Computer Networking

Lent Term M/W/F 11-midday
LT1 in Gates Building

Slide Set 4

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Topic 5a – Transport

Our goals:

- understand principles behind transport layer services:
  - multiplexing/ demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- learn about transport layer protocols in the Internet:
  - UDP: connectionless transport
  - TCP: connection-oriented transport
  - TCP congestion control

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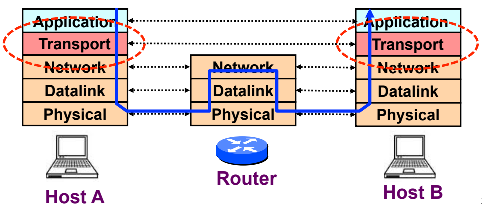
Transport Layer

- Commonly a layer at end-hosts, between the application and network layer



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Why a transport layer?

- IP packets are addressed to a host but end-to-end communication is between application processes at hosts
  - Need a way to decide which packets go to which applications (*more multiplexing*)

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Why a transport layer?

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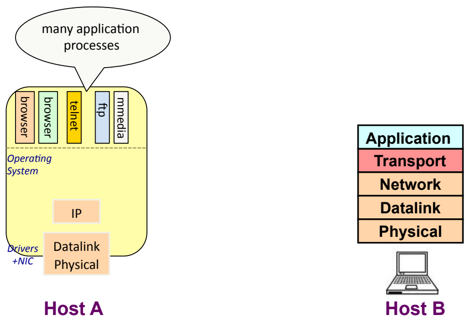

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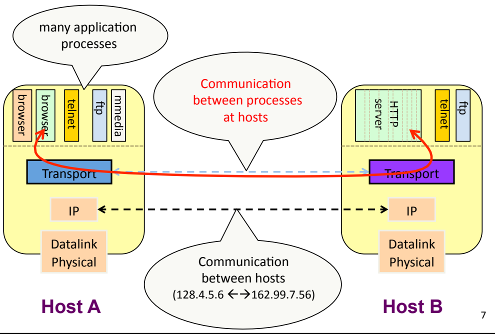
Why a transport layer?



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Why a transport layer?

- IP packets are addressed to a host but end-to-end communication is between application processes at hosts
  - Need a way to decide which packets go to which applications (mux/demux)
- IP provides a weak service model (*best-effort*)
  - Packets can be corrupted, delayed, dropped, reordered, duplicated
  - No guidance on how much traffic to send and when
  - Dealing with this is tedious for application developers

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Role of the Transport Layer

- Communication between application processes
  - Multiplexing between application processes
  - Implemented using *ports*

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Role of the Transport Layer

- Communication between application processes
- Provide common end-to-end services for app layer [optional]
  - Reliable, in-order data delivery
  - Paced data delivery: flow and congestion-control
    - too fast may overwhelm the network
    - too slow is not efficient

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Role of the Transport Layer

- Communication between processes
- Provide common end-to-end services for app layer [optional]
- TCP and UDP are the common transport protocols
  - also SCTP, MTCP, SST, RDP, DCCP, …

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Role of the Transport Layer

- Communication between processes
- Provide common end-to-end services for app layer [optional]
- TCP and UDP are the common transport protocols
- UDP is a minimalist, no-frills transport protocol
  - only provides mux/demux capabilities

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Role of the Transport Layer

- Communication between processes
- Provide common end-to-end services for app layer [optional]
- TCP and UDP are the common transport protocols
- UDP is a minimalist, no-frills transport protocol
- TCP is the *totus porcus* protocol
  - offers apps a reliable, in-order, byte-stream abstraction
  - with congestion control
  - but **no** performance (delay, bandwidth, ...) guarantees

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Role of the Transport Layer

- Communication between processes
  - mux/demux from and to application processes
  - implemented using ports

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Context: Applications and Sockets

- Socket: software abstraction by which an application process exchanges network messages with the (transport layer in the) operating system
  - socketID = socket(..., socket.TYPE)
  - socketID.sendto(message, ...)
  - socketID.recvfrom(...)
- Two important types of sockets
  - UDP socket: TYPE is SOCK_DGRAM
  - TCP socket: TYPE is SOCK_STREAM

