CS162
Operating Systems and Systems Programming
Lecture 17
TCP, Flow Control, Reliability

November 4, 2013
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Quiz 16.2: Layering

- Q1: True _ False _ Layering improves application performance
- Q2: True _ False _ Routers forward a packet based on its destination address
- Q3: True _ False _ "Best Effort" packet delivery ensures that packets are delivered in order
- Q4: True _ False _ Port numbers belong to network layer
- Q5: True _ False _ The hosts on Berkeley's campus share the same IP address prefix

Quiz 16.2: Layering

- Q1: True _ False X Layering improves application performance
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- Q3: True _ False X "Best Effort" packet delivery ensures that packets are delivered in order
- Q4: True _ False X Port numbers belong to network layer
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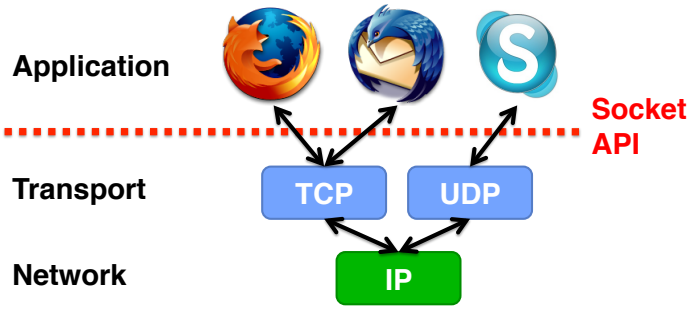
Goals for Today

- Socket API
- TCP
  - Open connection (3-way handshake)
  - Reliable transfer
  - Tear-down connection
  - Flow control

Socket API

- Socket API: Network programming interface



BSD Socket API

- Created at UC Berkeley (1980s)
- Most popular network API
- Ported to various OSes, various languages
  - Windows Winsock, BSD, OS X, Linux, Solaris, …
  - Socket modules in Java, Python, Perl, …
- Similar to Unix file I/O API
  - In the form of *file descriptor* (sort of handle).
  - Can share same `read()`/`write()`/`close()` system calls

TCP: Transport Control Protocol

- Reliable, in-order, and at most once delivery
- Stream oriented: messages can be of arbitrary length
- Provides multiplexing/demultiplexing to IP
- Provides congestion and flow control
- Application examples: file transfer, chat

TCP Service

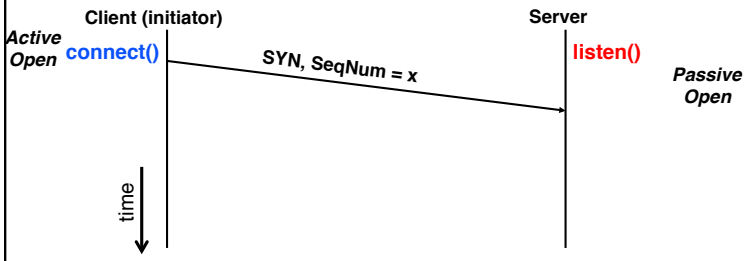
1) Open connection: 3-way handshaking
2) Reliable byte stream transfer from (IPa, TCP_Port1) to (IPb, TCP_Port2)
   - Indication if connection fails: Reset
3) Close (tear-down) connection

Open Connection: 3-Way Handshaking

- Goal: agree on a set of parameters, i.e., the start sequence number for each side
  - Starting sequence number: sequence of first byte in stream
  - Starting sequence numbers are random

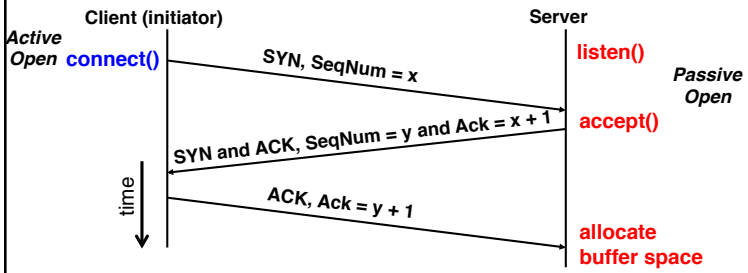
Open Connection: 3-Way Handshaking

- Server waits for new connection calling listen()
- Sender call connect() passing socket which contains server's IP address and port number
  - OS sends a special packet (SYN) containing a proposal for first sequence number, x



Open Connection: 3-Way Handshaking

- If it has enough resources, server calls accept() to accept connection, and sends back a SYN ACK packet containing
  - Client's sequence number incremented by one, (x + 1)
    - » Why is this needed?
  - A sequence number proposal, y, for first byte server will send

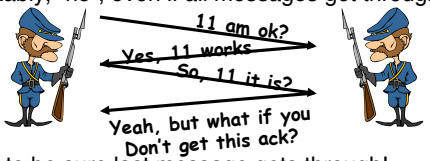


3-Way Handshaking (cont'd)

- Three-way handshake adds 1 RTT delay
- Why?
  - Congestion control: SYN (40 byte) acts as cheap probe
  - Protects against delayed packets from other connection (would confuse receiver)

