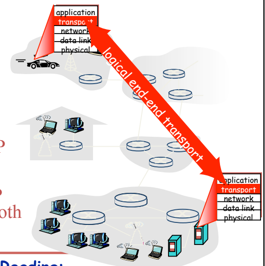
Transport Layer

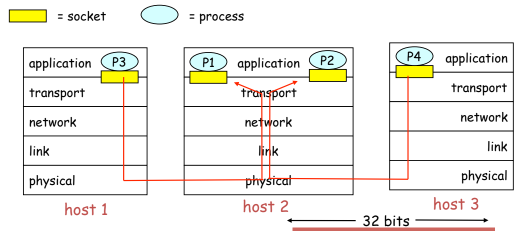
- The Transport Layer is responsible for providing logical communication between processes. Uses the services of the Network Layer to (try) to transfer data between hosts.
- The TL relies on the services of the Network layer protocol, so it is limited in the services it can provide.
- TCP/IP provides
  - congestion
  - flow control
  - connection setup
  - reliable, in-order delivery of data
  - best effort delivery
  - multiplexing of data from several processes
- TCP/IP doesn't provide (although some other protocol suites do, and TCP/IP may in future)
  - bandwidth or latency guarantees

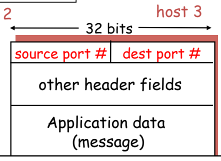
TCP
UDP
both



Reading: Kurose & Ross: Ch 3

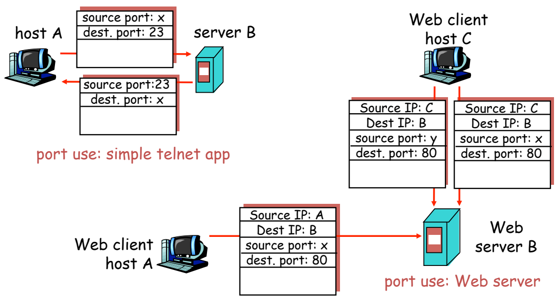
Multiplexing/Demultiplexing





TCP/UDP segment format

Multiplexing/demultiplexing: examples



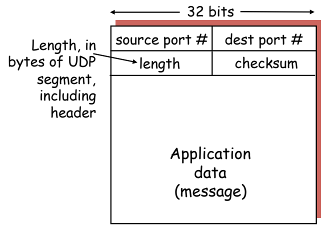
port use: simple telnet app

port use: Web server

User Datagram Protocol (UDP)

RFC 768

- no connection establishment (which can add delay)
- simple: no connection state at sender, receiver
- small segment header
- no congestion control: UDP can blast away as fast as desired
- can add reliability at application layer
- 1's complement checksum can be used to detect (but not correct) errors. (example)
- segments can be lost or delivered to application out of order.
- each segment is independent of others.

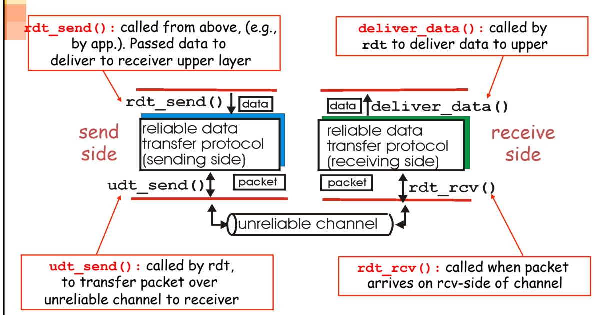


UDP segment format

Reliable Data Transfer

- Many applications want reliable data transfer, so many transport layer protocols provide this.
- The service level of the underlying network may vary. Assume the TL needs to deal with errors and loss of data packets.
- Start with the assumption of a reliable network and progressively add in mechanisms for dealing with errors… (on blackboard)

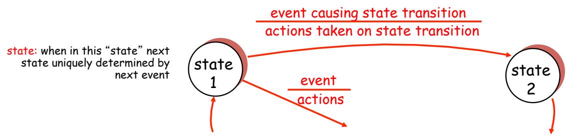
Reliable data transfer: getting started



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



Rdt1.0: reliable transfer over a reliable channel

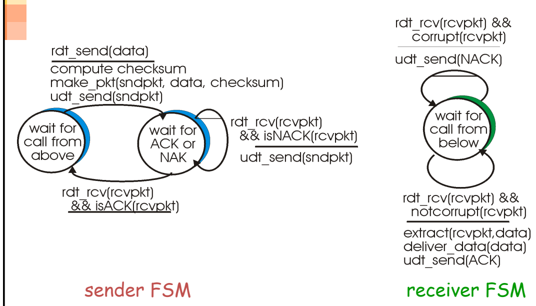
- underlying channel perfectly reliable
  - no bit erros
  - no loss of packets
- separate FSMs for sender, receiver:
  - sender sends data into underlying channel
  - receiver read data from underlying channel