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Ports

- Problem: deciding which app (socket) gets which packets
- Solution: ***port*** as a transport layer identifier
  - 16 bit identifier
  - OS stores mapping between sockets and *ports*
  - a packet carries a source and destination port number in its transport layer header
- For UDP ports (SOCK_DGRAM)
  - OS stores (local port, local IP address) ←→ socket
- For TCP ports (SOCK_STREAM)
  - OS stores (local port, local IP, remote port, remote IP) ←→ socket

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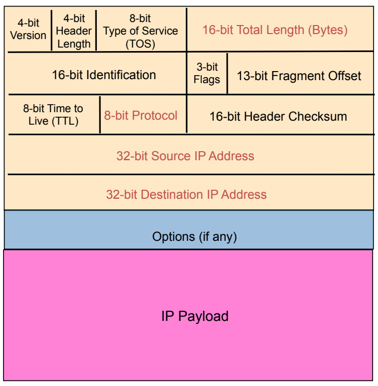

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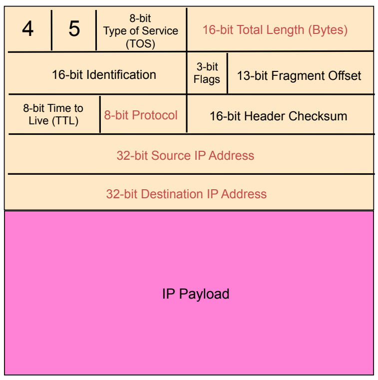

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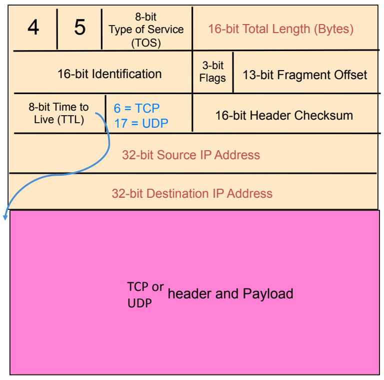

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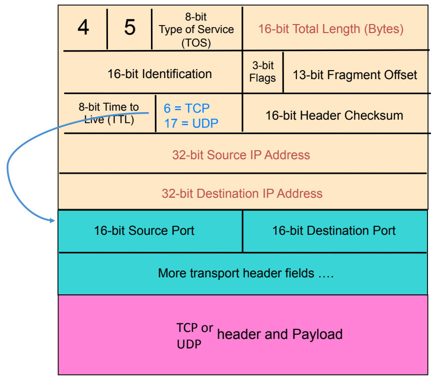

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Recap: Multiplexing and Demultiplexing

- Host receives IP packets
  - Each IP header has source and destination IP address
  - Each Transport Layer header has source and destination port number
- Host uses IP addresses and port numbers to direct the message to appropriate socket

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More on Ports

- Separate 16-bit port address space for UDP and TCP
- “Well known” ports (0-1023): everyone agrees which services run on these ports
  - e.g., ssh:22, http:80
  - helps client know server’s port
- Ephemeral ports (most 1024-65535): dynamically selected: as the source port for a client process

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UDP: User Datagram Protocol

- Lightweight communication between processes
  - Avoid overhead and delays of ordered, reliable delivery
- UDP described in RFC 768 – (1980!)
  - Destination IP address and port to support demultiplexing
  - Optional error checking on the packet contents
    - (checksum field of 0 means “don’t verify checksum”)

SRC port	DST port
checksum	length
DATA	

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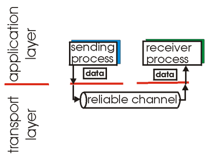
Principles of Reliable data transfer

- important in app., transport, link layers
- top-10 list of important networking topics!



(a) provided service

- In a perfect world, reliable transport is easy

But the Internet default is *best-effort*

- All the bad things best-effort can do
  - a packet is corrupted (bit errors)
  - a packet is lost
  - a packet is delayed (*why?*)
  - packets are reordered (*why?*)
  - a packet is duplicated (*why?*)

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Principles of Reliable data transfer

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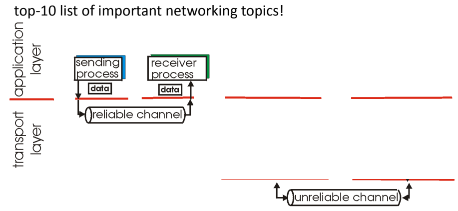

(a) provided service (b) service implementation

- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

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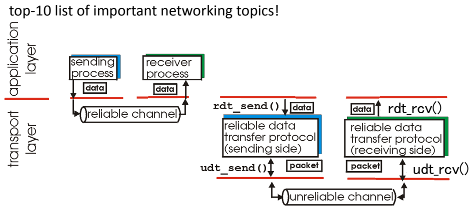
Principles of Reliable data transfer

- important in app., transport, link layers
- top-10 list of important networking topics!



