Reliable Data Delivery

Lecture 13

http://www.cs.rutgers.edu/~sn624/352-F24

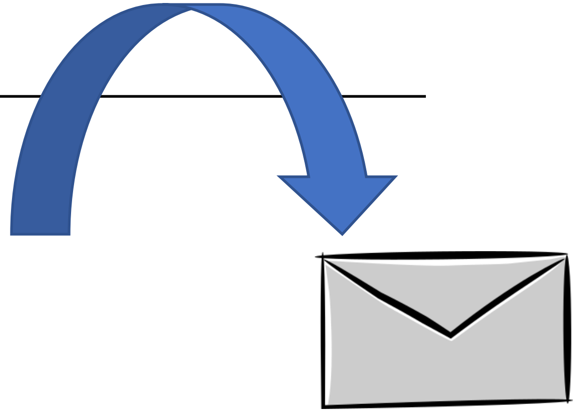
Srinivas Narayana

RUTGERS
UNIVERSITY | NEW BRUNSWICK

Review

- UDP: best-effort delivery + demultiplexing + error detection
- Checksum function: 1s complement of the 1s complement sum
- Sender: compute checksum & write.
  - Receiver: compute checksum, compare to 0

	1 1 1 0 0 1 1 0 0 1 1 0 0 1 1 0
	1 1 0 1 0 1 0 1 0 1 0 1 0 1 0 1
wraparound	1 1 0 1 1 1 0 1 1 1 0 1 1 1 0 1 1
sum	1 0 1 1 1 0 1 1 1 0 1 1 1 1 0 0
checksum	0 1 0 0 0 1 0 0 0 1 0 0 0 0 1 1



https://datatracker.ietf.org/doc/html/rfc1071

Not on the exam

Checksum, why you being weird?

- Need a function that is fast to compute, catches likely errors, and easy to verify. Some design considerations:
- Basic bit-wise: AND, OR: many inputs map to the same output
- XOR: can catch single bit-flips, but not an even number of 1s/0s flipping
  - Some sort of addition is preferable to this (carries will show errors)
- Addition is commutative, associative, has an identity element (0), is efficient to calculate
  - Checksum can appear anywhere in the packet
  - Compute checksum by placing a 0 in place originally
  - Use operations at the natural bit-width of the machine (16 bits was common)
- (Regular) two's complement addition: errors in higher order bit positions can be missed (the final carry-out bit isn't part of the checksum)
  - One's complement: adding the final carry-out to the result helps ☺
- Why complement? Why not compare the checksum rather than to 0?
  - CPUs have ways of detecting if the last result was 0

Some observations on checksums

- Checksums don’t detect all bit errors
  - Consider (x, y) vs. (x – 1, y + 1) as adjacent 16-bit values in packet
  - Analogy: you can’t assume the package hasn’t been meddled with if its weight matches the one on the stamp. More smarts needed for that. ☺
  - But it’s a lightweight method that works well in many cases

- Checksums are part of the packet; they can get corrupted too
  - The receiver will just declare an error if it finds an error

Some observations on checksums

- Checksums are insufficient for reliable data delivery
  - If a packet is lost, so is its checksum
- UDP and TCP use the same checksum function
  - TCP also uses the lightweight error detection capability
  - However, TCP has more mature mechanisms for reliable data delivery (up next!)
- Checksum is a mechanism to detect errors, not correct them
  - Even when they detect errors, checksums don’t tell you where they lie

Playing with checksums

- Let’s craft some UDP packets (again)!

- `sudo tcpdump -i lo udp –XAvvv # observe packets`
- `sudo scapy # tool used to send crafted packets`
- `send(IP(dst="127.0.0.1")/UDP(sport=1024, dport=2048)/"hello world”, iface="lo")`

- Now can you craft two UDP packets with an identical checksum?

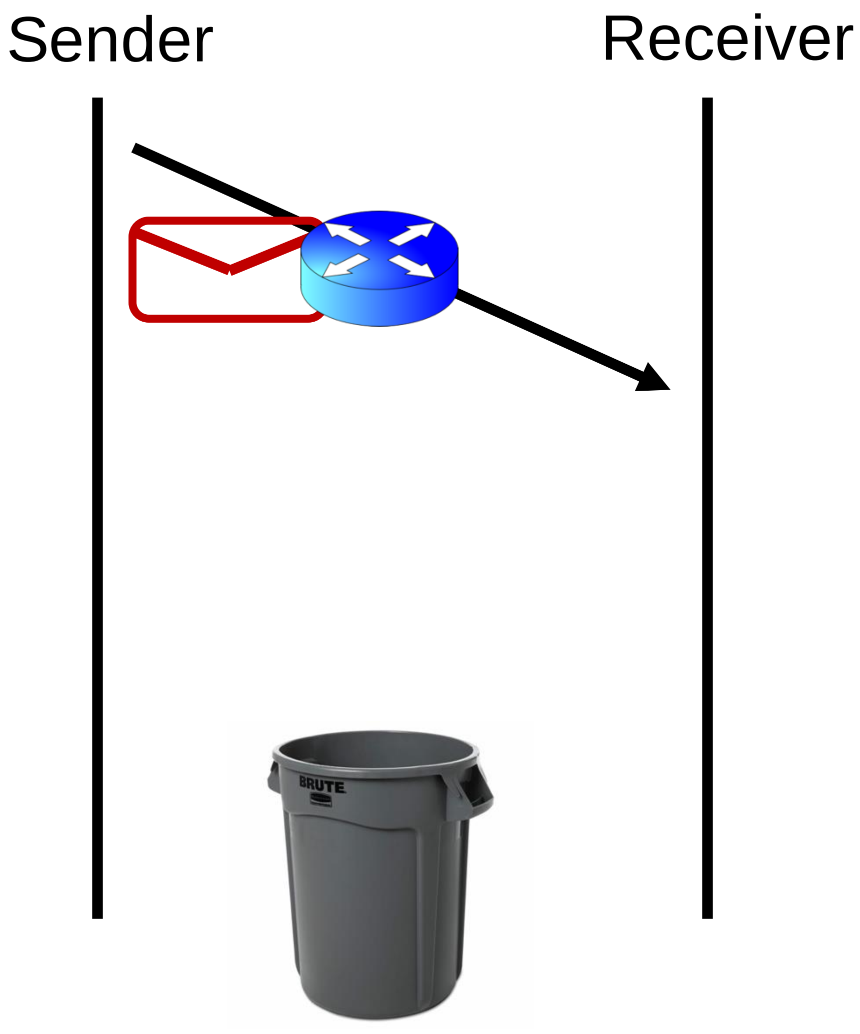
Summary of UDP

- A simple transport: Send or receive a single packet from/to the correct application process. That's it
  - Just a thin shim around network layer's best-effort delivery
  - No connection building, no latency
  - Suitable for one-off request/response messages
  - Sometimes suitable for loss-tolerant but delay-sensitive applications

- No reliability, performance, or ordering guarantees
- Can do basic error detection (bit flips) using checksums
  - Error detection is necessary to deliver data reliably, but it is insufficient

