

CS314s2- 27

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

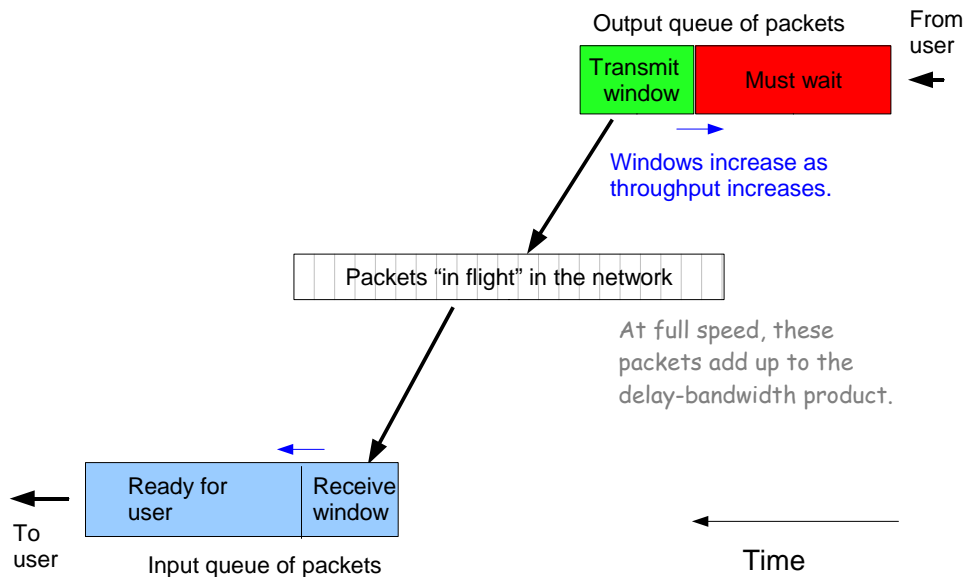
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Two approaches to flow control

- 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.

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Sliding windows in action



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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.

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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 Gbit 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

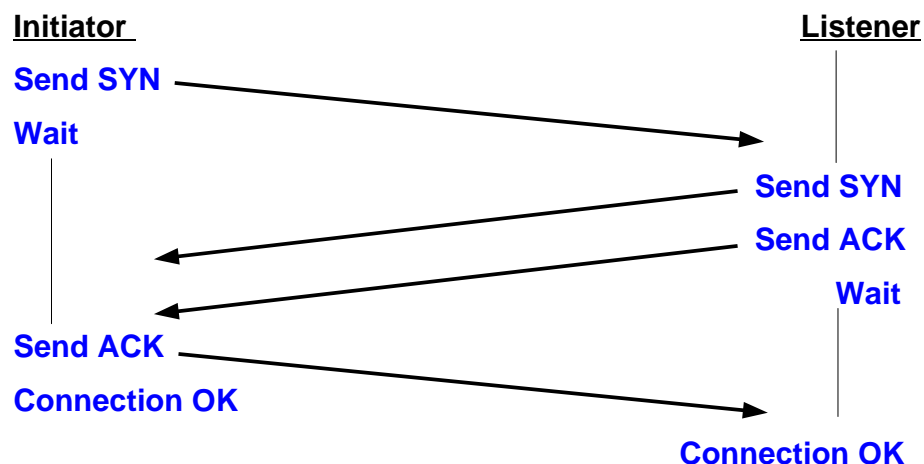
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TCP connection phases

- 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

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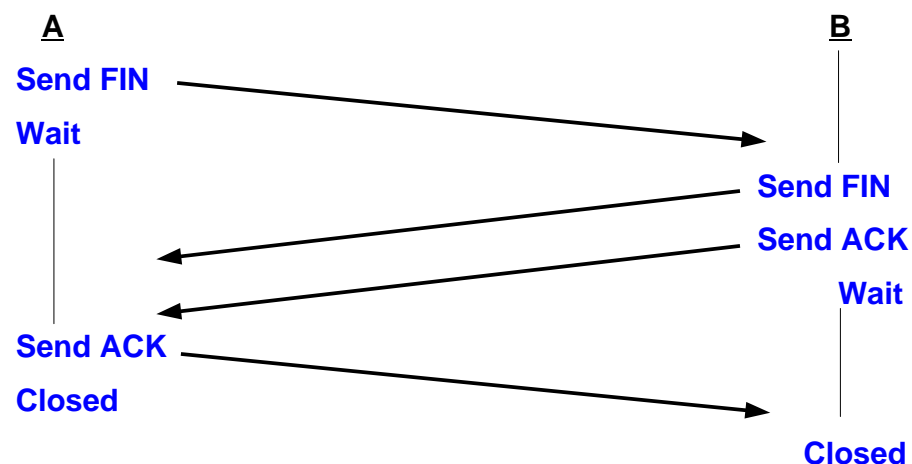
Connection establishment



- Note that whoever goes first, there is one SYN and one ACK in each direction.
- This will work even if both initiate simultaneously

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Disconnection



- Doesn't matter which end closes first.
 - If one end dies, a timeout will eventually close the other end.

