Transport / Network layer error detection

- typically implemented in software, so need simple & fast schemes
- <u>Internet checksum</u> used by IP, TCP, UDP
  - bytes of data treated as 16-bit integers & summed w̄ 16 bit 1s complement arithmetic
  - 1s complement of this sum is the Internet checksum

eg.

0110 0110 0110 0000	(1)
0101 0101 0101 0101	(2)
1000 1111 0000 1100	(3)

- add values 2 at a time, adding any overflow back into the LSB

```
   0110 0110 0110 0000     (1)
 + 0101 0101 0101 0101     (2)
 ---------------------
   1011 1011 1011 0101     no overflow
 + 1000 1111 0000 1100     (3)
 ---------------------
 1 0100 1010 1100 0001
 +                   1     add overflow
 ---------------------
   0100 1010 1100 0010

   1011 0101 0011 1101     invert all bits
                           to obtain 1s
                           Complement
```

(other implementations may be more efficient depending on the machine)

- checksum is transmitted in packet header
- receiver repeats calculation on received data & compares with checksum (or equivalently adds data + checksum with 16 bit 1s complement arithmetic & checks that the result is 1111 1111 1111 1111)
- In TCP & UDP, the Internet checksum is computed over all fields (header + data)

  In IP, the checksum is computed over the IP header
- relatively weak protection against errors when compared with CRC
- although many link layer protocols provide error detection, no guarantee that all links do; also bit errors may be introduced when a packet is stored in a router's memory → transport layer error detection still needed but can be more lightweight since most errors are picked up by stronger error detection at link level

- although a pure end-to-end argument might suggest that link layer error detection is superfluous, it allows for functions to be incompletely provided at lower levels as a performance optimization (eg it is a waste to transmit a corrupted packet all the way to the end host before detecting the error)

Reliable Transmission

- consider problem of reliable transmission across an unreliable link
- usually accomplished using a combination of 2 fundamental mechanisms — acknowledgments & timeouts
  - receipt of an ACK indicates to the sender of the original frame that it was successfully received
  - otherwise, after a period of time without receiving an ACK (timeout) the sender retransmits
  - general strategy is called automatic repeat request (ARQ)
  - error detection code used
- alternative — forward error correction (FEC) using error-correcting code (more redundancy)
  - useful when errors occur frequently, eg in wireless, or retransmission latency is unacceptable

ARQ strategies

- **Stop-&-Wait**
  - after transmitting a frame, sender waits for an ack before transmitting the next frame
  - if ack is not received within the timeout period, the sender retransmits
  - main disadvantage: can only have 1 outstanding frame at a time, which may be far below link capacity

    eg. a 1.5 Mbps link with a 45 ms RTT
    - delay × bandwidth product = 67.5 Kb
    - if only 1 frame of size 1 KB is sent per RTT, rate $= \dfrac{\text{bits per frame}}{\text{time per frame}}$

$$\leq \frac{1024 \times 8}{0.045}$$

$$= 182 \text{ Kbps}$$

$$\approx \frac{1}{8} \times \text{link capacity}$$

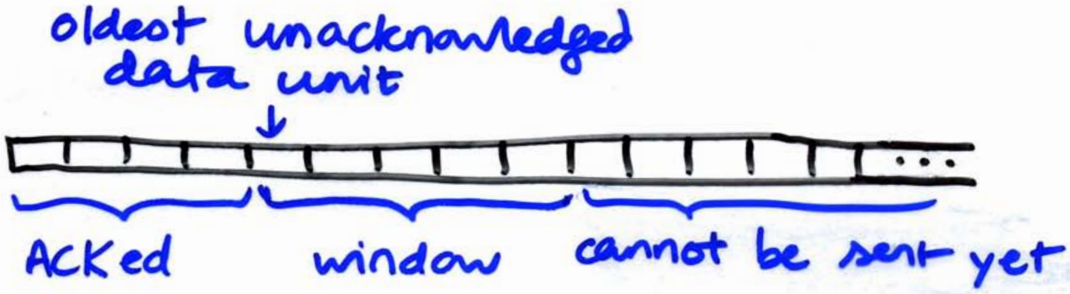
    - to keep the pipe full, sender should be able to transmit 8 frames before having to wait for an ack