Reliable data delivery

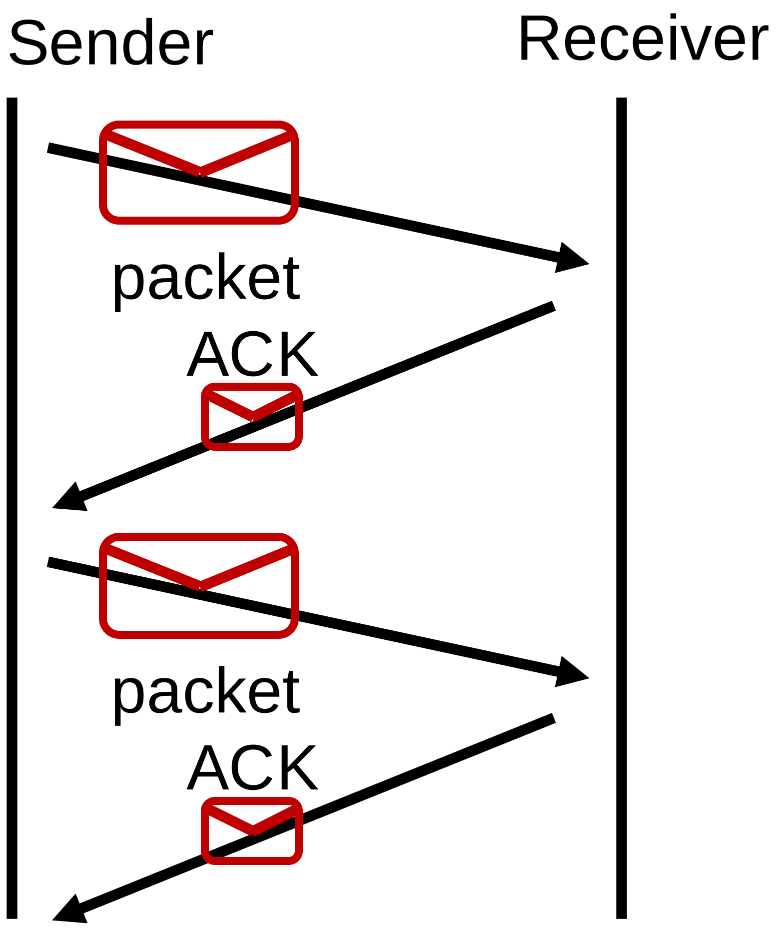
Packet loss



- How might a sender and receiver ensure that data is delivered reliably (despite some packets being lost)?
- TCP uses three mechanisms

Coping with packet loss: (1) ACK

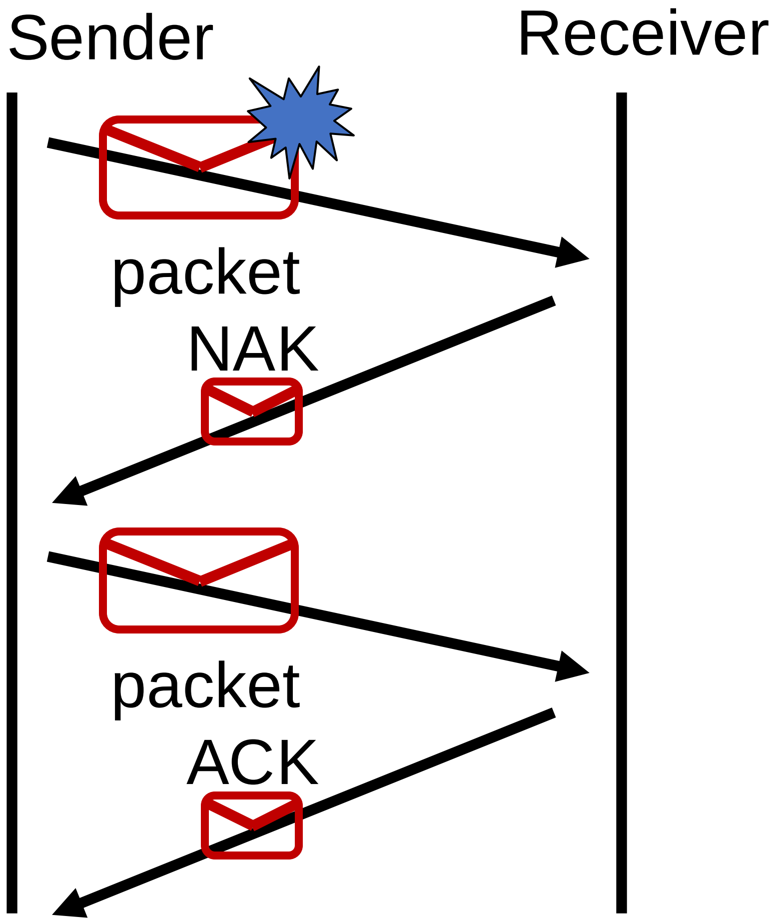
- Key idea: Receiver returns an acknowledgment (ACK) per packet sent
- If sender receives an ACK, it knows that the receiver got the packet.



Coping with packet corruption: (1) ACK

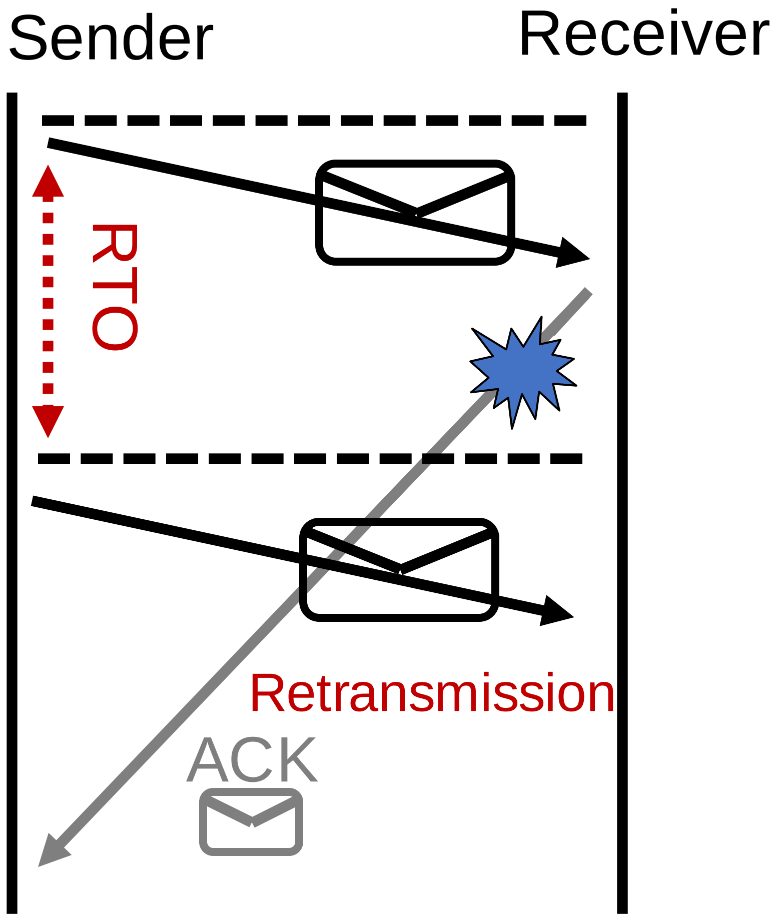
- ACKs also work to detect packet corruption on the way to the receiver
  - One possibility: A receiver could send a negative acknowledgment, or a NAK, if it receives a corrupted packet
  - Q: How to detect corrupted packet?
    - One method: Checksum!

- TCP only uses positive ACKs.



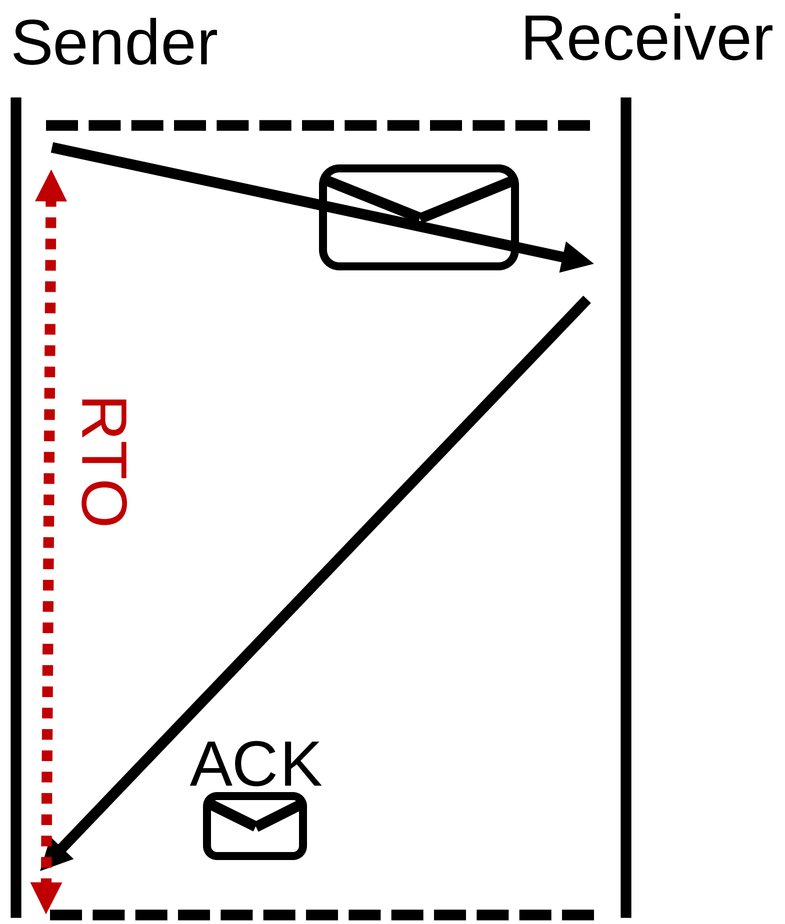
Coping with packet loss: (2) RTO

- What if a packet is dropped?
- Key idea: Wait for a duration of time (called retransmission timeout or RTO) before re-sending the packet
- In TCP, the onus is on the sender to retransmit lost data when ACKs are not received
- Note that retransmission works also if ACKs are lost or delayed