(a) provided service (b) service implementation

- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

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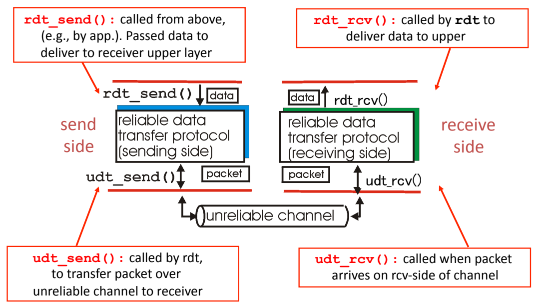
Reliable data transfer: getting started



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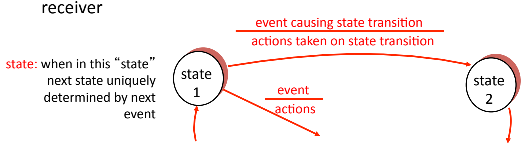
Reliable data transfer: getting started

We'll:

- incrementally develop sender, receiver sides of reliable data transfer protocol (rdt)
- consider only unidirectional data transfer
  - but control info will flow on both directions!
- use finite state machines (FSM) to specify sender, receiver



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KR state machines – a note.

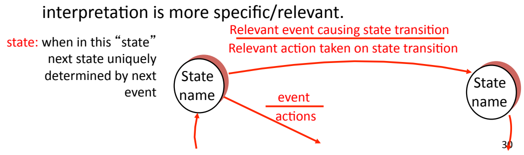
Beware

Kurose and Ross has a confusing/confused attitude to state-machines.

I've attempted to normalise the representation.

UPSHOT: these slides have differing information to the KR book (from which the RDT example is taken.)

in KR "actions taken" appear wide-ranging, my interpretation is more specific/relevant.



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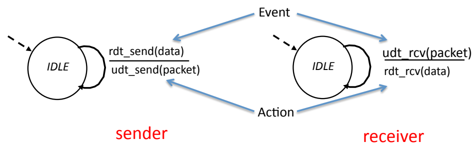
Rdt1.0: reliable transfer over a reliable channel

- underlying channel perfectly reliable
  - no bit errors
  - no loss of packets
- separate FSMs for sender, receiver:
  - sender sends data into underlying channel
  - receiver read data from underlying channel



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Rdt2.0: channel with bit errors

- underlying channel may flip bits in packet
  - checksum to detect bit errors
- *the* question: how to recover from errors:
  - *acknowledgements (ACKs):* receiver explicitly tells sender that packet received is OK
  - *negative acknowledgements (NAKs):* receiver explicitly tells sender that packet had errors
  - sender retransmits packet on receipt of NAK
- new mechanisms in `rdt2.0` (beyond `rdt1.0`):
  - error detection
  - receiver feedback: control msgs (ACK,NAK) receiver->sender

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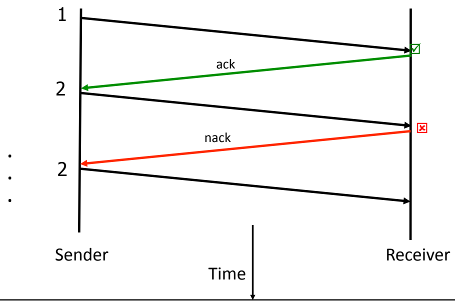
Dealing with Packet Corruption



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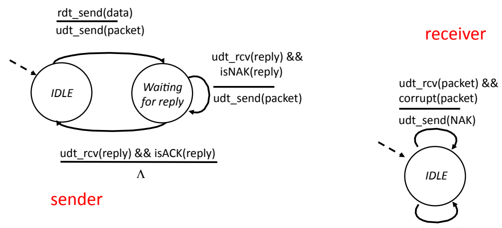
rdt2.0: FSM specification



Note: the sender holds a copy of the packet being sent until the delivery is acknowledged.

