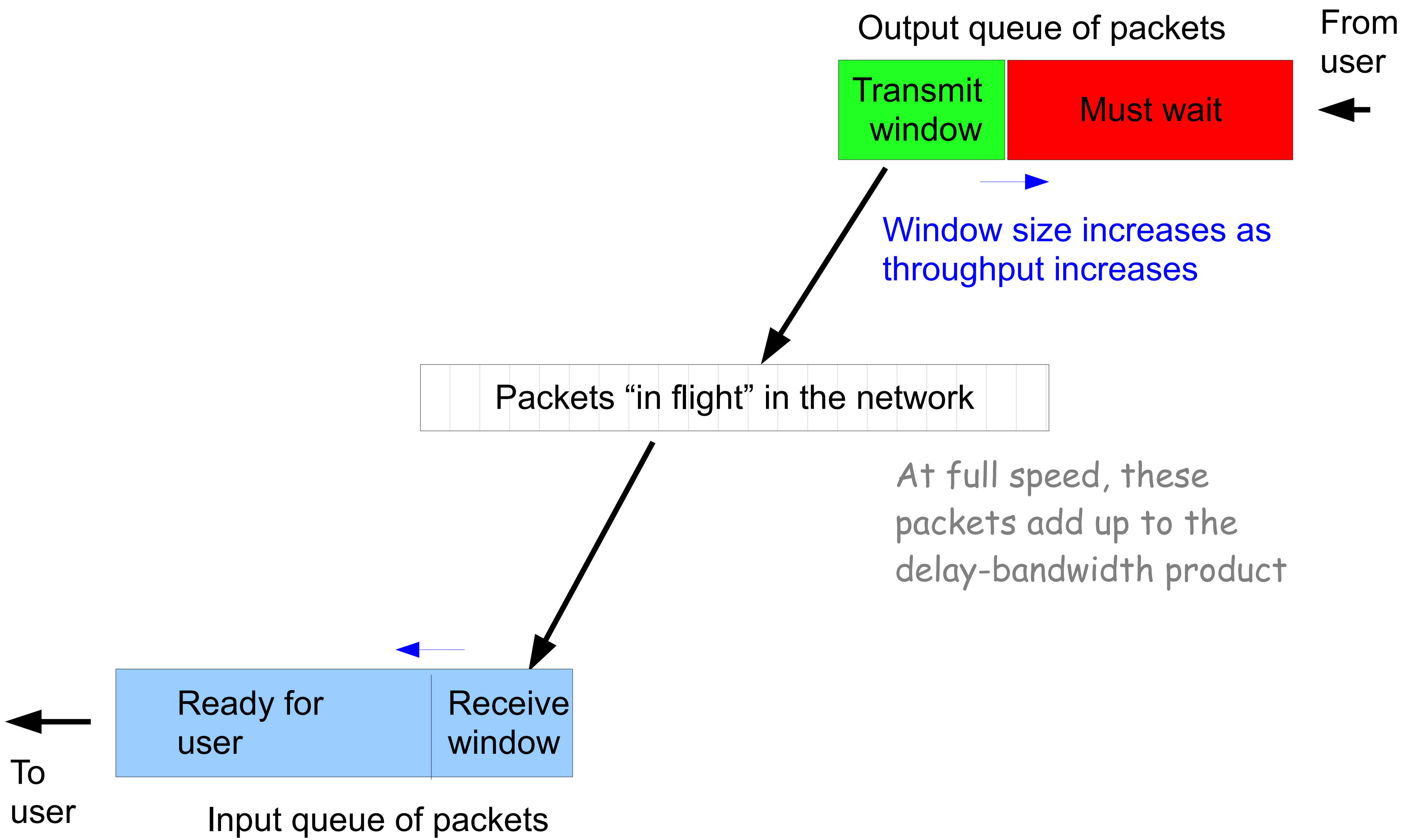
TCP: Transmission Control Protocol

- IP is an *unreliable* datagram protocol
  - congestion or transmission errors cause lost packets
  - multiple routes may lead to out-of-order delivery
- If senders send too fast, routers or receivers cannot keep up (making congestion worse)
- When many senders compete, capacity must be fairly shared
- TCP's job is to fix those three problems
  - flow control
  - retransmission after errors