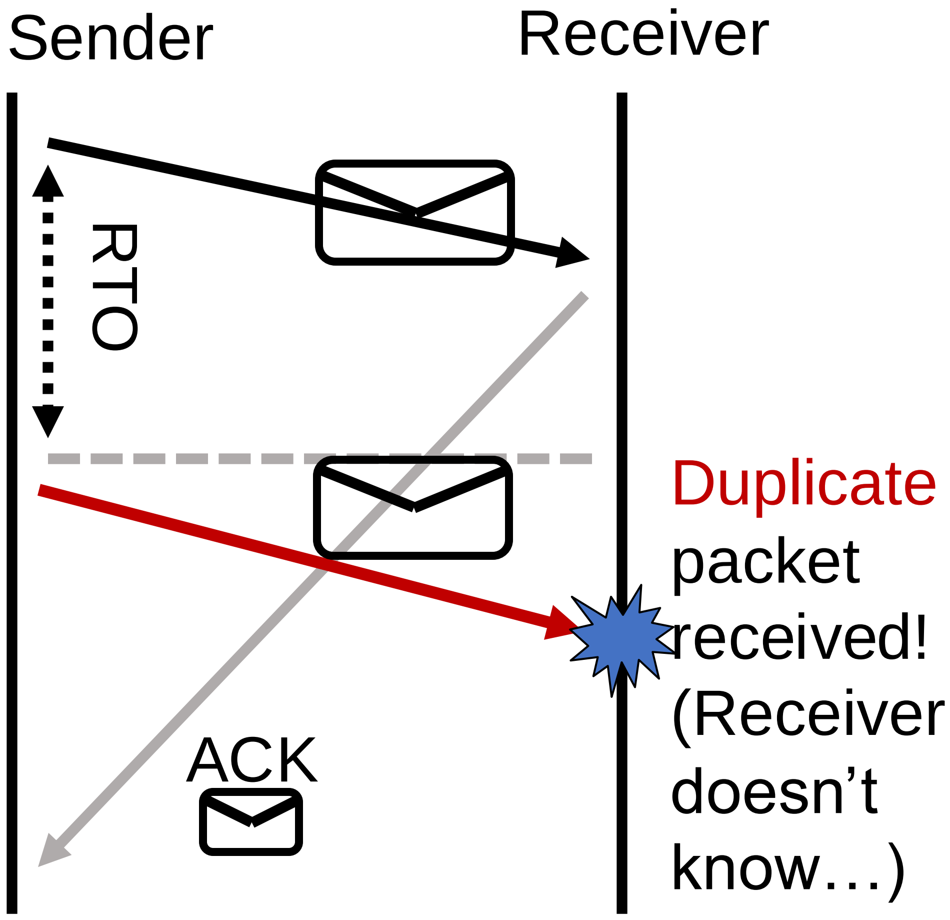
How should the RTO be set?

- A good RTO must predict the round-trip time (RTT) between the sender and receiver
  - RTT: the time to send a single packet and receive a (corresponding) single ACK at the sender
- Intuition: If an ACK hasn't returned, and our (best estimate of) RTT has elapsed, the packet was likely dropped.
- RTT can be measured directly at the sender. No receiver or router help needed.



Coping with packet duplication

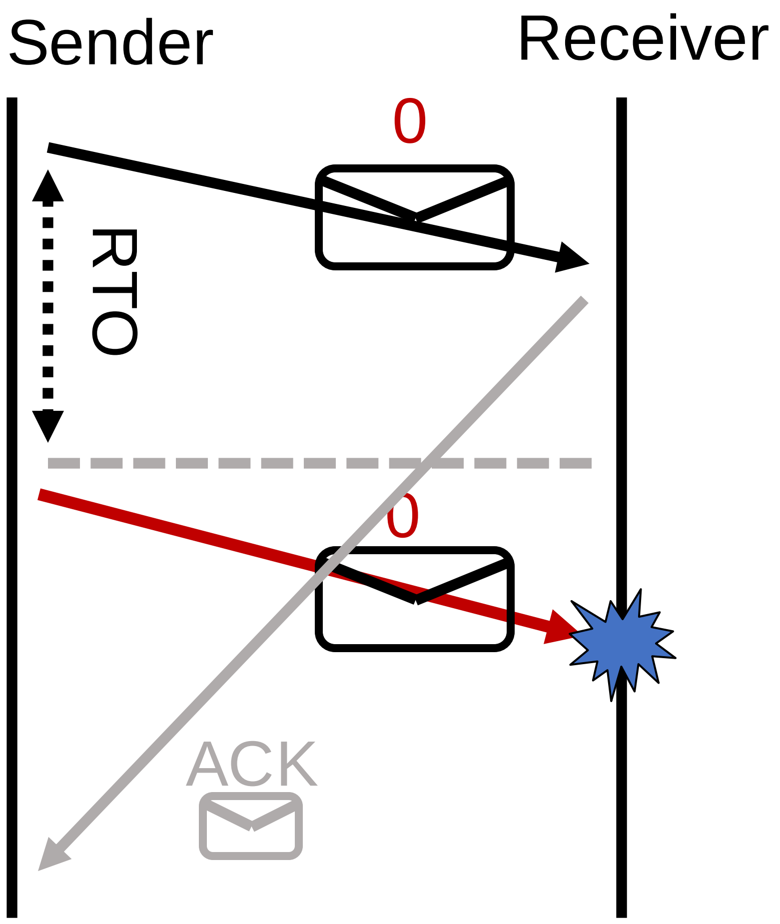
- If ACKs delayed beyond the RTO, sender may retransmit the same data
  - Receiver wouldn't know that it just received duplicate data from this retransmitted packet
- Add some identification to each packet to help distinguish between adjacent transmissions
  - This is known as the sequence number



Coping with packet loss: (3) Sequence #s

- A bad scenario: Suppose an ACK was delayed beyond the RTO; sender ended up retransmitting the packet.

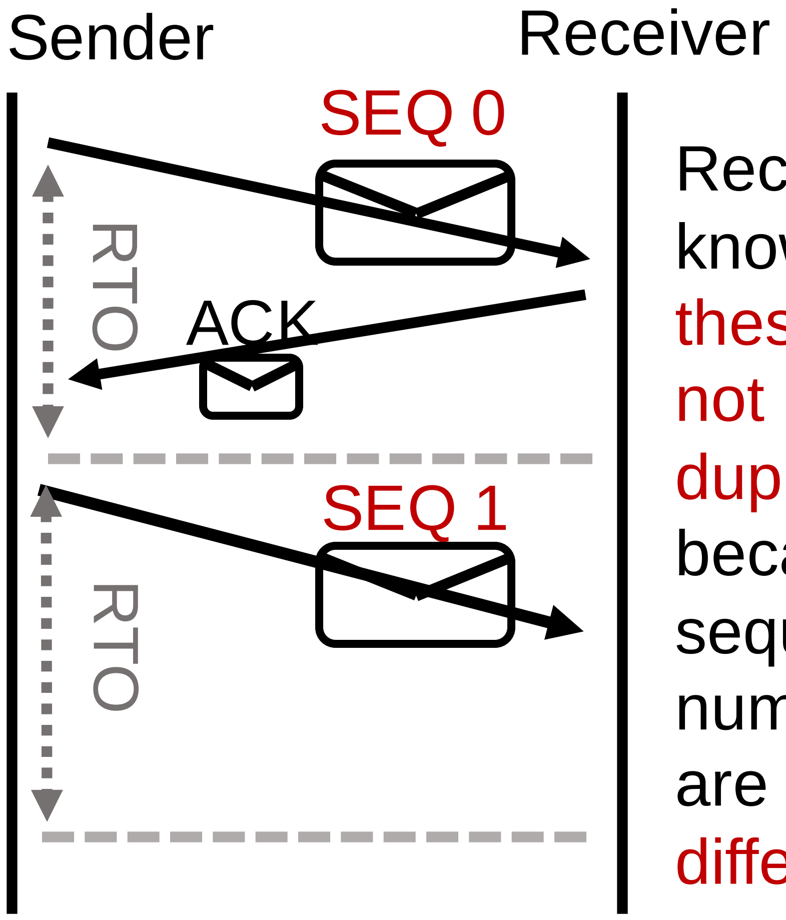
- At the receiver: sequence number helps disambiguate a fresh transmission from a retransmission
  - Sequence number same as earlier: retransmission
  - Fresh sequence number: fresh data