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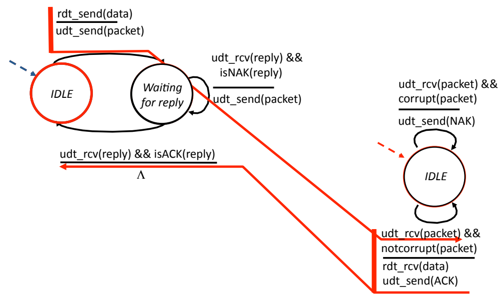
rdt2.0: operation with no errors



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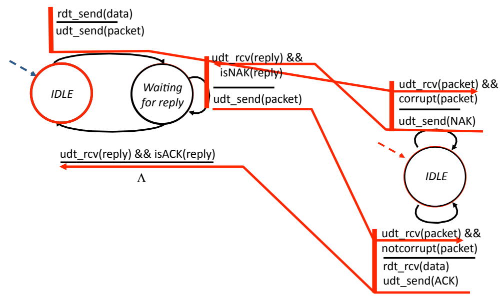
rdt2.0: error scenario



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rdt2.0 has a fatal flaw!

What happens if ACK/NAK corrupted?
- sender doesn't know what happened at receiver!
- can't just retransmit: possible duplicate

Handling duplicates:
- sender retransmits current packet if ACK/NAK garbled
- sender adds *sequence number* to each packet
- receiver discards (doesn't deliver) duplicate packet

> **stop and wait**
> Sender sends one packet, then waits for receiver response

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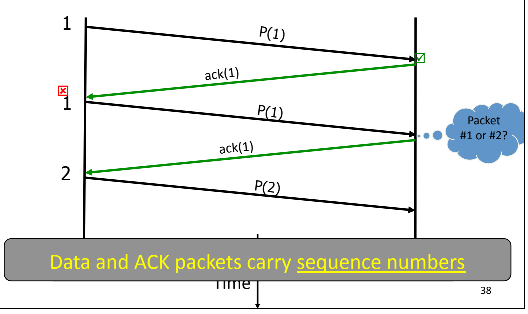
Dealing with Packet Corruption



Data and ACK packets carry sequence numbers

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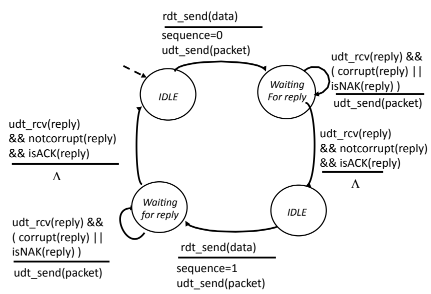
rdt2.1: sender, handles garbled ACK/NAKs



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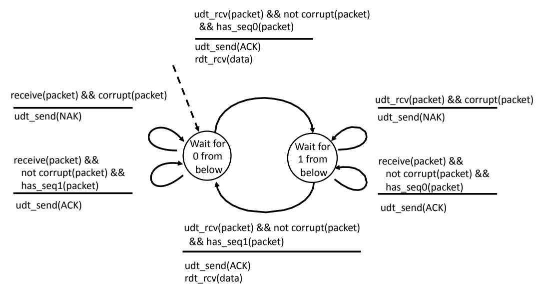
rdt2.1: receiver, handles garbled ACK/NAKs



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rdt2.1: discussion

Sender:
- seq # added to pkt
- two seq. #'s (0,1) will suffice. Why?
- must check if received ACK/NAK corrupted
- twice as many states
  - state must "remember" whether "current" pkt has a 0 or 1 sequence number

Receiver:
- must check if received packet is duplicate
  - state indicates whether 0 or 1 is expected pkt seq #
- note: receiver can *not* know if its last ACK/NAK received OK at sender

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rdt2.2: a NAK-free protocol

- same functionality as rdt2.1, using ACKs only
- instead of NAK, receiver sends ACK for last pkt received OK
  - receiver must *explicitly* include seq # of pkt being ACKed
- duplicate ACK at sender results in same action as NAK: *retransmit current pkt*

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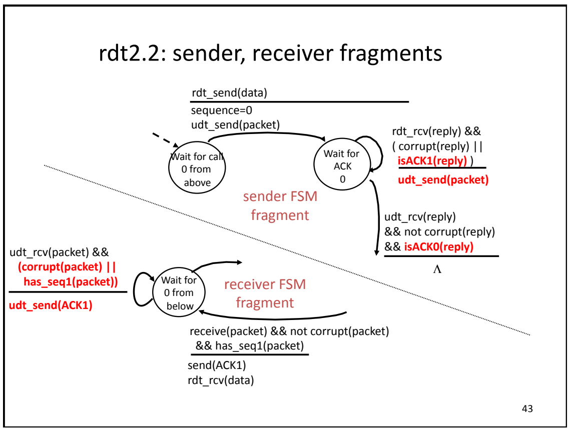

rdt3.0: channels with errors *and* loss

New assumption: underlying channel can also lose packets (data or ACKs)