General's Paradox

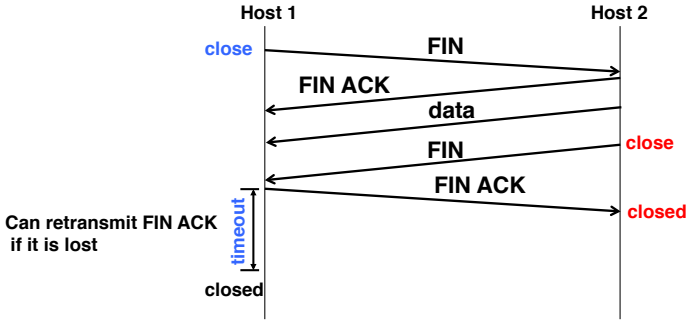
- General's paradox:
  - Constraints of problem:
    - » Two generals, on separate mountains
    - » Can only communicate via messengers
    - » Messengers can be captured
  - Problem: need to coordinate attack
    - » If they attack at different times, they all die
    - » If they attack at same time, they win
  - Named after Custer, who died at Little Big Horn because he arrived a couple of days too early
- Can messages over an unreliable network be used to guarantee two entities do something simultaneously?
  - Remarkably, "no", even if all messages get through



  - No way to be sure last message gets through!

Close Connection

- Goal: both sides agree to close the connection
- 4-way connection tear down



Reliable Transfer

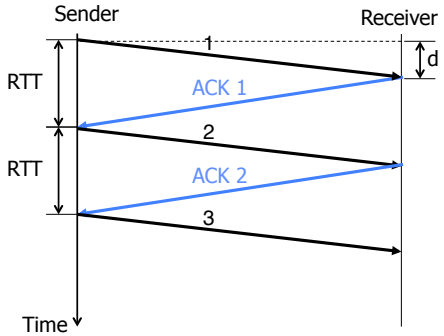
- Retransmit missing packets
  - Numbering of packets and ACKs
- Do this efficiently
  - Keep transmitting whenever possible
  - Detect missing packets and retransmit quickly
- Two schemes
  - Stop & Wait
  - Sliding Window (Go-back-n and Selective Repeat)

Detecting Packet Loss?

- Timeouts
  - Sender timeouts on not receiving ACK
- Missing ACKs
  - Receiver ACKs each packet
  - Sender detects a missing packet when seeing a gap in the sequence of ACKs
  - Need to be careful! Packets and ACKs might be reordered
- NACK: Negative ACK
  - Receiver sends a NACK specifying a packet it is missing

Stop & Wait w/o Errors

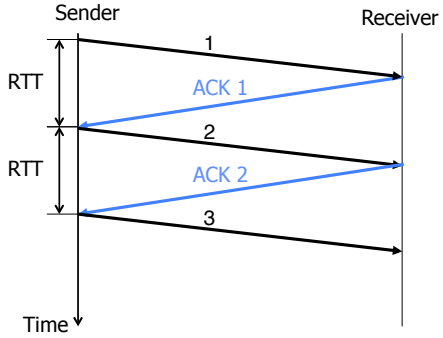
- Send; wait for ack; repeat
- RTT: Round Trip Time (RTT): time it takes a packet to travel from sender to receiver and back
  - One-way latency (d): one way delay from sender and receiver



RTT = 2*d
(if latency is symmetric)

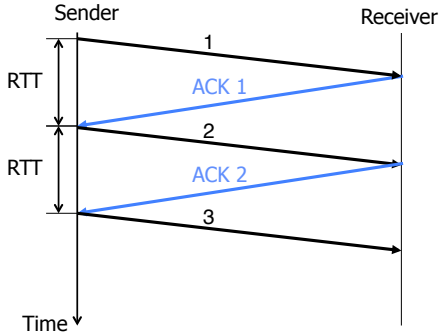
Stop & Wait w/o Errors

- How many packets can you send?
- 1 packet / RTT
- Throughput: number of bits delivered to receiver per sec



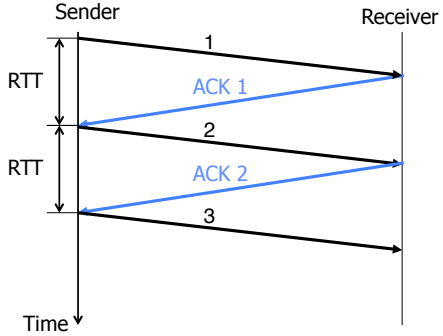
Stop & Wait w/o Errors

- Say, RTT = 100ms
- 1 packet = 1500 bytes
- Throughput = 1500*8bits/0.1s = 120 Kbps



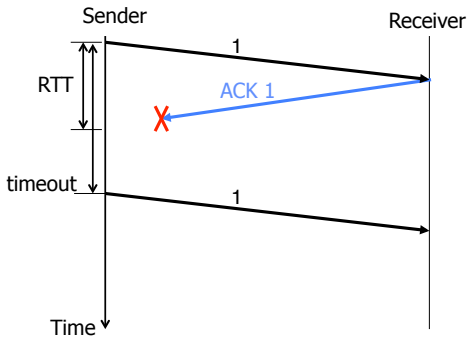
Stop & Wait w/o Errors

- Can be highly inefficient for high capacity links
- Throughput doesn't depend on the network capacity → even if capacity is 1Gbps, we can only send 120 Kbps!



Stop & Wait with Errors

- If a loss wait for a retransmission timeout and retransmit
- Ho do you pick the timeout?