- Concurrent logical channels eg. ARPANET

data link protocol

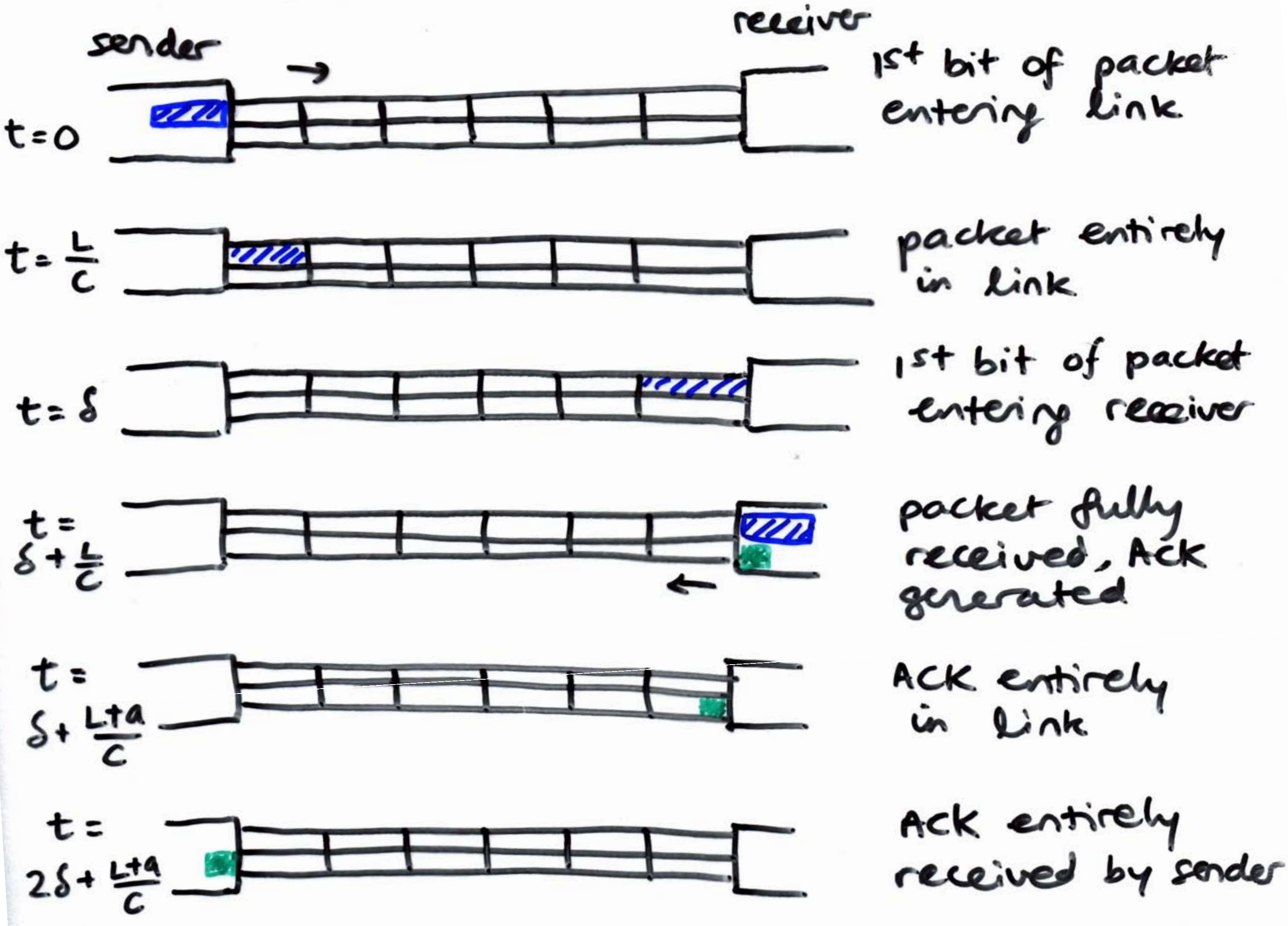
- multiplex several logical channels onto a single point-to point link
- run stop-&-wait algorithm on each logical channel
- when sender has a frame to send, it uses the lowest idle channel
- does not preserve frame order

- **Sliding Window** eg TCP, X.25 link layer
  - limit the amount of unacknowledged data transmitted — limit is called the transmission <u>window</u> W
  - when W units of data are outstanding, the sender stops & waits for an ACK



  - left edge of window corresponds to oldest unacknowledged data unit; if ACKed, window shifts one unit to the right
  - for a fixed window size W, packet transmission rate $= \dfrac{W}{RTT} \geq$ goodput (actual data rate) since there are retransmissions usually

- consider transmission of a length-L packet & a length-a ACK over a link of capacity C & propagation delay δ



- Base RTT (time taken for a packet to be sent & its ACK received) $= 2\delta + \frac{L+a}{C}$
- BRTT in terms of packet transmission time $\frac{L}{C}$

$$= \frac{2\delta C}{L} + 1 + \frac{a}{L} \approx \underbrace{\frac{2\delta C}{L}}_{} + 1 \quad \text{if } a \ll L$$

bandwidth × (RT propagation) delay product <u>in packets</u> $:= \Delta$

- for a single transmitter to use full link capacity, window size should ideally be $\Delta + 1$ packets

full round-trip pipe if no losses

- forward link contains $\frac{\Delta}{2}$ back-to-back packets not yet received
- receiver has received $\frac{\Delta}{2} + 1$ packets
- reverse link contains $\frac{\Delta}{2}$ ACKs spaced by packet times

- multi-hop: sender does not know Δ, nor how many other transfers are sharing the pipe
  → must learn the correct window size adaptively

- even if optimal window size is known, if the sender is not directly connected to the bottleneck link, it should not start by sending a full window of packets all at once → packets would pile up at the bottleneck & might be dropped if buffer space is insufficient

→ transmitter should increase window size gradually (slow-start)

TCP service model

- <u>connection-oriented service</u>
  - client & server exchange transport-layer control info with each other before starting to exchange application-layer messages (handshaking)
  - after handshaking, a <u>TCP connection</u> exists between the sockets of the 2 processes
  - full-duplex connection (both processes can send messages to each other simultaneously)
  - when application finishes sending the connection is torn down
- <u>reliable & sequential packet transport service</u>
  - communicating processes can rely on TCP to correctly deliver all data in order to the receiving socket
- TCP also includes flow control & congestion control
- <u>Secure Sockets Layer</u>: application-layer enhancement to TCP providing security services

- TCP runs the sliding window protocol end-to-end over the Internet
  - need an <u>explicit connection establishment</u> phase to exchange parameters & establish some shared state, & a <u>connection teardown</u> phase to let each host know it is OK to free this state
  - RTTs vary widely across different connections & even during a connection
    → need <u>adaptive timeout</u>
  - packets may be reordered or substantially delayed
    → TCP assumes a <u>maximum segment lifetime</u> MSL (estimate of how long a packet might live in the Internet), 120s
  - Internet hosts vary widely
    → each host needs to learn the other side's receiving buffer space (<u>flow control</u>)
  - links traversed en route to destination may vary widely in capacity / congestion
    → unlike the case of a single link, congestion is not directly visible to sender in the form of queue buildup at the sender
    → need <u>congestion control</u> mechanisms