- checksum, seq. #, ACKs, retransmissions will be of help, but not enough

Approach: sender waits "reasonable" amount of time for ACK

- retransmits if no ACK received in this time
- if pkt (or ACK) just delayed (not lost):
  - retransmission will be duplicate, but use of seq. #'s already handles this
  - receiver must specify seq # of pkt being ACKed
- requires countdown timer

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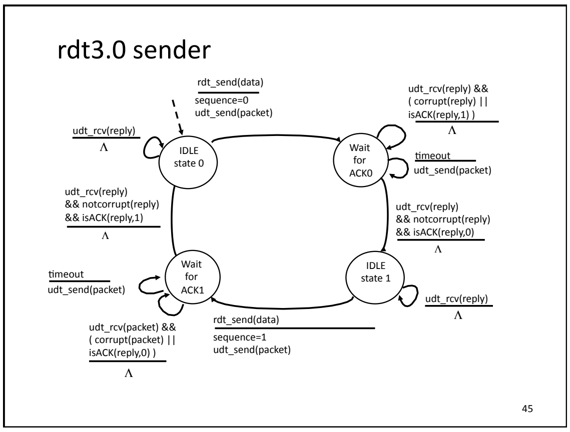

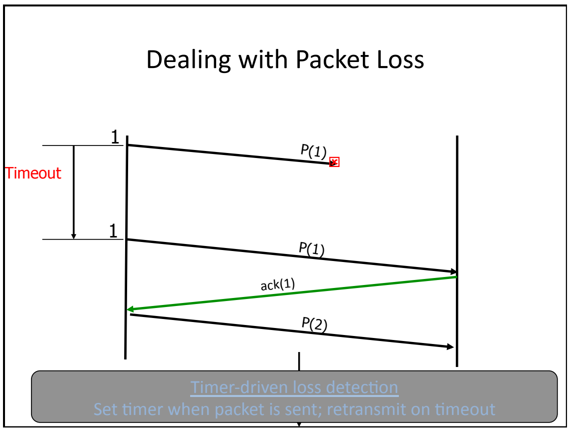

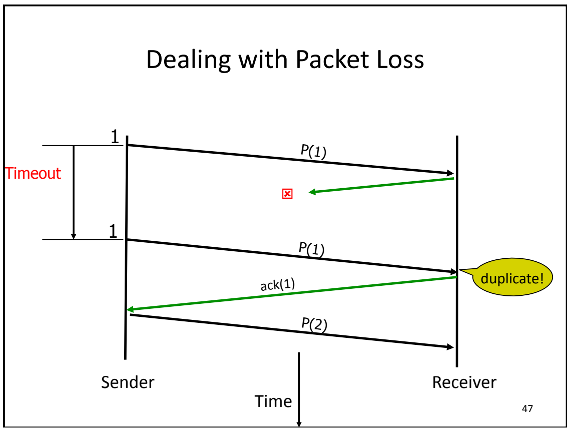

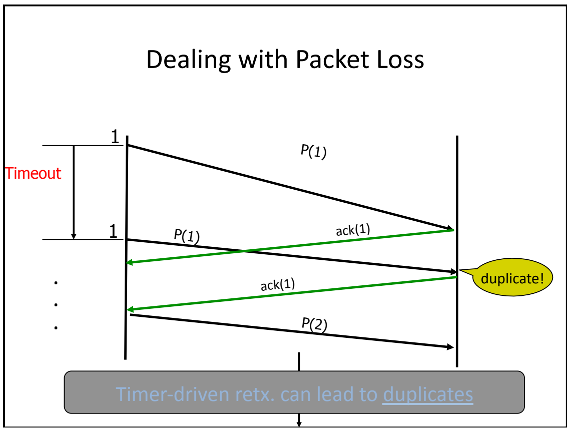

Performance of rdt3.0

- rdt3.0 works, but performance stinks
- ex: 1 Gbps link, 15 ms prop. delay, 8000 bit packet:

$$d_{trans} = \frac{L}{R} = \frac{8000\text{bits}}{10^9\text{bps}} = 8\text{microseconds}$$

- U_{sender}: utilization – fraction of time sender busy sending

$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

- 1KB pkt every 30 msec -> 33kB/sec throughput over 1 Gbps link
- network protocol limits use of physical resources!