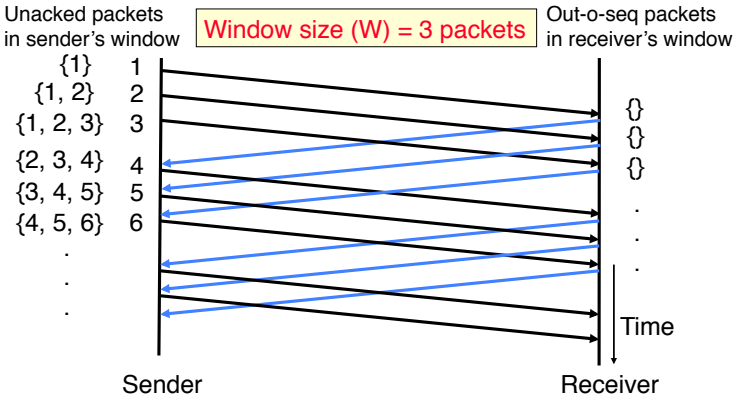


Sliding Window

- *window* = set of adjacent sequence numbers
- The size of the set is the *window size*
- Assume window size is n
- Let A be the last ACK'd packet of sender without gap; then window of sender = {A+1, A+2, …, A+n}
- Sender can send packets in its window
- Let B be the last received packet without gap by receiver, then window of receiver = {B+1,…, B+n}
- Receiver can accept out of sequence, if in window

Sliding Window w/o Errors

- Throughput = W*packet_size/RTT



Example: Sliding Window w/o Errors

- Assume
  - Link capacity, C = 1Gbps
  - Latency between end-hosts, RTT = 80ms
  - packet_length = 1000 bytes
- What is the window size W to match link's capacity, C?
- Solution

  We want Throughput = C

  Throughput = W*packet_size/RTT

  C = W*packet_size/RTT

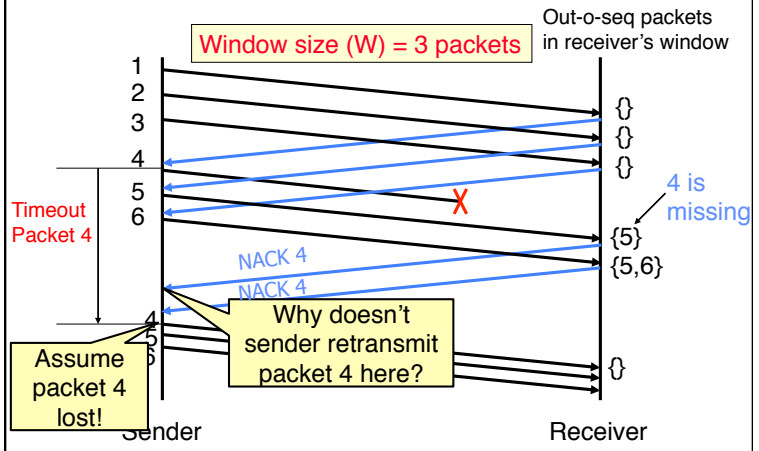
  W = C*RTT/packet_size = 10^9bps*80*10^{-3}s/(8000b) = 10^4 packets

Window size ~ Bandwidth (Capacity), delay (RTT/2)

Sliding Window with Errors

- Two approaches
  - Go-Back-n (GBN)
  - Selective Repeat (SR)
- In the absence of errors they behave identically
- Go-Back-n (GBN)
  - Transmit up to *n* unacknowledged packets
  - If timeout for ACK(*k*), retransmit *k*, *k+1*, …
  - Typically uses NACKs instead of ACKs
    - » Recall, NACK specifies first in-sequence packet missed by receiver

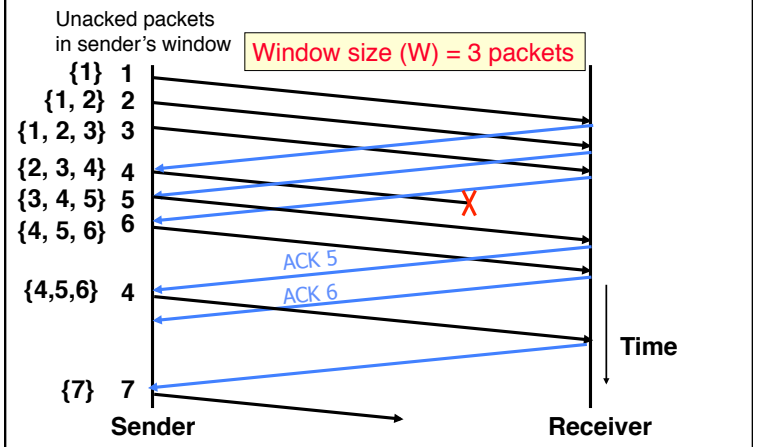
GBN Example with Errors



Selective Repeat (SR)

- Sender: transmit up to *n* unacknowledged packets
- Assume packet *k* is lost
- Receiver: indicate packet *k* is missing (use ACKs)
- Sender: retransmit packet *k*

SR Example with Errors



Summary

- TCP: Reliable Byte Stream
  - Open connection (3-way handshaking)
  - Close connection: no perfect solution; no way for two parties to agree in the presence of arbitrary message losses (General's Paradox)
- Reliable transmission
  - S&W not efficient for links with large capacity (bandwidth) delay product
  - Sliding window more efficient but more complex

CS 162 Collaboration Policy

Discussing algorithms/testing strategies with other groups

Helping debug someone else's code (in another group)

Sharing code or test cases with another group

Copying OR reading another group's code or test cases

Copying OR reading online code or test cases from from prior years

We are comparing all project submissions against all submissions from several prior years and will take actions (described on the course overview page) against offenders