Two approaches to Flow Control

- Rate Control - sender determines the maximum safe sending rate and never exceeds it
- Sliding Window - sender sends up to a “window” full of data but then pauses for an acknowledgement
  - Window size is adjusted dynamically to match network capacity
    - Window size is also known as "credit"
  - Missing acknowledgement causes retransmission
- TCP is a sliding window protocol

Sliding windows in action



Cranking up to speed

- TCP starts slowly
  - Initial window size is small
  - Send packets until window is empty
  - Increase window size as data flow accelerates
  - Decrease window size if data flow slows down
  - Retransmit when acknowledgments don't arrive
- Note that when A is talking to B and B is talking to A, the paths may be asymmetric, so TCP windows work independently in the two directions
- Note that TCP transmits a stream of *bytes* as far as user programs are concerned, broken up into *segments* by TCP itself

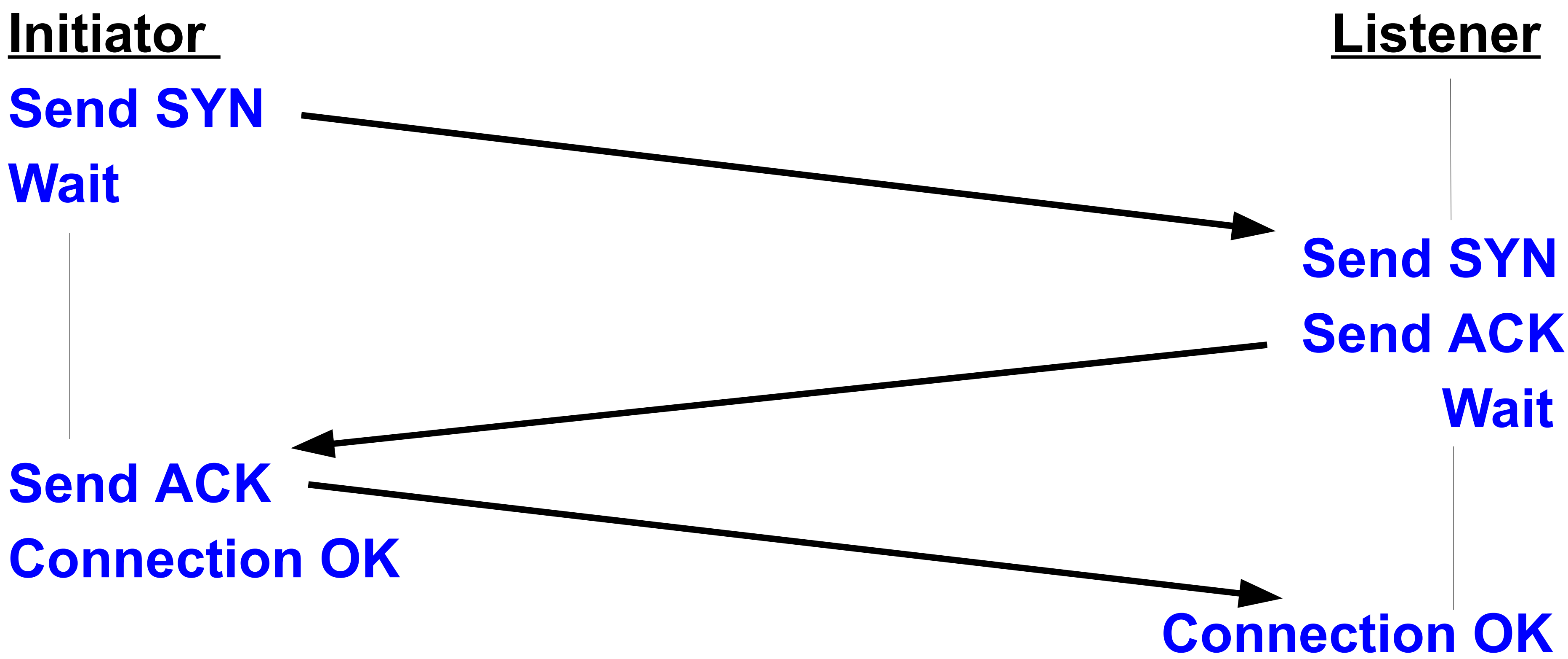
Why is this better than rate control?