(a) rdt1.0: sending side (b) rdt1.0: receiving side

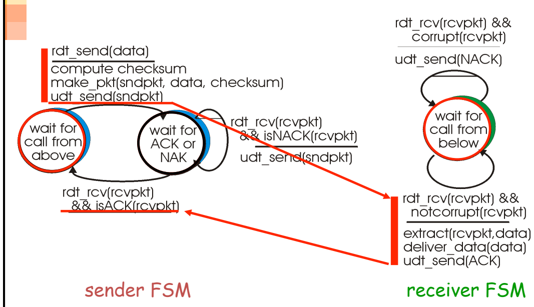
Rdt2.0: channel with bit errors

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

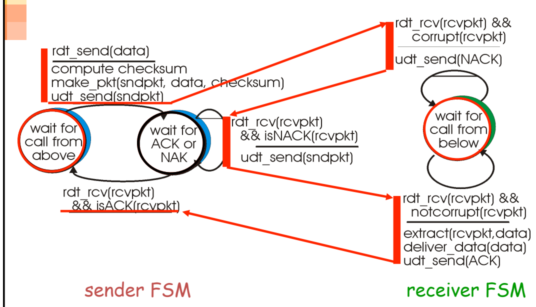
rdt2.0: FSM specification



rdt2.0: in action (no errors)



rdt2.0: in action (error scenario)



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

What to do?

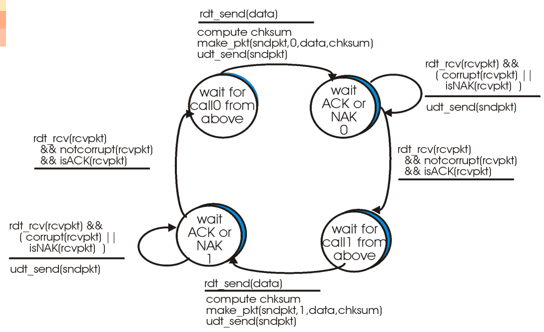
- sender ACKs/NAKs receiver's ACK/NAK? What if sender ACK/NAK lost?
- retransmit, but this might cause retransmission of correctly received pkt!

Handling duplicates:

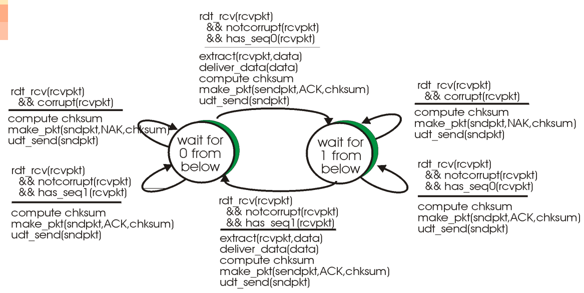
- sender adds *sequence number* to each pkt
- sender retransmits current pkt if ACK/NAK garbled
- receiver discards (doesn't deliver up) duplicate pkt

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

rdt2.1: sender, handles garbled ACK/NAKs



rdt2.1: receiver, handles garbled ACK/NAKs



rdt2.1: discussion

Sender:

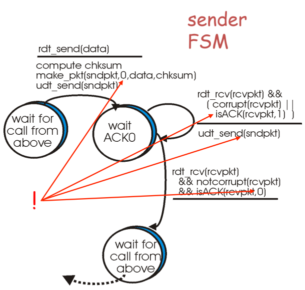
- seq # added to pkt
- two seq. #'s (0,1) will suffice. Why?
- must check if received ACK/NAK corrupted
- twice as many states
  - state must "remember" whether "current" pkt has 0 or 1 seq. #

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

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*



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

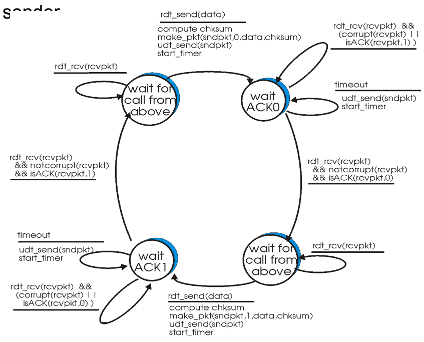
Q: how to deal with loss?