Coping with packet loss: (3) Sequence #s

- A good scenario: packet successfully received and ACK returned within RTO

- Sequence numbers of successively transmitted packets are different



Receiver knows these are not duplicate, because sequence numbers are different

Coping with packet loss: (3) Sequence #s

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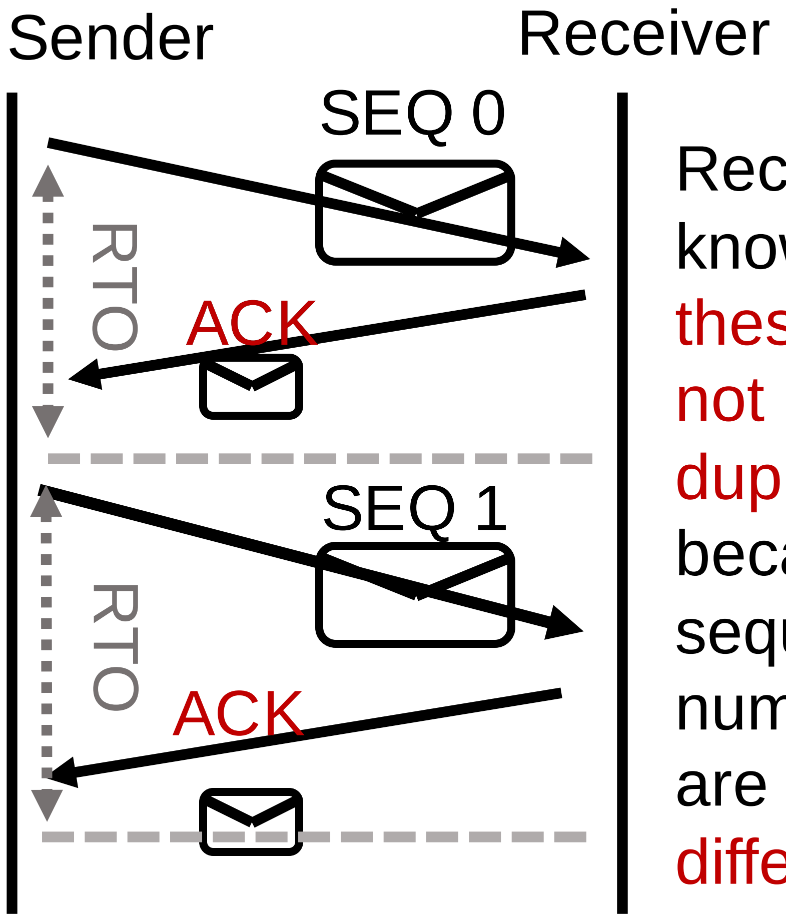
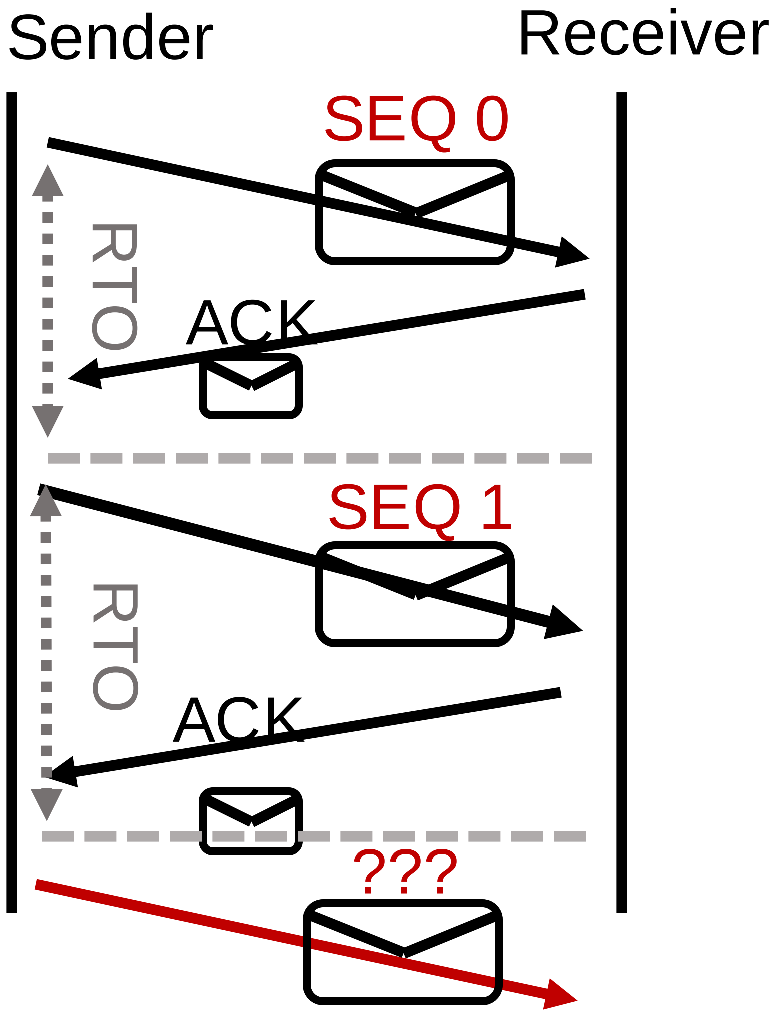


Receiver knows these are not duplicate, because sequence numbers are different

Q: What is the seq# of third packet?

- Goal: Avoid ambiguity on which packet was received/ACK'ed from both the sender and receiver's perspective
- One option: increment seq#: 2, 3, …
- Alternative: since seq # 0 was successfully ACK'ed earlier, it is OK to reuse seq #0 for next transmission.
- Seq #s reusable if older packets with those seq #s known to be delivered