- The sliding window approach works over an enormous range of speeds
  - It was designed in the days of 9600 baud modems, and it works (with some tuning) in the days of 10 Gb/s links
  - Rate control works best in fixed-speed networks
- It works reasonably well as router load increases towards 100%
  - Sharing between thousands or millions of competing TCP sessions is reasonably fair
  - Rate control has real trouble sharing fairly at that scale
- Retransmission fits naturally into TCP
  - Rate control protocols have to “break step” to deal with retransmissions

TCP connection phases

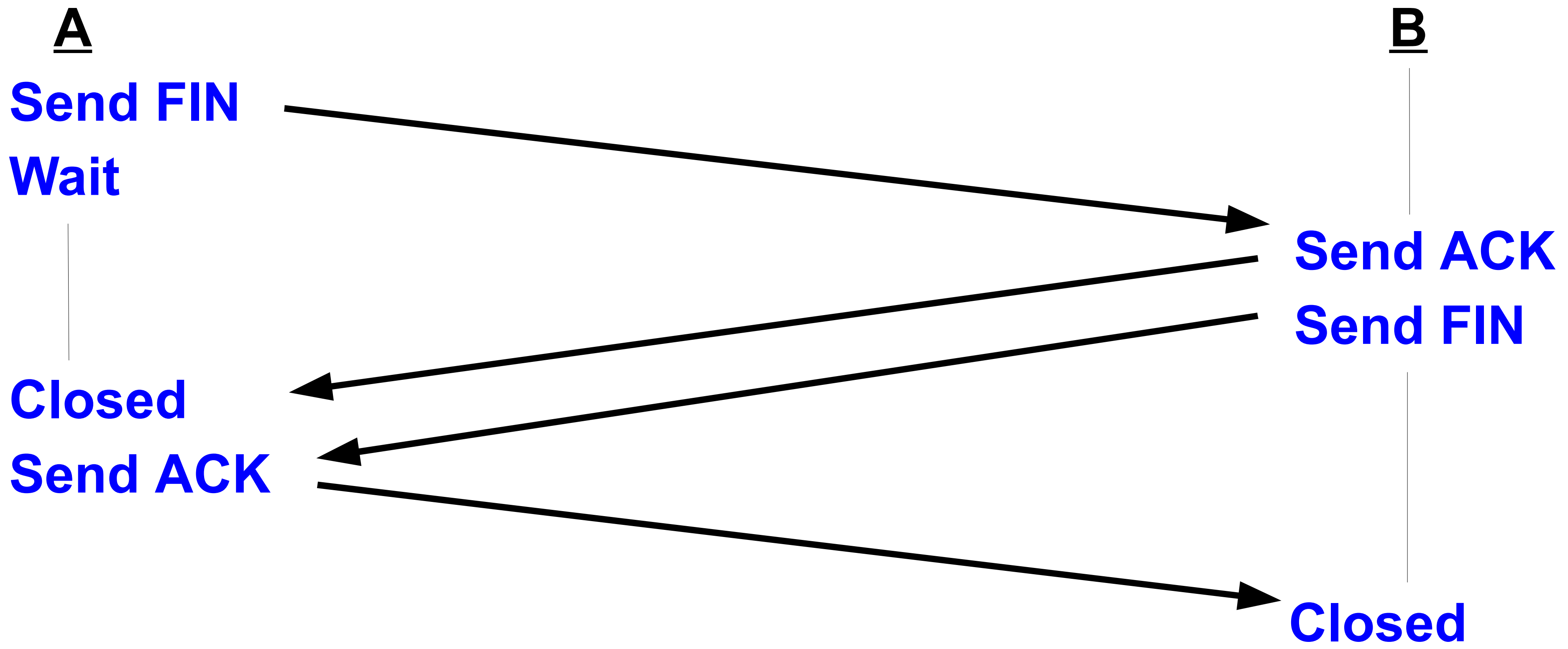
- A TCP connection has three main phases:
  - Establishment
  - Data transfer
  - Disconnection
- One end (the “listener”) has to be willing to accept incoming TCP connections, and the other end (the “initiator”) has to choose to start
- The listener is listening to a specific *port number* which serves as a meeting point
  - IP address + port number = *Layer 4* address