We know multiple students are using code from prior years – notify us *immediately* if you have and we will show leniency

Tue 11:59PM deadline to email cs162@cory

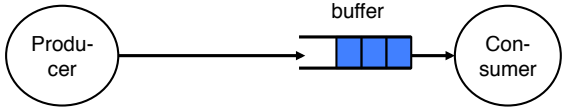
Our Services

- We have 2 faculty and 4 TAs ready to help you in this class
  - 12 hours of office hours every week
- Life at Berkeley can be very stressful!
- Campus has resources to help you cope
  - Dr. Shuangmei (Christine) Zhou in 241 Bechtel (christinez@uhs.berkeley.edu, 510-643-7850) has drop-in hours Tue 2-4pm and Wed 10am-12pm
  - The Tang center (http://www.uhs.berkeley.edu/students/counseling, 510-642-9494) has walk-in hours Mon-Fri 10am-5pm and their after hours number is 877-211-3686
  - *All discussions with Christine or other Tang center personnel are completely confidential and will not be shared with anyone*

5min Break

Flow Control

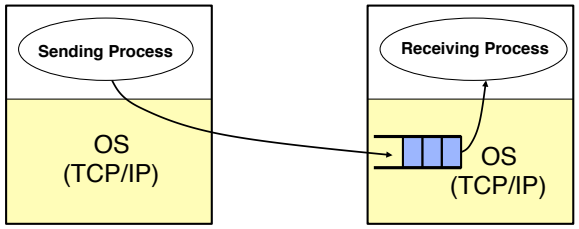
- Recall: Flow control ensures a fast sender does not overwhelm a slow receiver
- Example: Producer-consumer with bounded buffer (Lecture 5)
  - A buffer between producer and consumer
  - Producer puts items into buffer as long as buffer **not full**
  - Consumer consumes items from buffer



TCP Flow Control

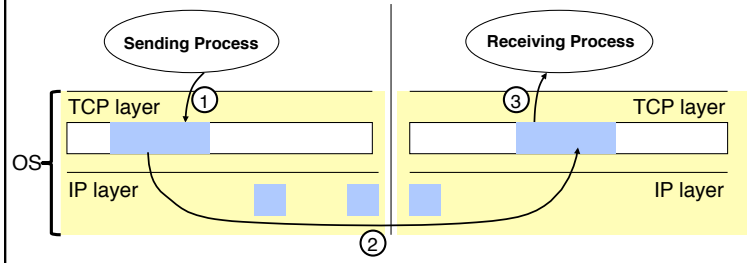
- TCP: sliding window protocol at byte (not packet) level
  - Go-back-N: TCP Tahoe, Reno, New Reno
  - Selective Repeat (SR): TCP Sack
- Receiver tells sender how many more bytes it can receive without overflowing its buffer (i.e., AdvertisedWindow)
- The ACK contains sequence number N of next byte the receiver expects, i.e., receiver has received all bytes in sequence up to and including N-1

TCP Flow Control



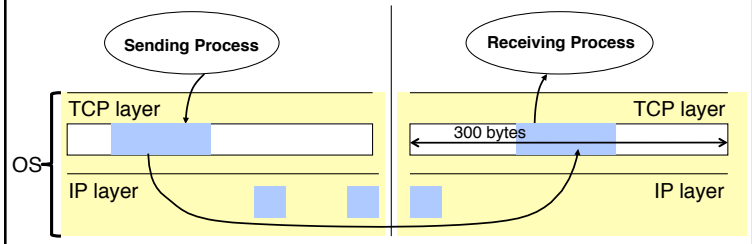
- TCP/IP implemented by OS (Kernel)
  - Cannot do context switching on sending/receiving every packet
    - » At 1Gbps, it takes 12 usec to send an 1500 bytes, and 0.8usec to send an 100 byte packet
- Need buffers to match …
  - sending app with sending TCP
  - receiving TCP with receiving app

TCP Flow Control



- Three pairs of producer-consumer's
  - ① sending process → sending TCP
  - ② Sending TCP → receiving TCP
  - ③ receiving TCP → receiving process

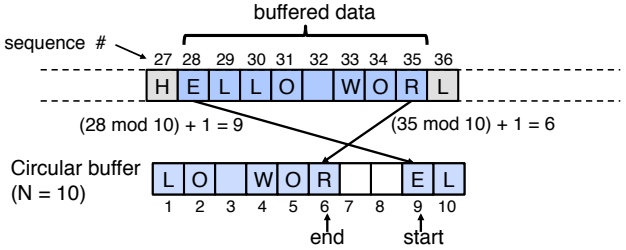
TCP Flow Control



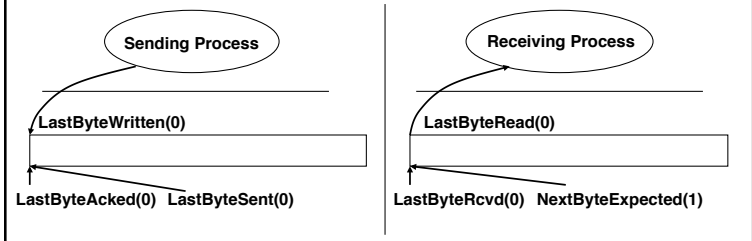
- Example assumptions:
  - Maximum IP packet size = 100 bytes
  - Size of the receiving buffer (MaxRcvBuf) = 300 bytes
- Recall, ack indicates the next expected byte in-sequence, not the last received byte
- Use circular buffers