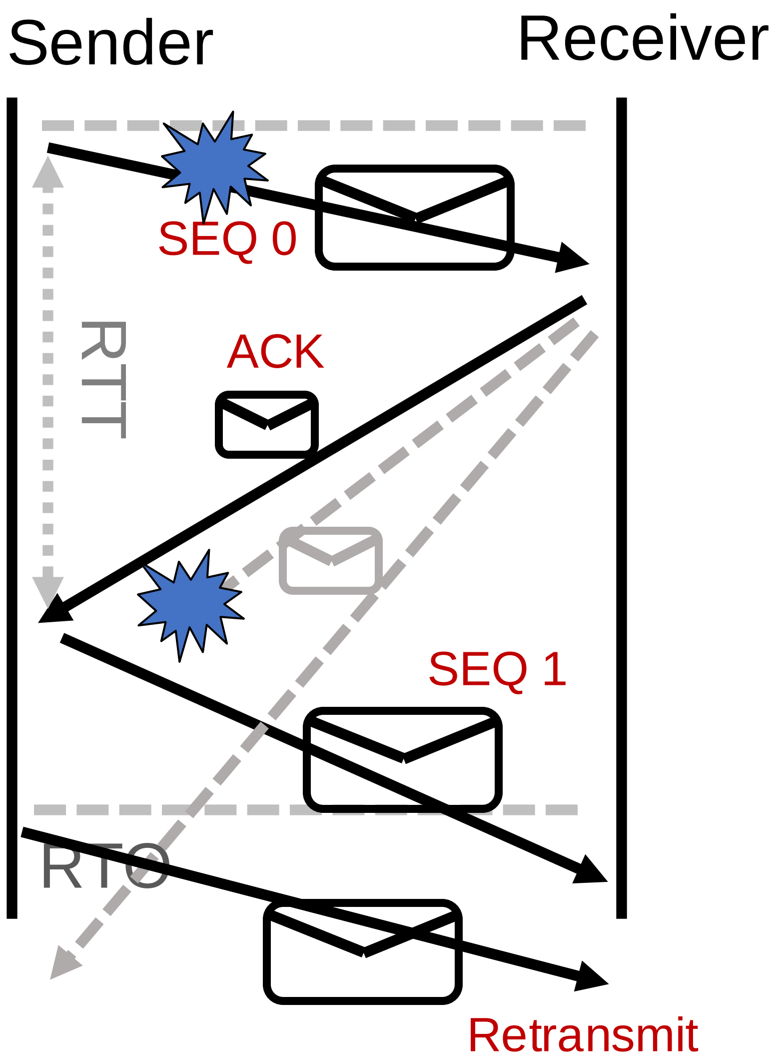


Stop-and-Wait Reliability

- Sender sends a single packet, then waits for an ACK to know the packet was successfully received. Then the sender transmits the next packet.

- If ACK is not received until a timeout (RTO), sender retransmits the packet

- Disambiguate duplicate vs. fresh packets using sequence numbers that change on “adjacent” packets



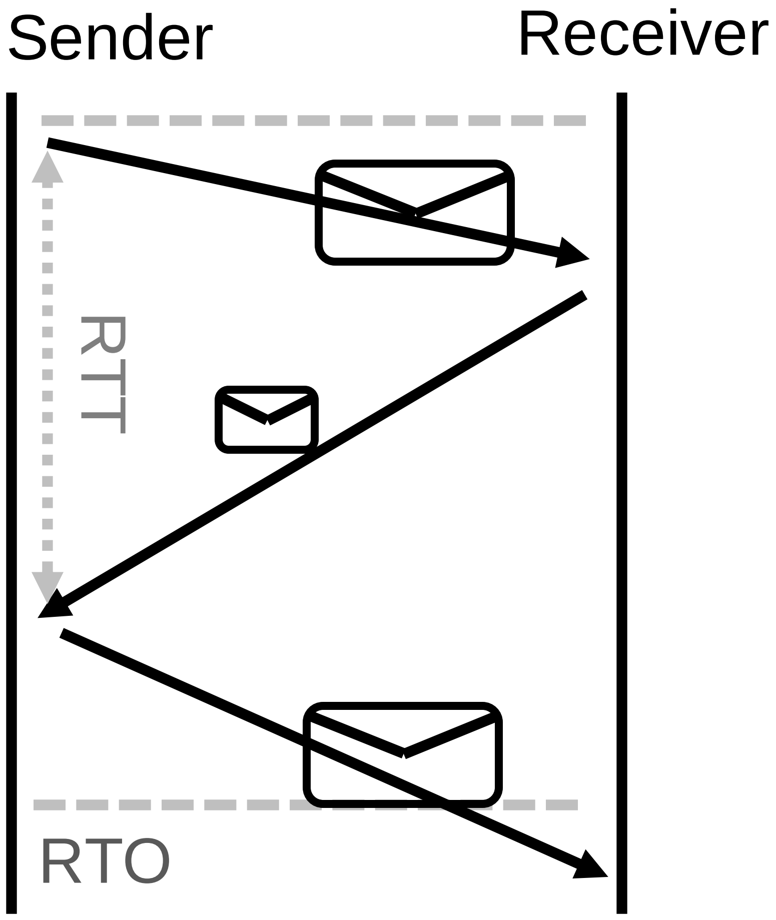
In principle, these three ideas are sufficient to implement reliable data delivery!

Efficiency problem with stop-and-wait

- Sender sends one packet, waits for an ACK (or RTO) before transmitting next one
  - Unfortunately, too slow ☹

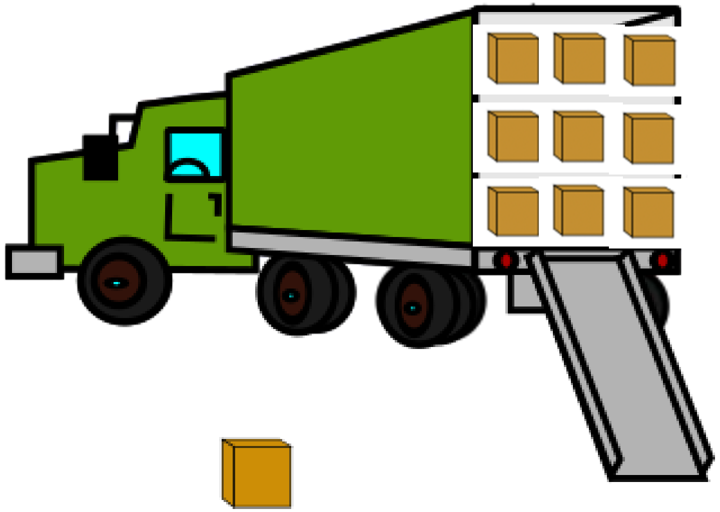
- Suppose RTO = RTT = 100 milliseconds
- Packet size (bytes in 1 packet) = 12,000 bits
- Bandwidth of links from sender to receiver = 12 Mbit/s (1 M = 10^6)

- Rate of data transfer = data size / time



120 Kilobit/s == 1% of bw!

Sending one packet per RTT makes the data transfer rate limited by the time between the endpoints, rather than the bandwidth.



Ensure you got the (one) box safely; make N trips

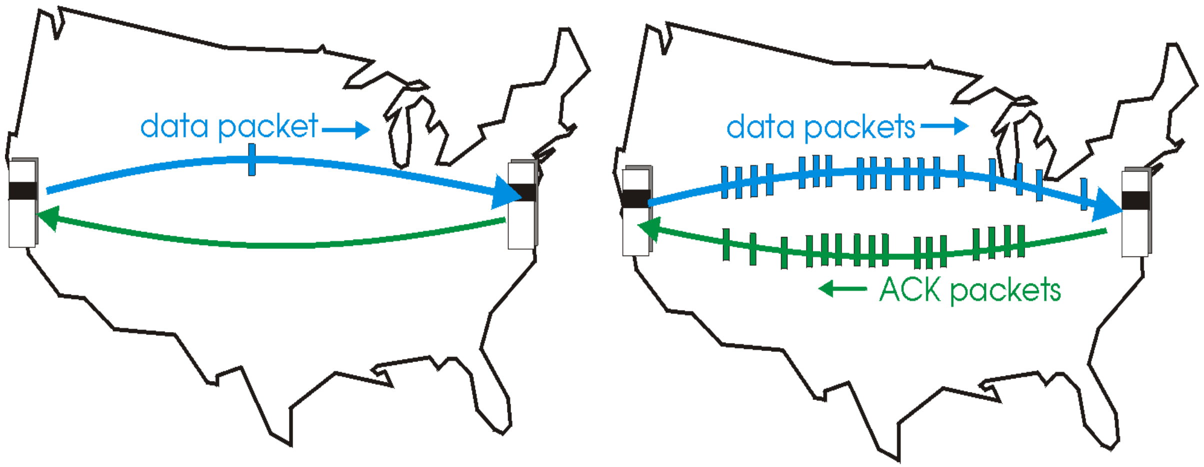
Ensure you get N boxes safely; make just 1 trip!