Connection establishment



- Note that whoever goes first, there is one SYN and one ACK in each direction
  - This will work even if both initiate simultaneously
  - Listener's SYN and ACK are usually sent in a single packet

Disconnection



- Doesn't matter which end closes first
  - If one end dies, a timeout will eventually close the other end
  - B may continue to send bytes from its buffer before sending its FIN

TCBs

- During connection establishment, each end creates a TCB (transmission control block) data structure
  - A TCB links the user program at each end to the TCP process
- Typical TCB contents:
  - local and remote port numbers for this connection
  - current send and receive window sizes
  - pointers into the send and receive buffers
  - status of send and receive sequence numbers
- To understand this we need to look at the TCP header format
  - The TCP header follows the IP header in a packet, when Protocol Number (IPv4) or Next Header (IPv6) is **6**

TCP header

```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|          Source Port          |       Destination Port        |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                        Sequence Number                        |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                    Acknowledgment Number                      |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|  Data |           |U|A|P|R|S|F|                               |
| Offset| Reserved  |R|C|S|S|Y|I|            Window             |
|       |           |G|K|H|T|N|N|                               |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|           Checksum            |         Urgent Pointer        |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                    Options                    | Zero Padding  |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                     data segment                            ...
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

- **Protocol Number or Next Header is 6**

TCP header fields (1)

- Port numbers - used to find TCBs at each end
- Sequence number
  - the sequence number of the first data byte in this TCP segment
  - goes up by 1 for each data byte sent on the connection
  - initialised in SYN packet (random value)
- Acknowledgement number
  - only valid in ACK packet
  - next sequence number the sender of the segment is expecting
  - in other words, sending Ack Number 12345 means “I have correctly received up to byte 12344”
  - a duplicate ACK means “I've **still** only received up to byte 12344”

TCP header fields (2)

- Data offset
  - Size of TCP header in 32 bit words
- URG - urgent bit (not too important)