Circular Buffer

- Assume
  - A buffer of size N
  - A stream of bytes, where bytes have increasing sequence numbers
    - » Think of stream as an unbounded array of bytes and of sequence number as indexes in this array
- Buffer stores at most N consecutive bytes from the stream
- Byte k stored at position (k mod N) + 1 in the buffer

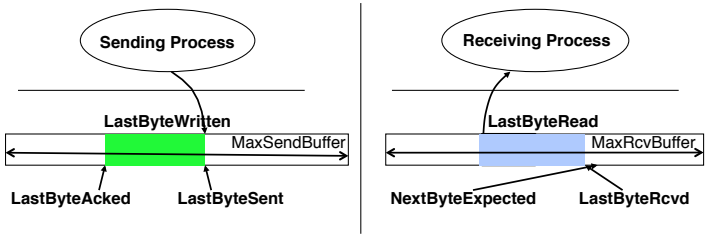


TCP Flow Control



- LastByteWritten: last byte written by sending process
- LastByteSent: last byte sent by sender to receiver
- LastByteAcked: last ack received by sender from receiver
- LastByteRcvd: last byte received by receiver from sender
- NextByteExpected: last in-sequence byte expected by receiver
- LastByteRead: last byte read by the receiving process

TCP Flow Control



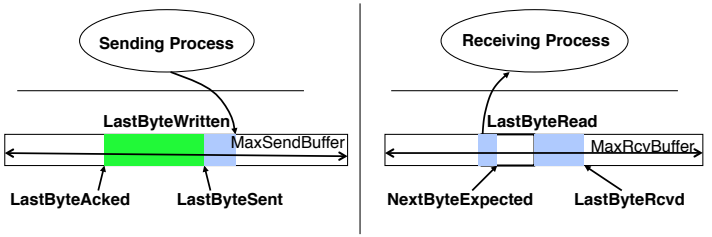
- AdvertisedWindow: number of bytes TCP receiver can receive

AdvertisedWindow = MaxRcvBuffer – (LastByteRcvd – LastByteRead)

- SenderWindow: number of bytes TCP sender can send

SenderWindow = AdvertisedWindow – (LastByteSent – LastByteAcked)

TCP Flow Control



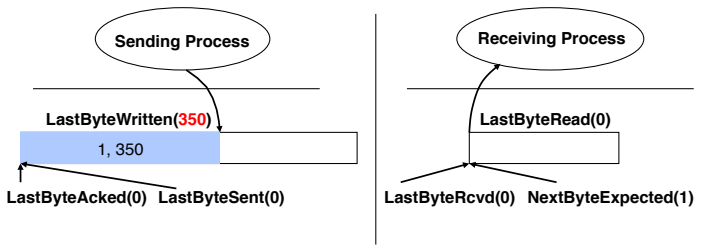
- Still true if receiver missed data….

AdvertisedWindow = MaxRcvBuffer – (LastByteRcvd – LastByteRead)

- WriteWindow: number of bytes sending process can write

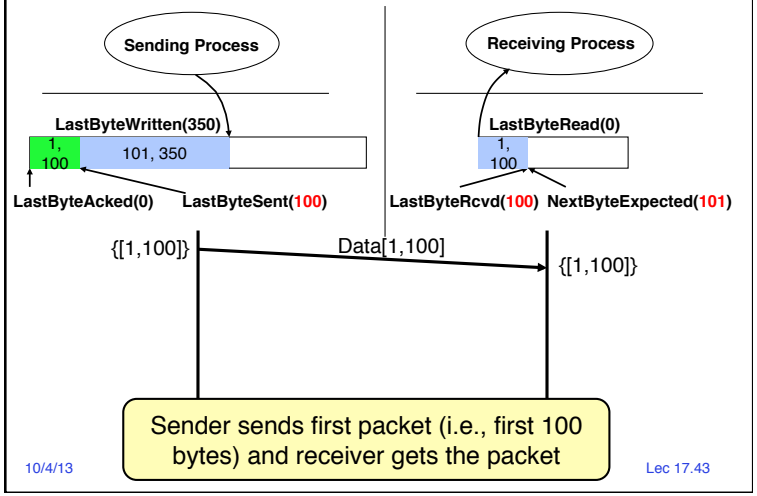
WriteWindow = MaxSendBuffer – (LastByteWritten – LastByteAcked)

TCP Flow Control

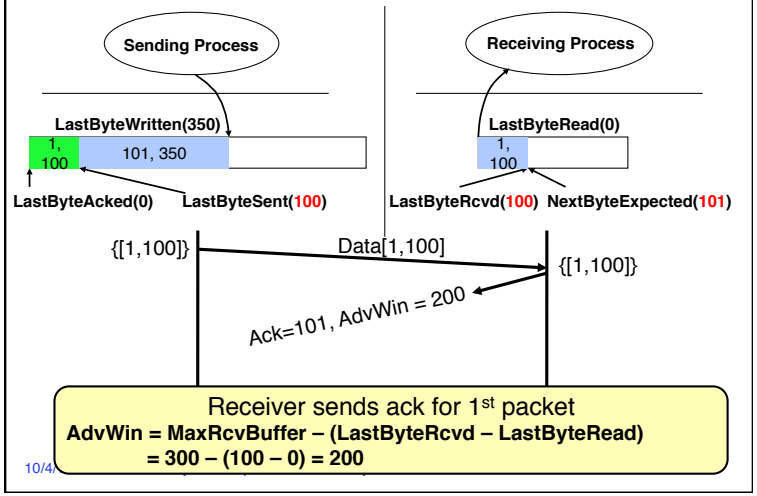


- Sending app sends 350 bytes
- Recall:
  - We assume IP only accepts packets no larger than 100 bytes
  - MaxRcvBuf = 300 bytes, so initial Advertised Window = 300 byets