Keep many packets in flight

Making reliable data transfer efficient

Pipelined reliability

- Data in flight: data that has been sent, but sender hasn’t yet received ACKs from the receiver
  - Note: can refer to packets in flight or bytes in flight
- New packets sent at the same time as older ones still in flight
- New packets sent at the same time as ACKs are returning
- More data moving in same time!
- Improves throughput
  - Rate of data transfer

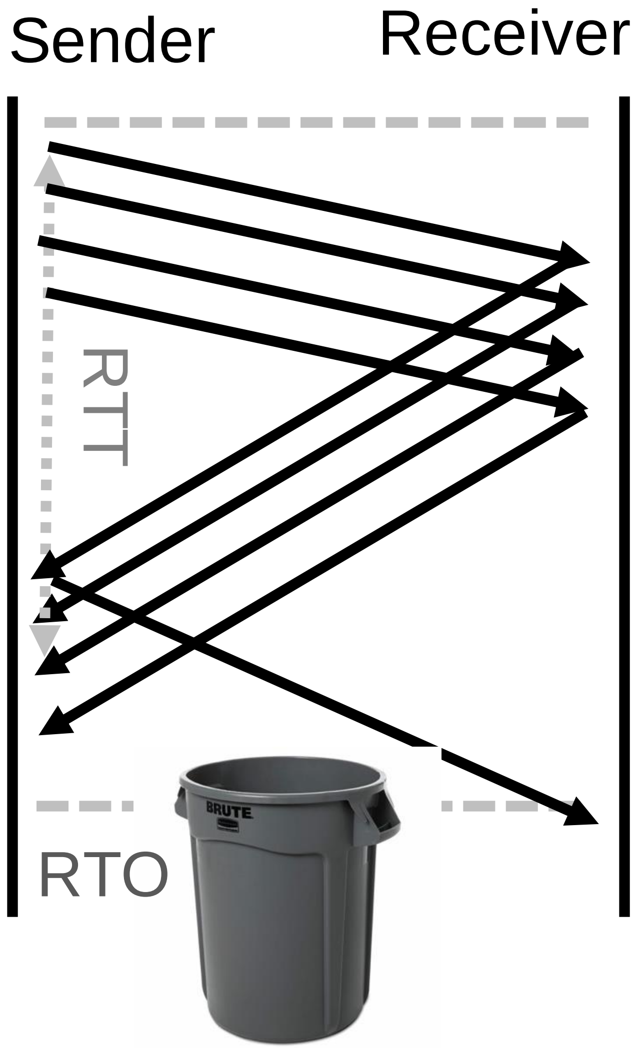


(a) a stop-and-wait protocol in operation

(b) a pipelined protocol in operation

Pipelined reliability

- Stop and wait: send 1 packet per RTT
- Pipelined: send N packets per RTT
- If there are N packets in flight, throughput improves by N times compared to stop-and-wait!



Pipelining makes reliable data transfer efficient.

However, pipelining also makes it more complex.

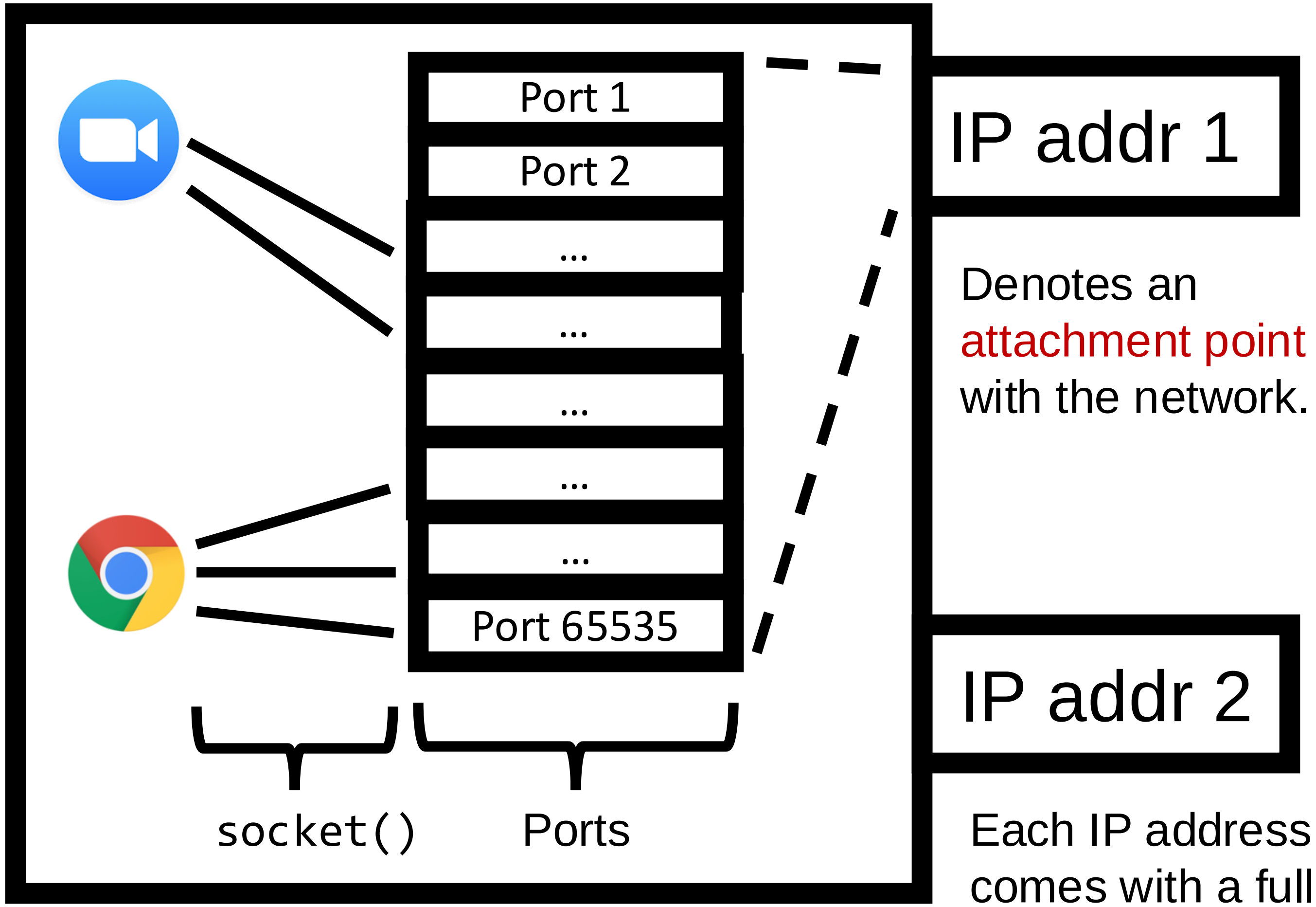
Which packets are currently in flight?

Which packets were successfully delivered?

Which packets should the sender retransmit?

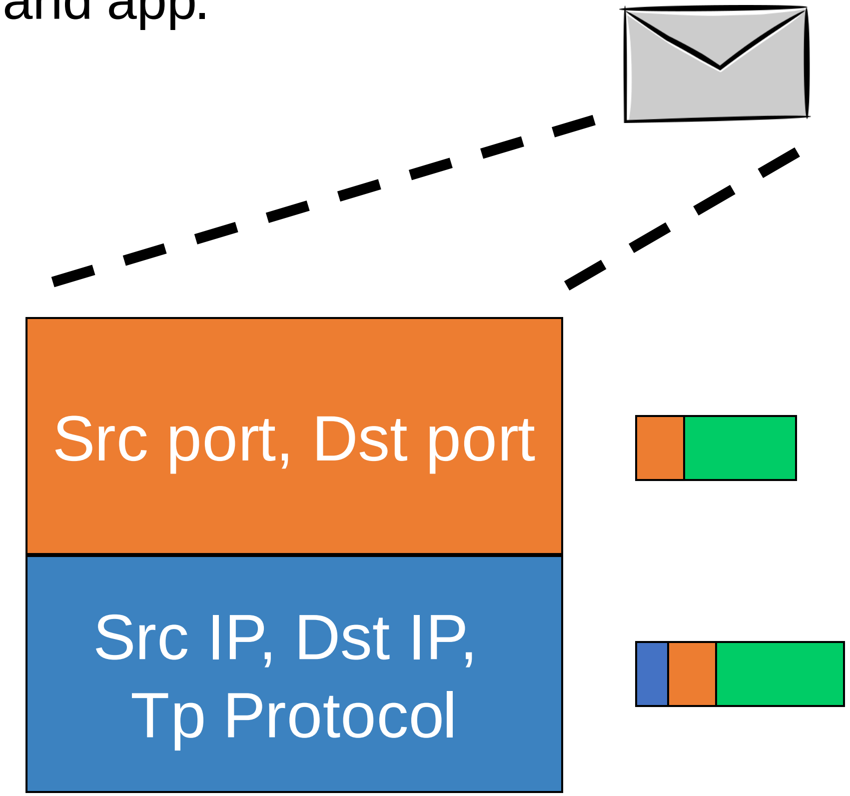
Review: Demultiplexing

Connection lookup: The operating system does a lookup using these data to determine the right socket and app.



