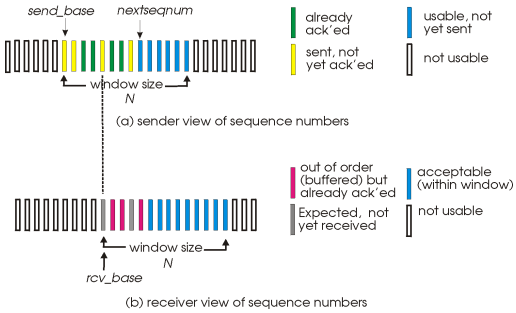
Selective Repeat

- receiver *individually* acknowledges all correctly received pkts
  - buffers pkts, as needed, for eventual in-order delivery to upper layer
- sender only resends pkts for which ACK not received
  - sender timer for each unACKed pkt
- sender window
  - N consecutive seq #'s
  - again limits seq #s of sent, unACKed pkts

Selective repeat: sender, receiver windows



Selective repeat

sender

data from above :
- if next available seq # in window, send pkt

timeout(n):
- resend pkt n, restart timer

ACK(n) in [sendbase,sendbase+N]:
- mark pkt n as received
- if n smallest unACKed pkt, advance window base to next unACKed seq #

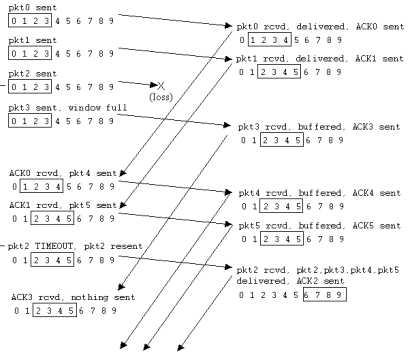
receiver

pkt n in [rcvbase, rcvbase+N-1]
- send ACK(n)
- out-of-order: buffer
- in-order: deliver (also deliver buffered, in-order pkts), advance window to next not-yet-received pkt

pkt n in [rcvbase-N,rcvbase-1]
- ACK(n)

otherwise:
- ignore

Selective repeat in action

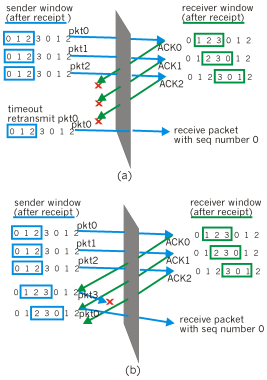


Selective repeat: dilemma

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 relationship between seq # size and window size?



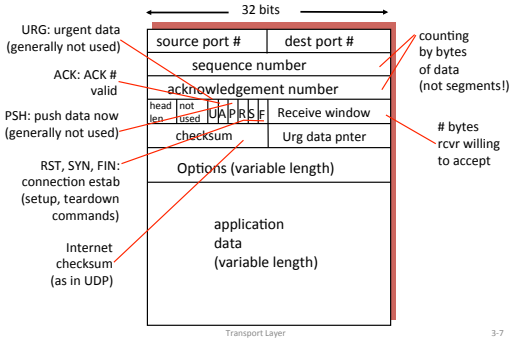
TCP: Overview

RFCs: 793, 1122, 1323, 2018, 2581

- point-to-point:
  - one sender, one receiver
- reliable, in-order *byte steam:*
  - no "message boundaries"
- pipelined:
  - TCP congestion and flow control set window size
- *send & receive buffers*

- full duplex data:
  - bi-directional data flow in same connection
  - MSS: maximum segment size
- connection-oriented:
  - handshaking (exchange of control msgs) init's sender, receiver state before data exchange
- flow controlled:
  - sender will not overwhelm receiver

TCP segment structure

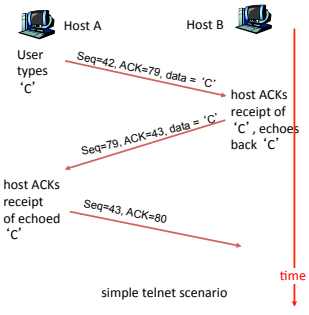


TCP seq. #'s and ACKs

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

ACKs:
- seq # of next byte expected from other side
- cumulative ACK

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



TCP reliable data transfer

- TCP creates rdt service on top of IP's unreliable service
- Pipelined segments
- Cumulative acks
- TCP uses single retransmission timer
- Retransmissions are triggered by:
  - timeout events
  - duplicate acks
- Initially consider simplified TCP sender:
  - ignore duplicate acks
  - ignore flow control, congestion control