- ACK - this is an ACK packet
- PSH - push bit (kick received data to the user)
- RST - reset bit (emergency disconnect)
- SYN - SYN packet (“synchronise sequence numbers”)
- FIN - FIN packet (“finished,” starts normal disconnect)

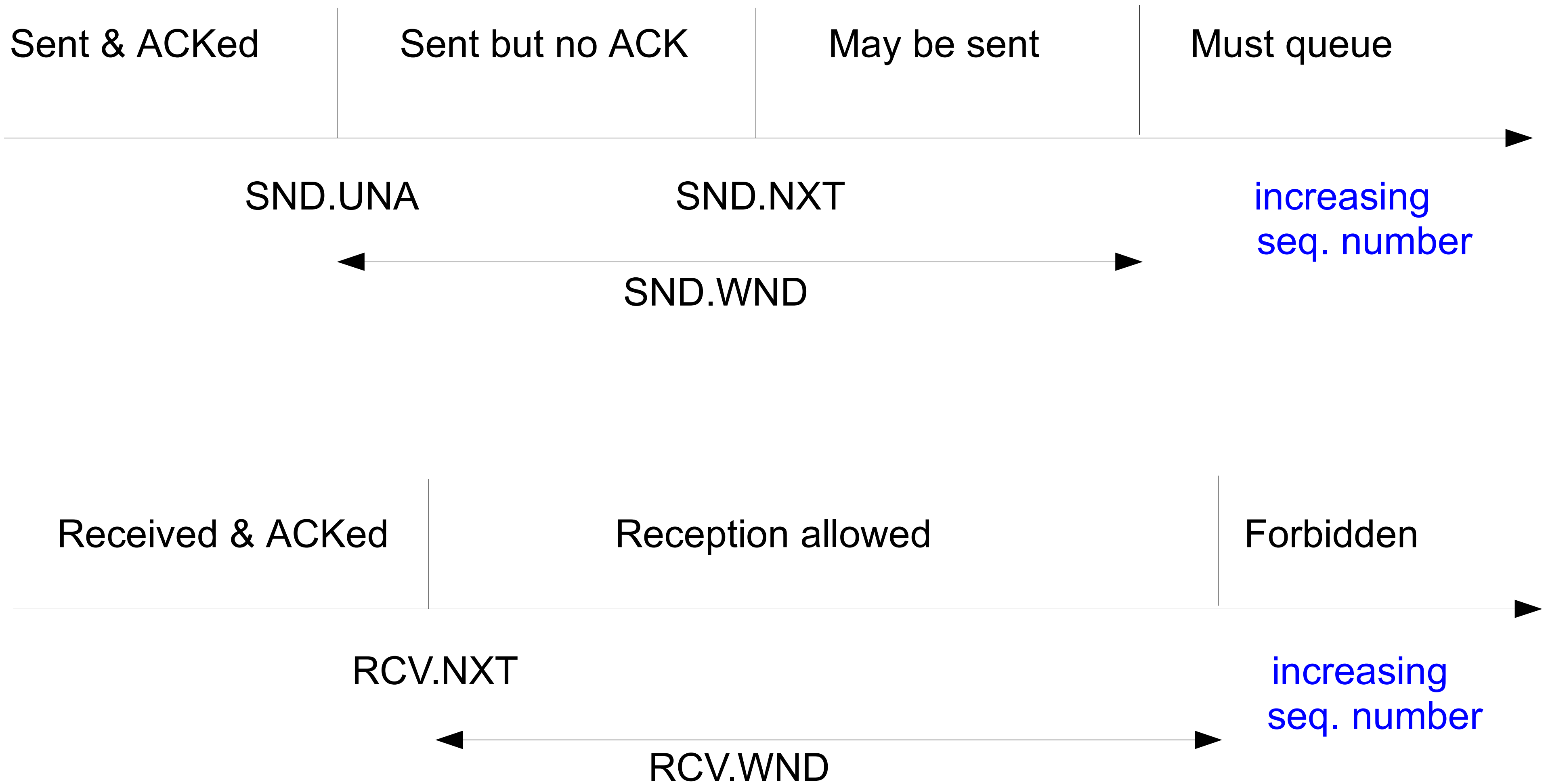
TCP header fields (3)

- Window
  - The number of data bytes beginning with the one indicated in the acknowledgment field which the sender of this segment is willing to accept
- Checksum (next slide)
- Urgent pointer (not too important)
- Options
  - For example, specify maximum receive segment size

TCP checksum

- *This is the primary protection against transmission errors in the Internet*
  - 16 bit one's-complement of the one's-complement sum of all 16 bit words in the TCP header and data
  - If a segment contains an odd number of bytes to be checksummed, the last byte is padded on the right with zeros to form a 16 bit word for checksum purposes. (The padding is not transmitted as part of the segment)
  - While computing the checksum, the checksum field itself is replaced with zeros
  - The checksum also covers a “pseudo header” conceptually prefixed to the TCP header. This pseudo header contains the Source & Destination IP Addresses, the Protocol or Next Header Number, and TCP segment length

Sequence number state at sender and receiver



TCP data transfer phase

- After SYN/ACK, the two ends know initial sequence numbers and initial window sizes
- Both ends may start sending, as long as they stay within the allowed sending window
  - sending a segment moves SND.NXT along
  - receiving an ACK for a given sequence number moves SND.UNA along
  - if SND.NXT = SND.UNA+SND.WND, wait
- Both ends receive
  - when a segment arrives, increase RCV.NXT and send ACK
  - if RCV.NXT reaches end of window (i.e. RCV.WND=0), only ACKs will be treated. Incoming data is discarded and not ACKed

Adjusting the Window

- The size of the send window (SND.WND) decides how much data can be sent without waiting for an ACK
  - SND.WND must be decreased when things are going slowly, and can be increased when things are going well
  - SND.WND tracks RCV.WND via ACK messages
  - The algorithm for adjusting RCV.WND is the most critical feature of a TCP implementation and has been modified many times
- See comments below on congestion control

Background slide

Algorithm for Send Window to track Receive Window

- Variables

  `SND.NXT` - next sequence number to be sent

  `SND.WND` - current send window size

  `LatestAckSeq` - acknowledgement number in latest ACK

  `CurrentSeq` - sequence number of segment carrying ACK

  `AckWindow` - receiver's window size in ACK

  `PreviousSeq, PreviousAck` - from previous window update

- Algorithm