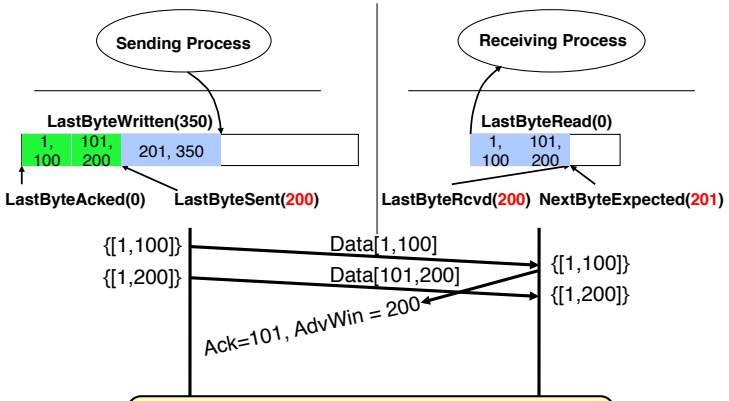
TCP Flow Control



TCP Flow Control

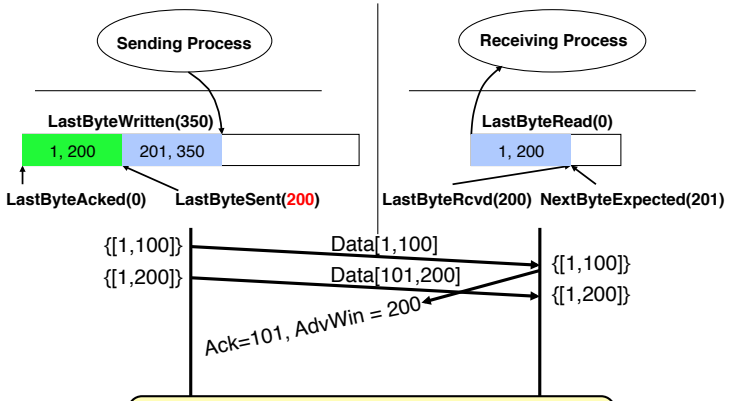


TCP Flow Control



Sender sends 2nd packet (i.e., next 100 bytes) and receiver gets the packet

TCP Flow Control



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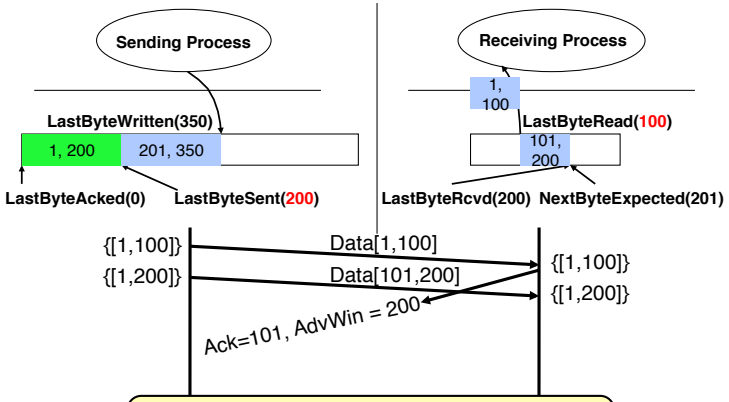
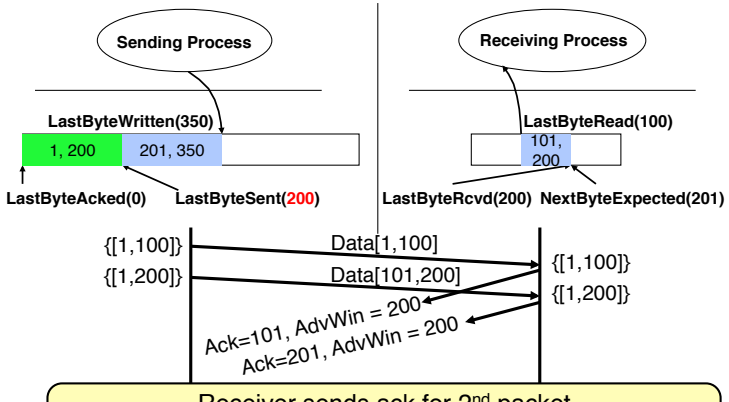