Machine

Each IP address comes with a full copy of its own ports.



Listing sockets and connections

- `ss`
- `iperf –s` and `iperf –s -u`

User Datagram Protocol

UDP: User Datagram Protocol [RFC 768]

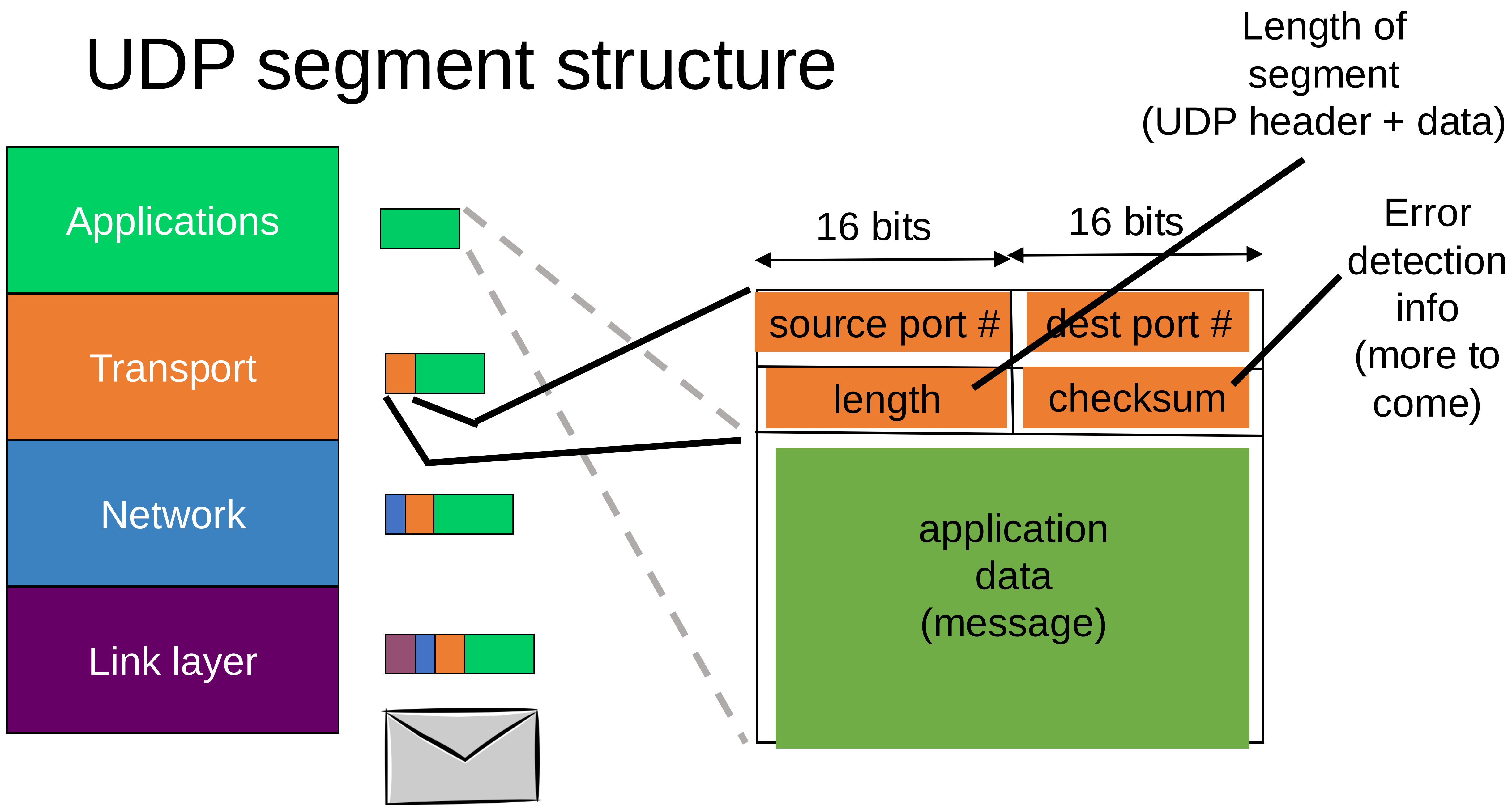
- Best effort service
  - UDP segments may be lost, corrupted, reordered
- UDP is connectionless
  - Each UDP segment handled independently of others (i.e. no “memory” across packets)
- Suitable for one-off req/resp
  - E.g., DNS uses UDP
- Early multimedia apps used UDP
  - Delay-sensitive but loss tolerant

Why are UDP’s guarantees even okay?

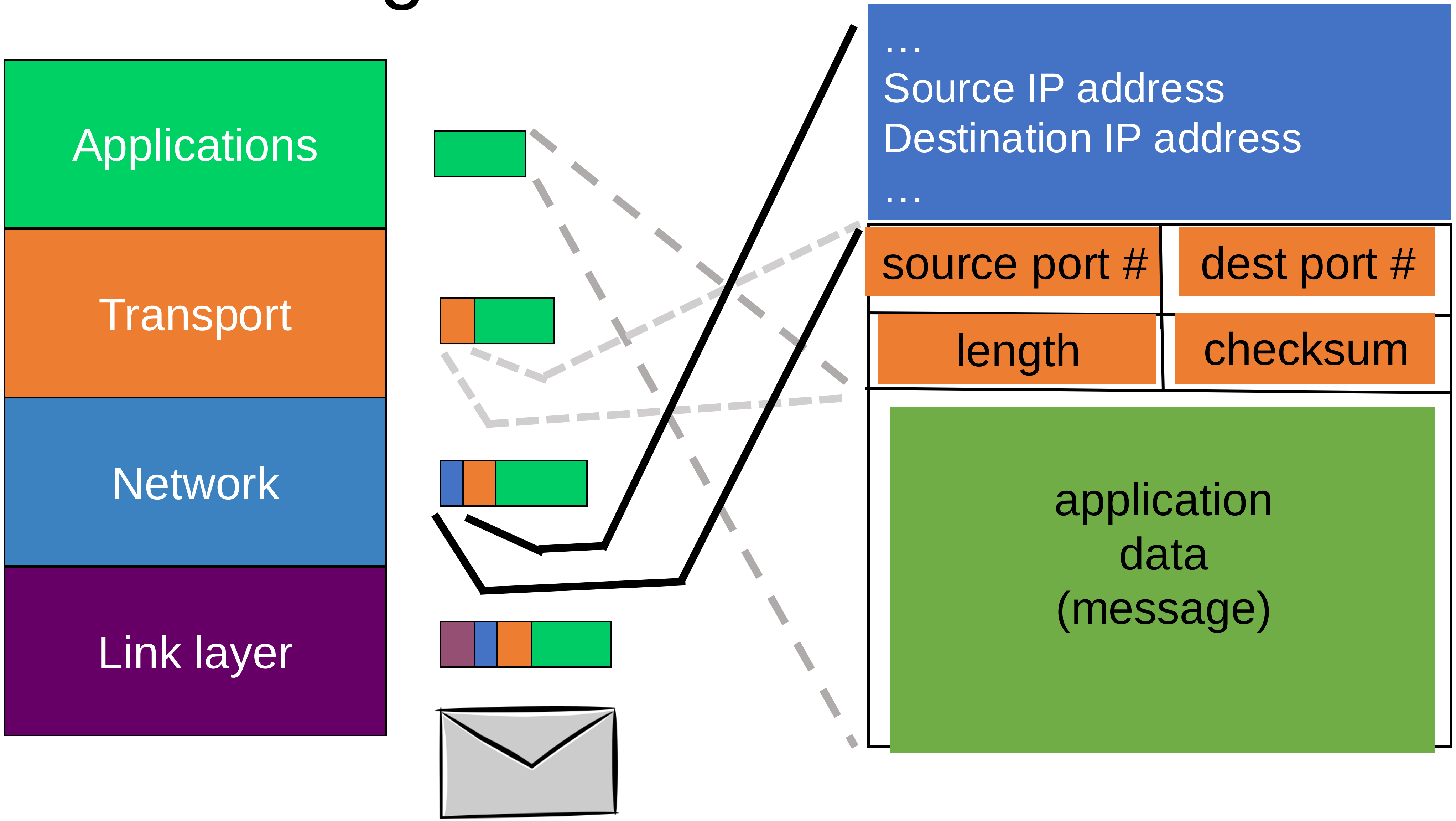
Simple & low overhead compared to TCP:

- No delays due to “connection establishment” (which TCP does)
  - UDP can send a packet immediately
- Small segment header (TCP’s is larger)
- UDP can blast data without control
  - TCP is more balanced and measured
- Less memory for connection “state” at sender & receiver relative to TCP

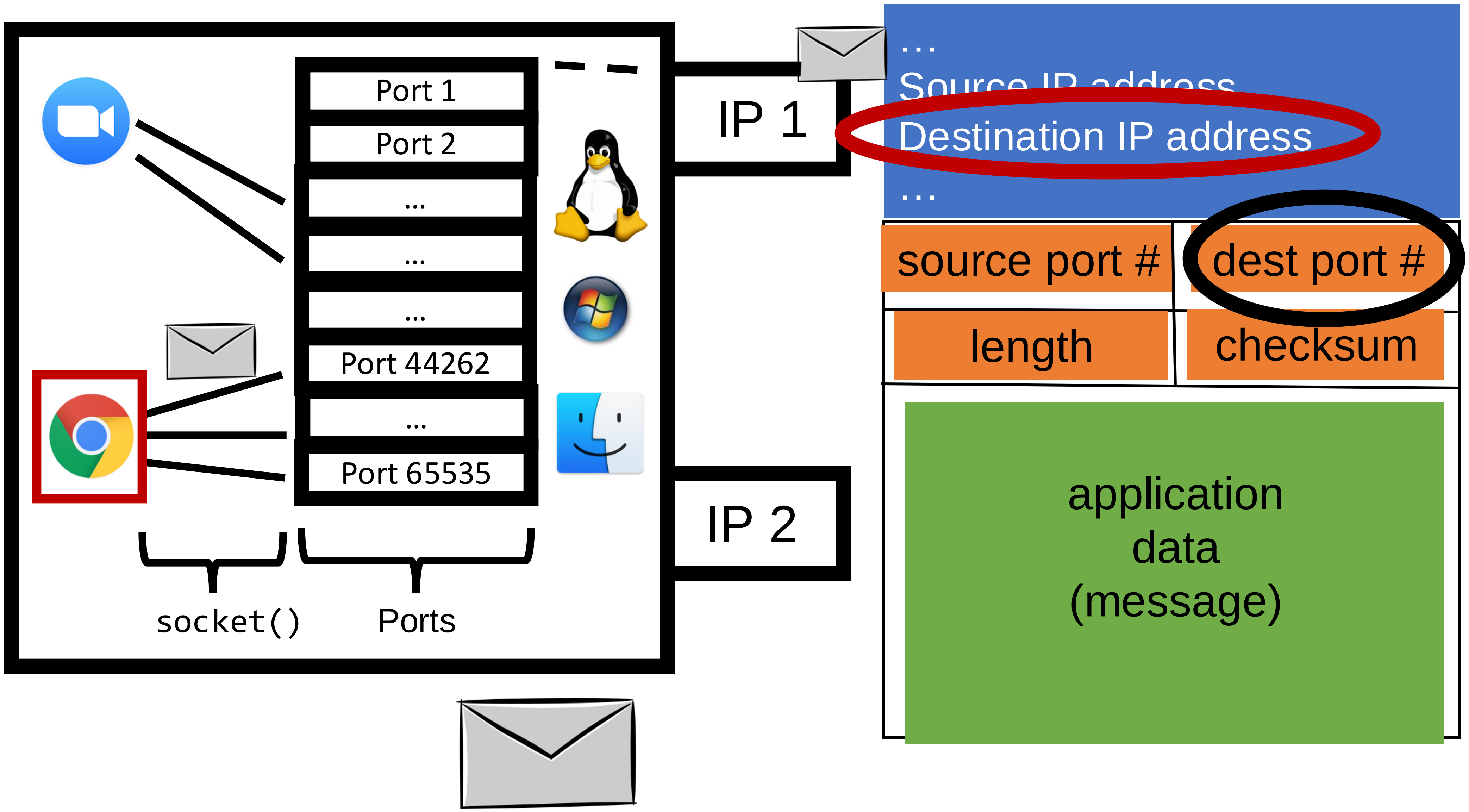
UDP segment structure



UDP segment structure



Review: UDP demultiplexing



Seeing UDP packets in action

- How to craft and send (UDP) packets?
  - It’s simpler than you think!

- `sudo tcpdump -i lo -XAvvv udp # observe packets`
- `sudo scapy # tool used to send crafted packets`
- Example:
  - `send(IP(dst="127.0.0.1")/UDP(sport=1024, dport=2048)/"hello world”, iface="lo")`
- See other fields of UDP using `UDP().fields_desc`
- Scapy can send and receive crafted packets!
  - However, it requires sudo (superuser privileges)

Error Detection in the Transport Layer

Why error detection?

- Network provides best effort service
- UDP is a simple and low overhead transport
  - Data may be corrupted along the way (e.g., 1 -> 0)
- However, simple error detection is possible!
  - Was the data I received the same data the remote machine sent?
- Error detection is a useful feature for all transport protocols including TCP

- Q: Suppose you’re sending a package to a friend. How would you detect tampering with that package?

Error Detection in UDP and TCP

- Key idea: have sender compute a function over the data
  - Store the result in the packet
  - Receiver can check the function's value in received packet

- An analogy: you're sending a package of goodies and want your recipient to know if goodies were leaked along the way

- Your idea: weigh the package; stamp the weight on the package
  - Have the recipient weigh the package and cross-check the weight with the stamped value

Requirements on error detection function

- Function must be easy to compute
- Function value must change if the packet changes
  - If the packet was modified through "likely" changes, the function value must change
- Function must be easy to verify

- UDP and TCP use a class of function called a checksum
  - Very common idea: used in multiple parts of networks and computer systems

UDP & TCP's Checksum function

Sender:

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

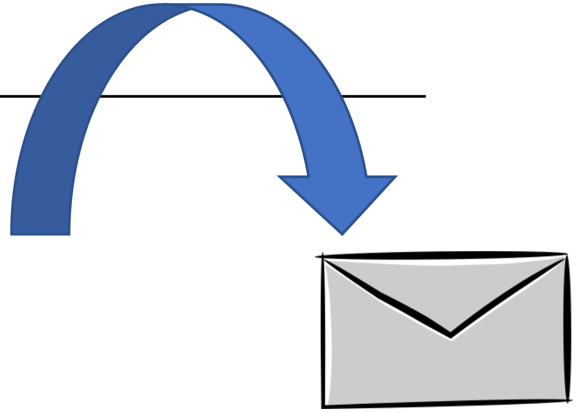
Receiver:

- compute a checksum of the received segment, including the checksum in packet itself
- check if the resulting (computed) checksum is 0
- NO – an error is detected
- YES – *assume* no error

Computing 1's complement sum

- Very similar to regular (unsigned) binary addition.
- However, when adding numbers, a carryout from the most significant bit needs to be added to the result
- Example: add two 16-bit integers

		1	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0
		1	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1
wraparound	1	1	0	1	1	1	0	1	1	1	0	1	1	1	0	1	1
sum		1	0	1	1	1	0	1	1	1	0	1	1	1	1	0	0
checksum		0	1	0	0	0	1	0	0	0	1	0	0	0	0	1	1



From the UDP specification (RFC 768)

- Checksum is the 16-bit one's complement of the one's complement sum of a pseudo header of information from the IP header, the UDP header, and the data, padded with zero octets at the end (if necessary) to make a multiple of two octets.

- The pseudo header conceptually prefixed to the UDP header contains the source address, the destination address, the protocol, and the UDP length.

Warning: Technical language ahead