```
if LatestAckSeq <= SND.NXT then      # waiting for ACKs
if (PreviousSeq < CurrentSeq or      # don't use stale
   (PreviousSeq == CurrentSeq        #   window size
    and PreviousAck <= LatestAckSeq))
then {SND.WND := AckWindow;         # update window
      PreviousSeq := CurrentSeq;
      PreviousAck := LatestAckSeq;}
```

Retransmission

- If an ACK does not arrive within a certain timeout, all segments since the previous ACK will be retransmitted
  - no difference whether packet was discarded due to congestion or lost due to transmission fault or checksum error
  - can be optimised with “Selective ACK” to avoid retransmitting correctly received segments
- The retransmission timeout is dynamically calculated
  - Typically by measuring a running average Round Trip Time (RTT) between sending a segment and receiving its ACK
  - Then set the timeout to, say, 2xRTT

Clarification about delay-bandwidth product

- The one-way delay in a TCP session is roughly half the RTT
- Therefore, the delay-bandwidth product is roughly

$$\frac{\textit{bandwidth x RTT}}{2}$$

- The TCP window size in a stable state is roughly

$$\textit{bandwidth x RTT}$$

  which is double the delay-bandwidth product, because the window has to allow for ACKs to come back
  - 'roughly' because the outbound delay (for data) and the return (for ACKs) will never be exactly equal
  - some web references get this wrong

- *bandwidth* = link transmission rate (b/s)

Congestion control

- TCP as described above is “greedy” - it will pump as much data as the path will take
  - With millions of connections, this leads to “congestive collapse” where saturated routers must discard most packets
- Modern TCPs use various techniques to avoid this, all of which amount to being “good neighbours”
  - Slow Start: start small and expand window gently
  - Congestion Avoidance: when duplicate ACKs indicate that later segments were lost, limit number of (re)transmissions
  - Fast Recovery: after 3 duplicate ACKs, retransmit once and wait. If still no ACK, revert to Slow Start
- Modern routers keep an eye out for greedy “cheats” and selectively discard their packets

A day in the life of a TCP session

- User A: Listen (portA)
- User B: Open (AddressA, portA)
- SYN/ACK exchange
- Data transfer phase
  - User A: Send (DataA)
  - User B: Receive (DataA)
  - User B: Send (DataB)
  - User A: Receive (DataB)
  - (repeat as required by application)
- User B: Close
- FIN/ACK exchange
- User A: Listen (portA)

Retransmission, windowing, and congestion control as needed

References

- Shay 11.4
- Any of the TCP/IP books listed for IPv4
- RFCs:
  - RFC 793, the original definition
  - Many advisory RFCs and other publications on implementation techniques to tune performance. (Implementing TCP is not for amateurs!)
  - RFC 2460 (IPv6) modifies TCP's checksum formula
  - RFC 3168 adds *Explicit Congestion Notification* to TCP and IP
  - RFC 4614 is a roadmap for TCP specifications