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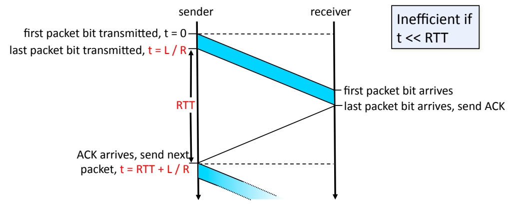
rdt3.0: stop-and-wait operation



$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

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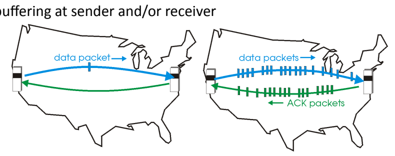
Pipelined (Packet-Window) protocols

Pipelining: sender allows multiple, “in-flight”, yet-to-be-acknowledged pkts
- range of sequence numbers must be increased
- buffering at sender and/or receiver



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

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A Sliding Packet Window

- window = set of adjacent sequence numbers
  - The size of the set is the window size; assume window size is *n*
- General idea: send up to *n* packets at a time
  - Sender can send packets in its window
  - Receiver can accept packets in its window
  - Window of acceptable packets “slides” on successful reception/ acknowledgement

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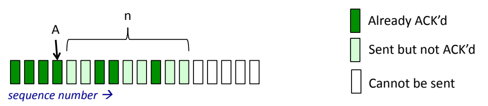
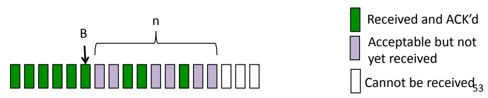
A Sliding Packet Window

- Let A be the last ack’d packet of sender without gap; then window of sender = {A+1, A+2, …, A+n}



- Let B be the last received packet without gap by receiver, then window of receiver = {B+1,…, B+n}



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Acknowledgements w/ Sliding Window

- Two common options
  - cumulative ACKs: ACK carries next in-order sequence number that the receiver expects

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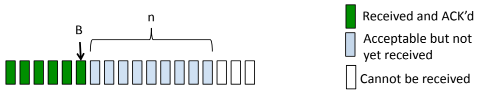
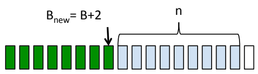
Cumulative Acknowledgements (1)

- At receiver



- After receiving B+1, B+2



- Receiver sends ACK(B_{new}+1)

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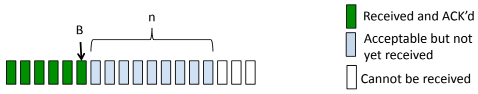
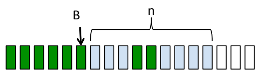
Cumulative Acknowledgements (2)

- At receiver



- After receiving B+4, B+5



How do we recover?

- Receiver sends ACK(B+1)

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Go-Back-N (GBN)

- Sender transmits up to *n* unacknowledged packets
- Receiver only accepts packets in order
  - discards out-of-order packets (i.e., packets other than *B+1*)
- Receiver uses cumulative acknowledgements
  - i.e., sequence# in ACK = next expected in-order sequence#
- Sender sets timer for 1st outstanding ack (A+1)
- If timeout, retransmit *A+1, … , A+n*

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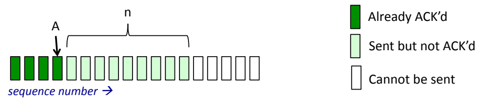
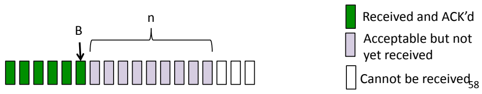
Sliding Window with GBN

- Let A be the last ack'd packet of sender without gap; then window of sender = {A+1, A+2, …, A+n}



- Let B be the last received packet without gap by receiver, then window of receiver = {B+1,…, B+n}