- sender waits until certain data or ACK lost, then retransmits
- yuck: drawbacks?

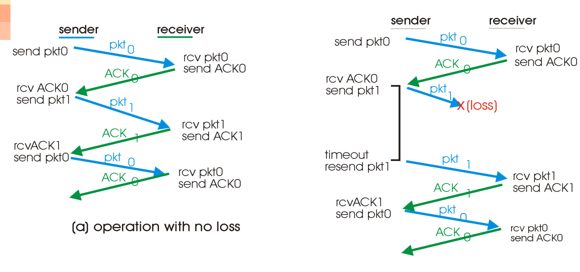
Approach: sender waits "reasonable" amount of time for ACK

- 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

rdt3.0 sender



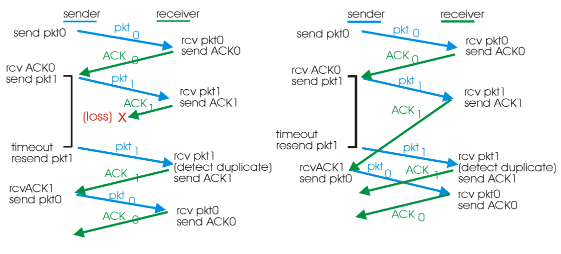
rdt3.0 in action



(a) operation with no loss

(b) lost packet

rdt3.0 in action



(c) lost ACK

(d) premature timeout

Need for Pipelining

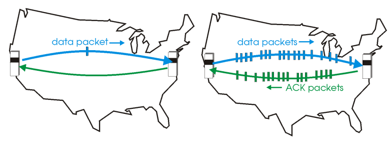
- 1 Gbps link, 15 ms e-e prop. delay, 1KB packet:

$$T_{transmit} = \frac{8Kb/pkt}{10^{**}9\ b/sec} = 8 \text{ microsec/packet}$$

$$\text{Utilization} = U = \frac{\text{fraction of time}}{\text{sender busy sending}} = \frac{8 \text{ microsec}}{30.008 \text{ msec}} = 0.00027$$

1KB pkt every 30 msec -> 33KB/sec thruput over 1 Gbps link

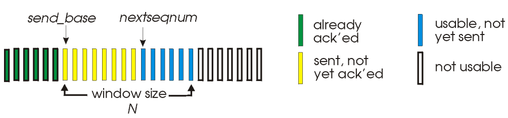
network protocol limits use of physical resources!



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

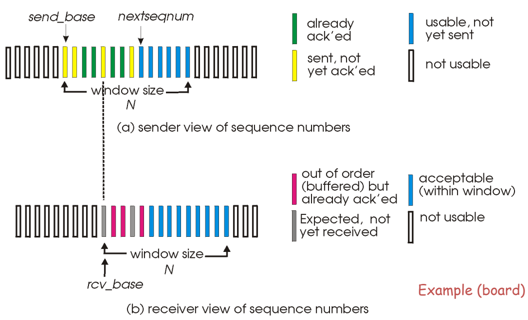
go-Back-N

- Sender



  - one timer
  - on timeout, retransmit all packets in window from last ACK + 1
- Receiver
  - cumulative ACK
  - discard out of order packets
- Example
- Advantages & disadvantages

Selective Repeat



Example (board)

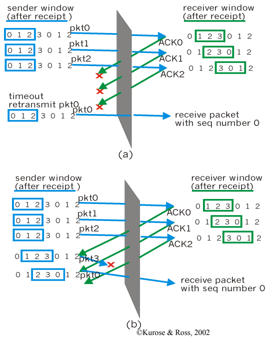
Computer Networking and Applications

Selective Repeat - window size

Example:

- seq #'s: 0, 1, 2, 3
- window size=3
- receiver sees no difference in two scenarios
- incorrectly passes duplicate data as new in (a)

Q: what is the relationship between seq# size and window size?