Receiving TCP delivers first 100 bytes to receiving process

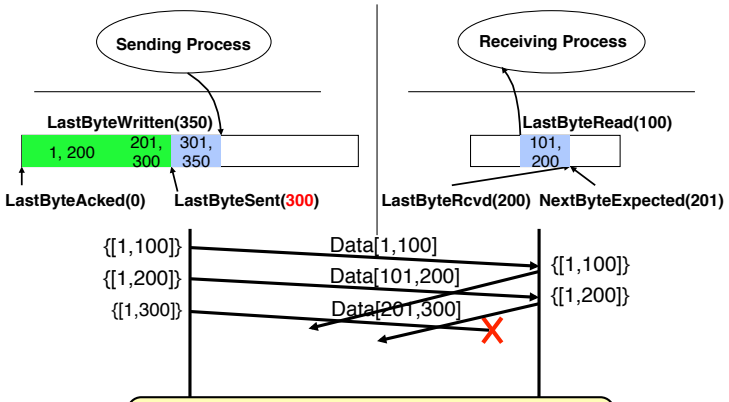
TCP Flow Control



Receiver sends ack for 2nd packet

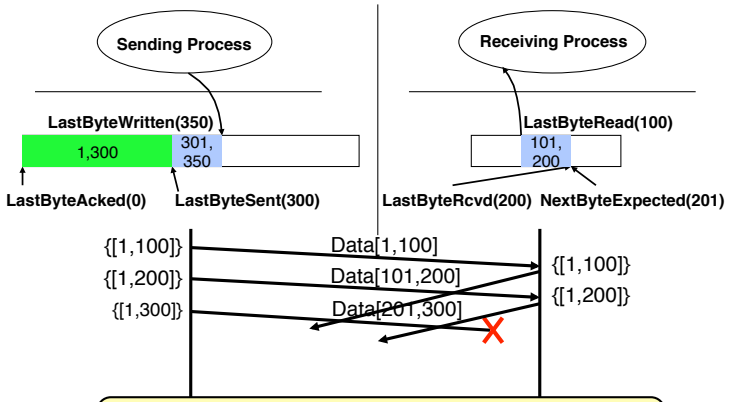
AdvWin = MaxRcvBuffer – (LastByteRcvd – LastByteRead)
= 300 – (200 – 100) = 200

TCP Flow Control



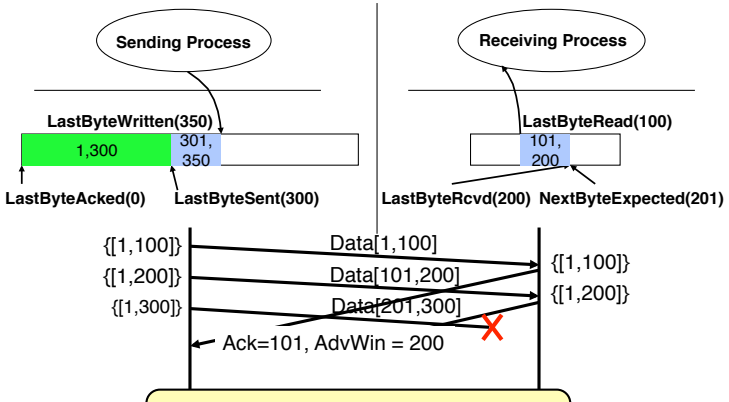
Sender sends 3rd packet (i.e., next 100 bytes) and the packet is lost

TCP Flow Control



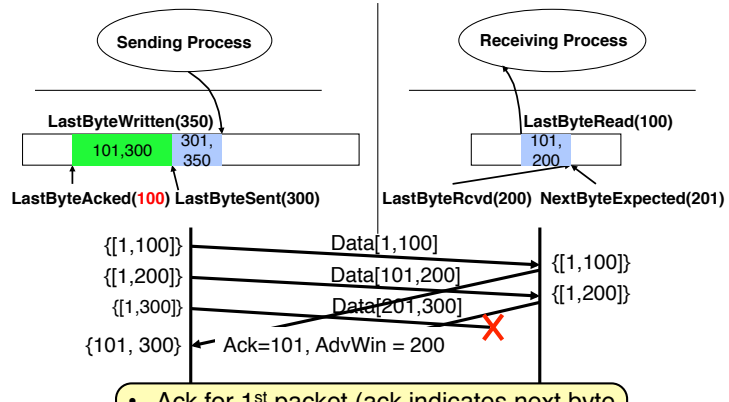
Sender stops sending as window full
SndWin = AdvWin – (LastByteSent – LastByteAcked)
= 300 – (300 – 0) = 0

TCP Flow Control



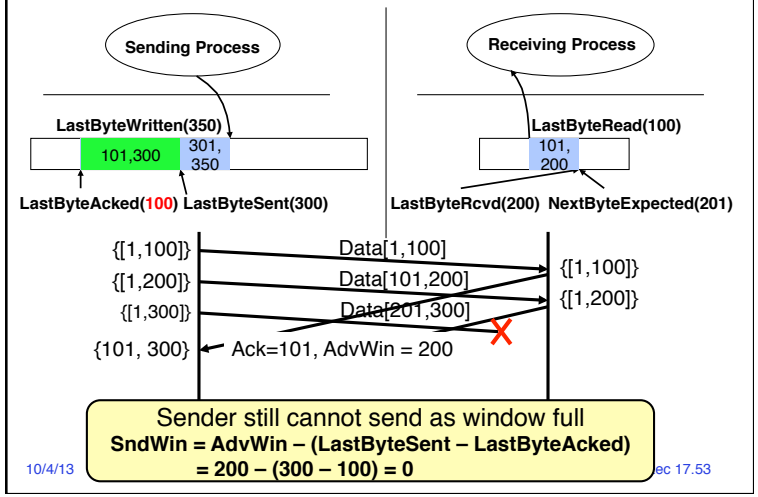
- Sender gets ack for 1st packet
- AdWin = 200