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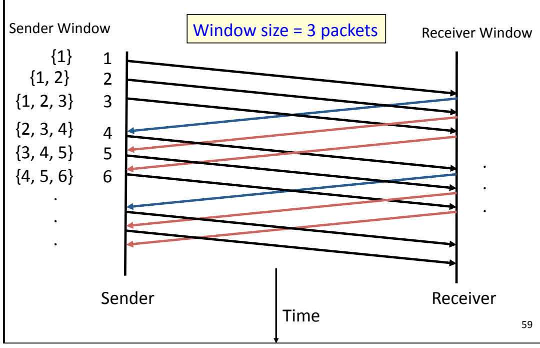
GBN Example w/o Errors



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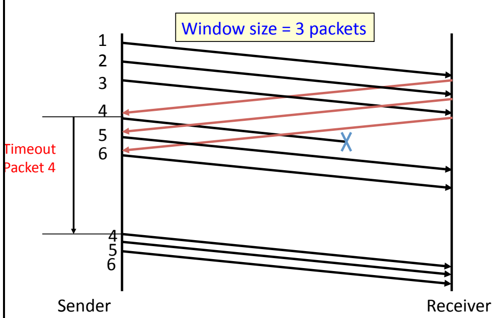
GBN Example with Errors



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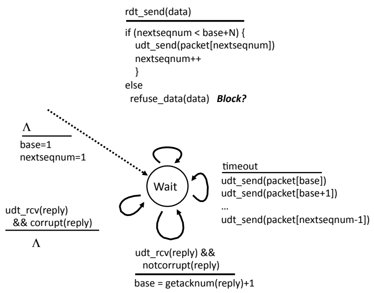
GBN: sender extended FSM



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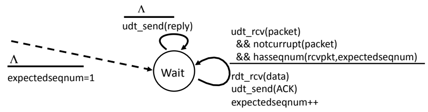
GBN: receiver extended FSM



ACK-only: always send an ACK for correctly-received packet with the highest *in-order* seq #
- may generate duplicate ACKs
- need only remember **`expectedseqnum`**

- out-of-order packet:
  - discard (don't buffer) -> no receiver buffering!
  - Re-ACK packet with highest in-order seq #

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Acknowledgements w/ Sliding Window

- Two common options
  - cumulative ACKs: ACK carries next in-order sequence number the receiver expects
  - selective ACKs: ACK individually acknowledges correctly received packets
- Selective ACKs offer more precise information but require more complicated book-keeping
- Many variants that differ in implementation details

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Selective Repeat (SR)

- Sender: transmit up to *n* unacknowledged packets
- Assume packet *k* is lost, *k+1* is not
- Receiver: indicates packet *k+1* correctly received
- Sender: retransmit only packet *k* on timeout
- Efficient in retransmissions but complex book-keeping
  - need a timer per packet

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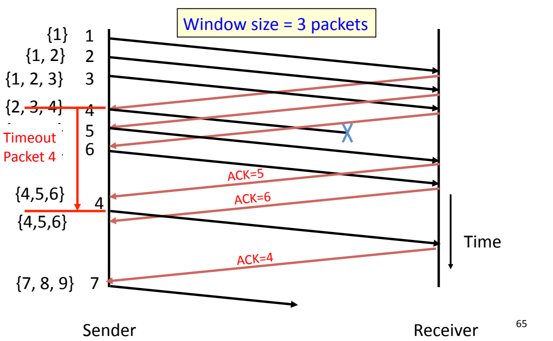
SR Example with Errors



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Observations

- With sliding windows, it is possible to fully utilize a link, provided the window size is large enough. Throughput is ~ (n/RTT)
  - Stop & Wait is like n = 1.
- Sender has to buffer all unacknowledged packets, because they may require retransmission
- Receiver may be able to accept out-of-order packets, but only up to its buffer limits
- Implementation complexity depends on protocol details (GBN vs. SR)

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Recap: components of a solution

- Checksums (for error detection)
- Timers (for loss detection)
- Acknowledgments
  - cumulative
  - selective
- Sequence numbers (duplicates, windows)
- Sliding Windows (for efficiency)

- Reliability protocols use the above to decide when and what to retransmit or acknowledge

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What does TCP do?

Most of our previous tricks + a few differences

- Sequence numbers are byte offsets
- Sender and receiver maintain a sliding window
- Receiver sends cumulative acknowledgements (like GBN)
- Sender maintains a single retx. timer
- Receivers do not drop out-of-sequence packets (like SR)
- Introduces fast retransmit : optimization that uses duplicate ACKs to trigger early retx (next time)
- Introduces timeout estimation algorithms (next time)

More in Topic 5b