Computer Networking and Applications

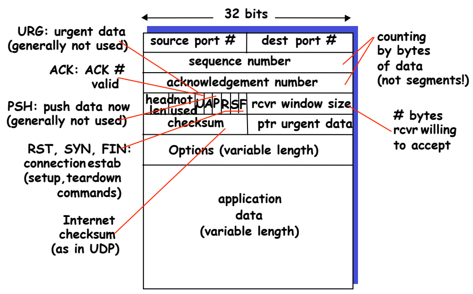
TCP

RFCs 793, 1122, 1323, 2018, 2581

- TCP is designed to provide the appearance of a reliable channel over the unreliable network layer (IP).
- In addition to the checksum and multiplexing as provided by UDP, TCP also provides flow control and congestion control and a reliable connection.
- The channel, or *connection*, is not a virtual circuit. All state information resides in the sending and receiving hosts, not in the routers.
- TCP deals with
  - Lost packets
  - Re-ordered packets
  - Delayed packets
- TCP is a modified hybrid of go-back-N and selective repeat. Cumulative ACKs. Only ACK up to correctly received segments.

Computer Networking and Applications

TCP segment structure



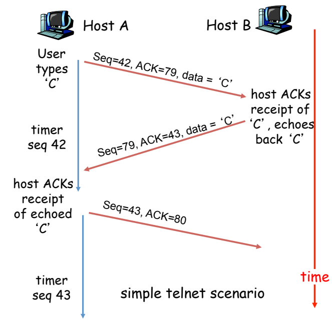
Computer Networking and Applications

Seq #s and ACKs in TCP

Seq. #'s:
- byte stream "number" of first byte in segment's data

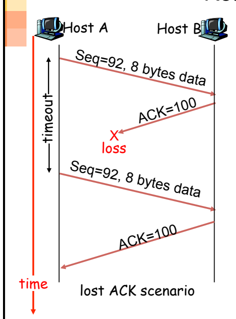
ACKs:
- seq # of next byte expected from other side
- cumulative ACK
- Piggybacked in data packets

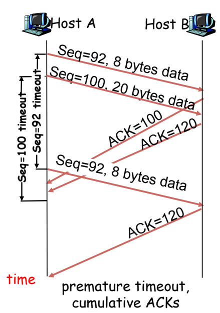
Q: how receiver handles out-of-order segments
- A: TCP spec doesn't say, - up to implementor



Computer Networking and Applications

Retransmission Scenarios





Computer Networking and Applications

How long should the timeout be?

- Timer should time out only rarely.
  - Ideally only when a segment is lost
  - Practically usually only when a segment is lost, may timeout on some long delays; but not average delays.
- The delay is expressed as a round trip time (RTT). Delay varies due to congestion and path. We need an estimate. Exponential Weighted Moving Average is used

`EstimatedRTT = (1 - x)*EstimatedRTT + x*SampleRTT`

Typically, existing estimate is weighted more (x = 0.125)

- To prevent delays from causing timeouts too frequently, the timer takes into consideration the deviation from the average.

`Timeout = EstimatedRTT + 4 * Deviation`

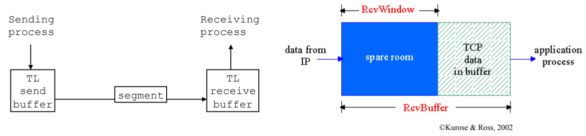
`Deviation=(1-x)*Deviation+x*|SampleRTT-EstimatedRTT|`

- Note that one ACK may acknowledge several segments, so calculating the timeout is non-trivial.

TCP - flow control

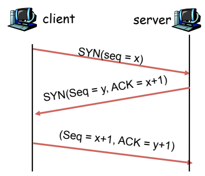
- Flow control exists to prevent the sender from overwhelming the receiver.
- Receiving host informs sender of the receive window size in the header of TCP segments (initially = receive buffer size)
- At sending host

  LastByteSent - LastByteAcked <= Receive Window
- What if the receive buffer is full? Receive window = 0. How will the sender know when more space is available?



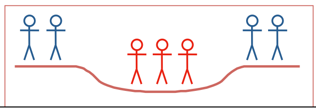
Establishing a Connection

- TCP connections are established by a 3-way handshake.
  - Request connection
  - Grant connection
  - Acknowledge
- Initial sequence numbers of the sender and receiver are exchanged as part of the connection establishment.
- A SYN bit in the header is set to 1 for the first two segments to indicate the set up.
- The third handshake is required to deal with duplicate segments/ACKs