TCP sender events:

data rcvd from app:
- Create segment with seq #
- seq # is byte-stream number of first data byte in segment
- start timer if not already running (think of timer as for oldest unacked segment)
- expiration interval: `TimeOutInterval`

timeout:
- retransmit segment that caused timeout
- restart timer

Ack rcvd:
- If acknowledges previously unacked segments
  - update what is known to be acked
  - start timer if there are outstanding segments

TCP sender (simplified)

```
NextSeqNum = InitialSeqNum
SendBase = InitialSeqNum

loop (forever) {
  switch(event)

  event: data received from application above
      create TCP segment with sequence number NextSeqNum
      if (timer currently not running)
          start timer
      pass segment to IP
      NextSeqNum = NextSeqNum + length(data)

  event: timer timeout
      retransmit not-yet-acknowledged segment with
          smallest sequence number
      start timer

  event: ACK received, with ACK field value of y
      if (y > SendBase) {
          SendBase = y
          if (there are currently not-yet-acknowledged segments)
              start timer
          }

} /* end of loop forever */
```

Comment:
- SendBase-1: last cumulatively ack'ed byte

Example:
- SendBase-1 = 71; y= 73, so the rcvr wants 73+ ; y > SendBase, so that new data is acked

TCP Round Trip Time and Timeout

Q: how to set TCP timeout value?
- longer than RTT
  - but RTT varies
- too short: premature timeout
  - unnecessary retransmissions
- too long: slow reaction to segment loss

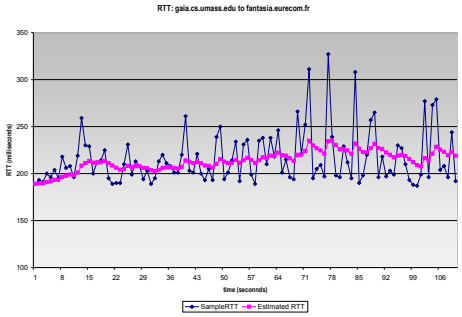
Q: how to estimate RTT?
- `SampleRTT`: measured time from segment transmission until ACK receipt
  - ignore retransmissions
- **`SampleRTT`** will vary, want estimated RTT "smoother"
  - average several recent measurements, not just current **`SampleRTT`**

TCP Round Trip Time and Timeout

$$\text{EstimatedRTT} = (1-\alpha)*\text{EstimatedRTT} + \alpha*\text{SampleRTT}$$

- Exponential weighted moving average
- influence of past sample decreases exponentially fast
- typical value: $\alpha = 0.125$

Example RTT estimation:



TCP Round Trip Time and Timeout

Setting the timeout

- **EstimtedRTT** plus "safety margin"
  - large variation in **EstimatedRTT** -> larger safety margin
- first estimate of how much SampleRTT deviates from EstimatedRTT:

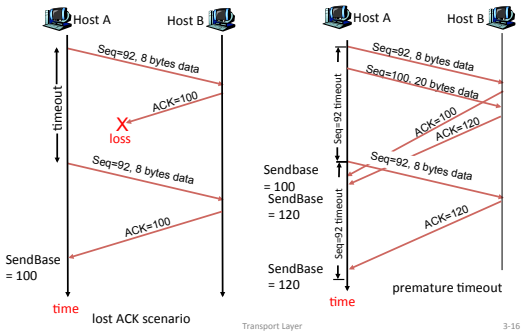
$$\text{DevRTT} = (1-\beta)*\text{DevRTT} + \beta*|\text{SampleRTT}-\text{EstimatedRTT}|$$

(typically, $\beta = 0.25$)

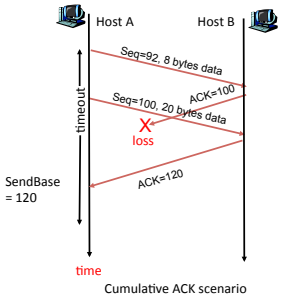
Then set timeout interval:

$$\text{TimeoutInterval} = \text{EstimatedRTT} + 4*\text{DevRTT}$$

TCP: retransmission scenarios



TCP retransmission scenarios (more)



Cumulative ACK scenario

TCP ACK generation [RFC 1122, RFC 2581]

Event at Receiver	TCP Receiver action
Arrival of in-order segment with expected seq #. All data up to expected seq # already ACKed	Delayed ACK. Wait up to 500ms for next segment. If no next segment, send ACK
Arrival of in-order segment with expected seq #. One other segment has ACK pending	Immediately send single cumulative ACK, ACKing both in-order segments
Arrival of out-of-order segment higher-than-expect seq. # . Gap detected	Immediately send duplicate ACK, indicating seq. # of next expected byte
Arrival of segment that partially or completely fills gap	Immediate send ACK, provided that segment startsat lower end of gap

