to X
- Flush buffer up to (but not including) LAR
- Send up to $\text{SWS} - (\text{LFS} - \text{LAR} + 1)$ frames
- Update LFS

Receiver:

- Receive packet with sequence number Y
- Forwind to (new) first hole & update NFE

  $\longrightarrow$ NFE need not be $Y + 1$
- Send cumulative ACK (i.e., NFE)
- Flush buffer up to (but not including) NFE to application
- Update LFA $\leftarrow$ NFE + RWS $- 1$

ACK variants:

- piggyback
- negative ACKs
- selective ACKs

Sequence number wrap-around problem:

$$\text{SWS} < (\text{MaxSeqNum} + 1)/2.$$

$\longrightarrow$ note: stop-and-wait is special binary case

Direct Link Communication II: Wired Media

Multi-Access Communication

Two classes:

- contention-based

  → e.g., CSMA/CD, CSMA/CA

  → used in Ethernet, WLAN

- contention-free

  → e.g., TDM, FDM, TDMA, CDMA, token ring

  → one more method?

  → used in telephony and broadband data networks

⟶ also called MAC (medium access control)

- broadband: FDM, TDMA, CDMA
- baseband: TDM, multiple access

Contention-based MAC for baseband:

⟶ keep in mind discussion group

⟶ how to keep discussion orderly?

Features:

- Time slots are available for grab

  $\longrightarrow$ "on-demand" TDM
- Can listen to channel activity...
- To grab channel slot is to send

  $\longrightarrow$ shoot-first-ask-later (e.g., TV talk shows)
- If ≥ 2 users grab at the same time, slot becomes junk

  $\longrightarrow$ collision

Why not just use TDM?

Benefits of contention-based MAC:

- when not too many users, faster response time

  → don't need to go through registration & reservation phase (TDM)

  → avoids admission control overhead

- decentralized

  → no central coordinator

  → simple; "self-organization"

Drawbacks of contention-based MAC:

- when many users, degraded response & throughput

  → collision wastes slots, i.e., bandwidth

- lack of QoS (quality of service) assurances

  → "you get is what you get"; best effort

  → problematic for real-time traffic, e.g., telephony

Thus when to use what?