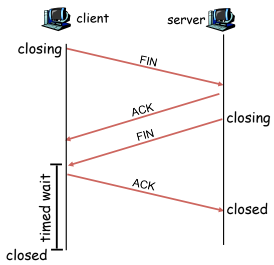
Terminating a Connection

- It is equally important to have an agreed termination of a connection
  - need to guarantee that all data has been delivered in both directions
- Such a guarantee is not possible
  (but assurance can be given with a high probability)
  - consider the problem in the diagram below
  - the blue armies (on the hills) are to attack the red army (in the valley)
  - the blue armies need to guarantee a coordinated attack



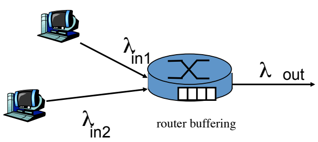
Terminating a connection in TCP

- timeouts at client and server guarantee that the connection will eventually be closed on both sides.
- timed wait increases the chance of server receiving a final ACK



Congestion

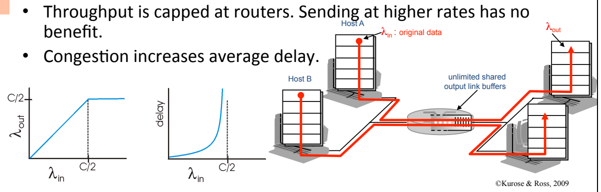
- Congestion can occur within the network. Too many sources and/ or too much data being sent for the routers.



- Can result in
  - packet loss (buffer overflows)
  - high delays (queuing in buffer)

Reasons to avoid Congestion

- Throughput is capped at routers. Sending at higher rates has no benefit.
- Congestion increases average delay.



- Retransmission of packets, dropped or delayed, adds overhead to network. For delayed packets, this is wasted overhead.
- When a packet is dropped, the effort of earlier routers, is wasted.
- Performance degrades exponentially as congestion level approaches capacity. We want to avoid reaching this state!

Approaches to avoiding congestion

- Network layer assisted
  - Routers provide explicit feedback of congestion and/or available rate.
  - ATM-ABR, Explicit Congestion Notification (TCP/IP - proposed)
- End-to-end
  - End systems attempt to infer congestion based on packet acknowledgment times, etc.
  - Current TCP/IP implementations
- Network layer assisted increases complexity of routers
- End-to-end may make an incorrect inference of congestion (e.g. in mobile networks)

TCP congestion control - Tahoe 1988

- In addition to the Receive Window (for flow control) the senders rate is also controlled by a Congestion Window

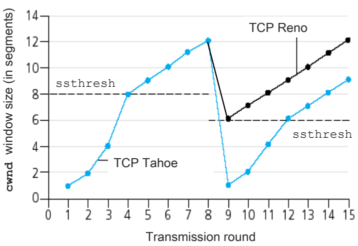
```
LastByteSent - LastByteAcked <= min (ReceiveWin, CongWin)
```

- Determining the congestion window

```
  – Initially CongWin = 1 Maximum Segment Size (MSS)
  – Set a threshold, Thresh = 65535 bytes
  – While CongWin <= Thresh
Slow    • Send a segment
Start   • If ACK is received before timeout,
            CongWin = CongWin + 1 MSS
        • If timeout occurs,
            Thresh = 1/2 min(ReceiveWin, CongWin), CongWin = 1 MSS
  – While CongWin > Thresh
Congestion • Send a segment
Avoidance  • If ACK is received before timeout,
            CongWin = CongWin + MSS * MSS/CongWin
        • If timeout occurs
            Thresh = 1/2 min(ReceiveWin, CongWin), CongWin = 1 MSS
```


TCP Congestion Control & extensions

- Extensions
- Reno - 1990
  - Fast retransmit
    - 3 duplicate ACKs - retransmit immediately
  - Fast recovery
    - 3 duplicate ACKs - stay in congestion avoidance, don't re-enter slow start
- Vegas - 1995
  - detect congestion before packet loss.



TCP Congestion Control - open issues

- Under congestion conditions, TCP provides fair sharing of available throughput
  - When a segment is not ACK'ed, the Threshold of senders will converge Send1 = 20/2 = 10 Send2 = 10/2 = 5
  - Only fair if each application has same number of TCP streams open!
- TCP doesn't spend much time in slow start (due to exponential increase) unless
  - Transmission rates are high relative to latency (not getting enough ACKs back to grow window size)
  - Object being sent is small (not enough time to escape slow start)
- Somewhat problematic for the web.
- UDP is often used to *avoid* TCP congestion control. This will be a problem if UDP traffic becomes more prevalent.
- TCP congestion control is still an active area of research with many variations on the basic Reno protocol.