Fast Retransmit

- Time-out period often relatively long:
  - long delay before resending lost packet
- Detect lost segments via duplicate ACKs.
  - Sender often sends many segments back-to-back
  - If segment is lost, there will likely be many duplicate ACKs.
- If sender receives 3 ACKs for the same data, it supposes that segment after ACKed data was lost:
  - fast retransmit: resend segment before timer expires

Fast retransmit algorithm:

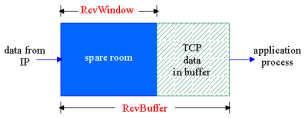
```
event: ACK received, with ACK field value of y
        if (y > SendBase) {
            SendBase = y
            if (there are currently not-yet-acknowledged segments)
                start timer
        }
        else {
            increment count of dup ACKs received for y
            if (count of dup ACKs received for y = 3) {
                resend segment with sequence number y
            }
```

a duplicate ACK for already ACKed segment

fast retransmit

TCP Flow Control

- receive side of TCP connection has a receive buffer:

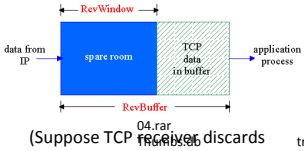


- speed-matching service: matching the send rate to the receiving app's drain rate

flow control: sender won't overflow receiver's buffer by transmitting too much, too fast

- app process may be slow at reading from buffer

TCP Flow control: how it works



(Suppose TCP receiver discards out-of-order segments)

- spare room in buffer

```
= RcvWindow
= RcvBuffer-[LastByteRcvd -
  LastByteRead]
```

- Rcvr advertises spare room by including value of **`RcvWindow`** in segments
- Sender limits unACKed data to **`RcvWindow`**
  - guarantees receive buffer doesn't overflow

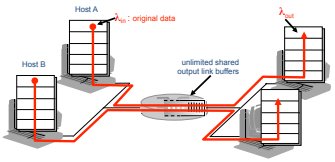
Principles of Congestion Control

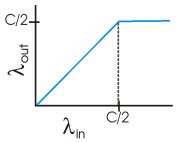
Congestion:

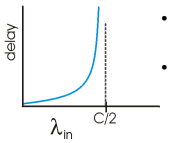
- informally: "too many sources sending too much data too fast for *network* to handle"
- different from flow control!
- manifestations:
  - lost packets (buffer overflow at routers)
  - long delays (queueing in router buffers)
- a top-10 problem!

Causes/costs of congestion: scenario 1

- two senders, two receivers
- one router, infinite buffers
- no retransmission



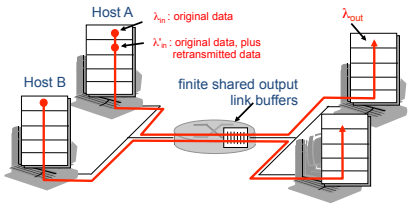




- large delays when congested
- maximum achievable throughput

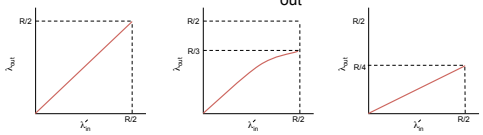
Causes/costs of congestion: scenario 2

- one router, *finite* buffers
- sender retransmission of lost packet



Causes/costs of congestion: scenario 2

- always: $\lambda_{in} = \lambda_{out}$ (goodput)
- "perfect" retransmission only when loss: $\lambda'_{in} > \lambda_{out}$
- retransmission of delayed (not lost) packet makes λ'_{in} larger (than perfect case) for same λ_{out}



a. b. c.

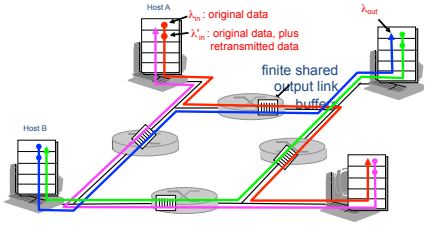
"costs" of congestion:

- more work (retrans) for given "goodput"
- unneeded retransmissions: link carries multiple copies of pkt

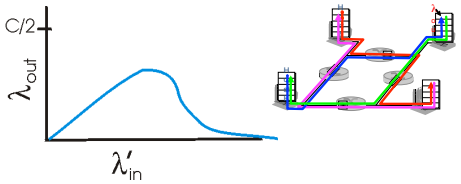
Causes/costs of congestion: scenario 3

- four senders
- multihop paths
- timeout/retransmit

Q: what happens as λ_{in} and λ'_{in} increase ?



Causes/costs of congestion: scenario 3



Another "cost" of congestion:

- when packet dropped, any "upstream transmission capacity used for that packet was wasted!