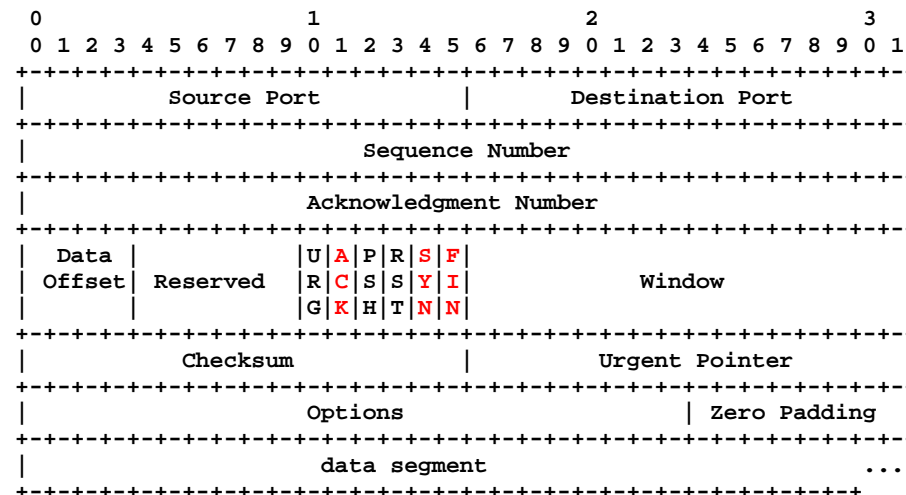
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TCBs

- 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**.

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TCP header



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

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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.
- 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."

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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)

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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.

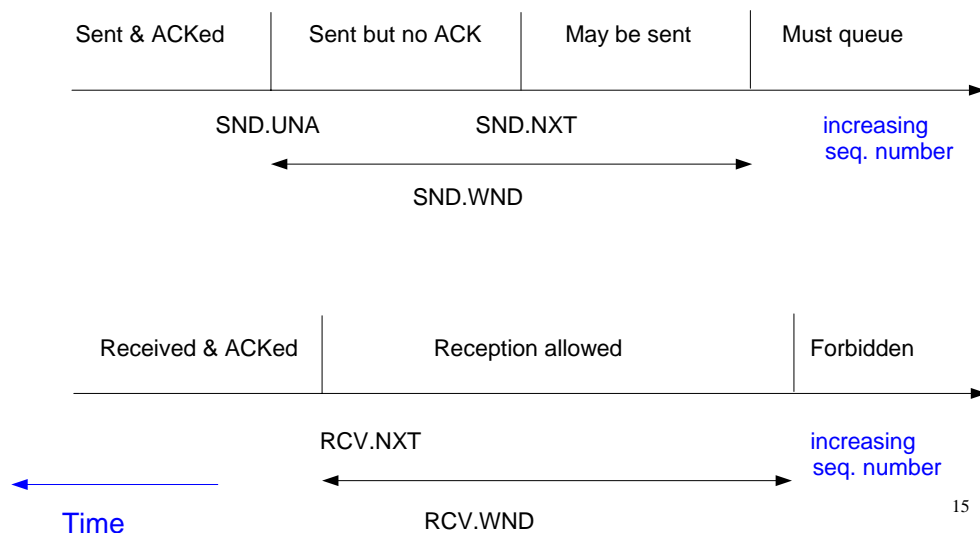
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TCP checksum

- 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 pad 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.

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Sequence number state at sender and receiver



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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.

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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

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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;}
```

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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.

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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

$$\frac{\text{throughput} \times \text{RTT}}{2}$$

- The TCP window size in a stable state is roughly

$$\text{throughput} \times \text{RTT}$$

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.

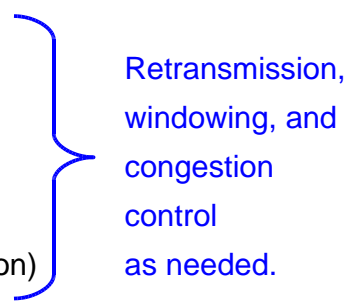
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Congestion control

- 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.

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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.

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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 checksum formula.
 - RFC 3168 adds *Explicit Congestion Notification* to TCP and IP.
 - RFC 4614 is a roadmap for TCP specifications.

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