TCP Flow Control



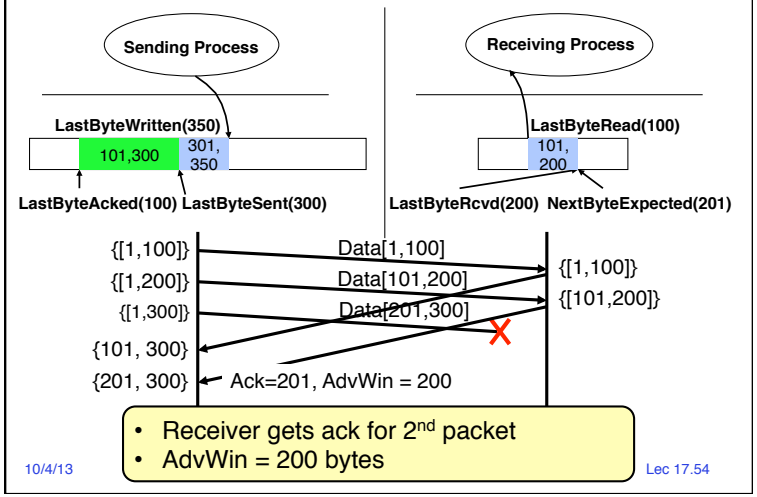
- Ack for 1st packet (ack indicates next byte expected by receiver)
- Receiver no longer needs first 100 bytes

TCP Flow Control



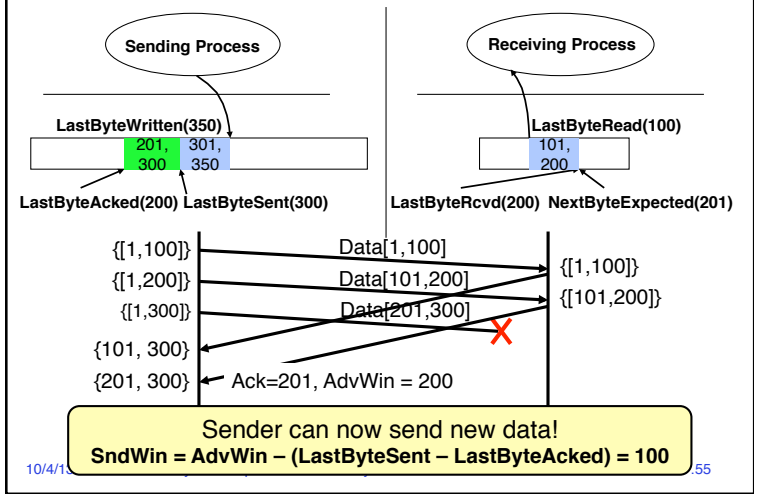
Sender still cannot send as window full
SndWin = AdvWin – (LastByteSent – LastByteAcked)
= 200 – (300 – 100) = 0

TCP Flow Control



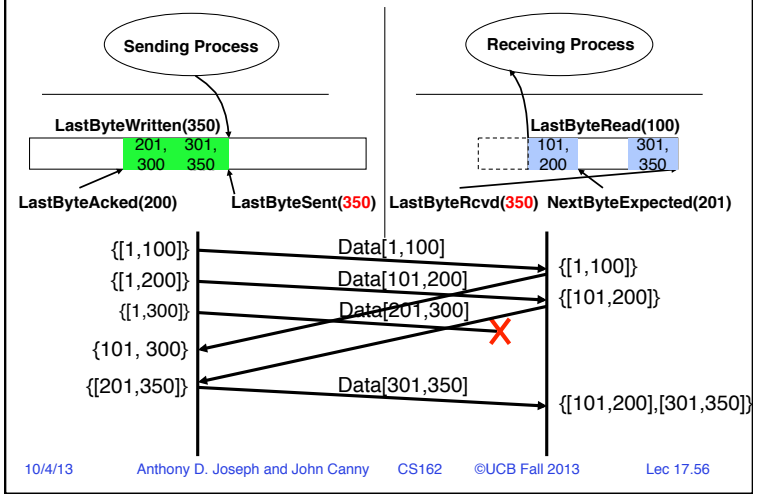
- Receiver gets ack for 2nd packet
- AdvWin = 200 bytes

TCP Flow Control

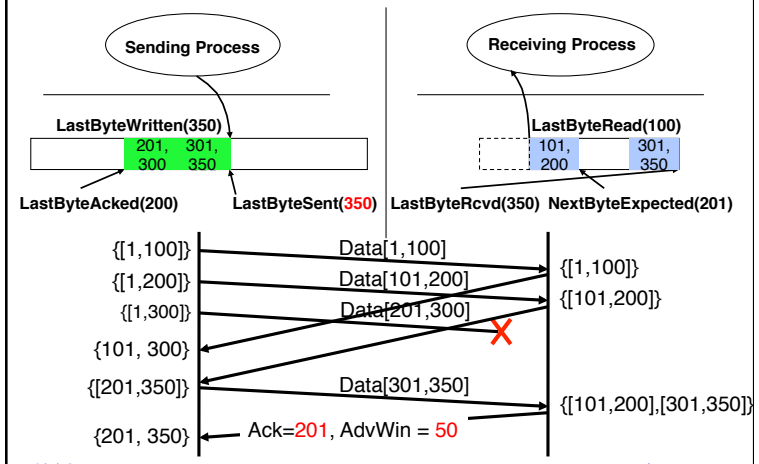


Sender can now send new data!
SndWin = AdvWin – (LastByteSent – LastByteAcked) = 100

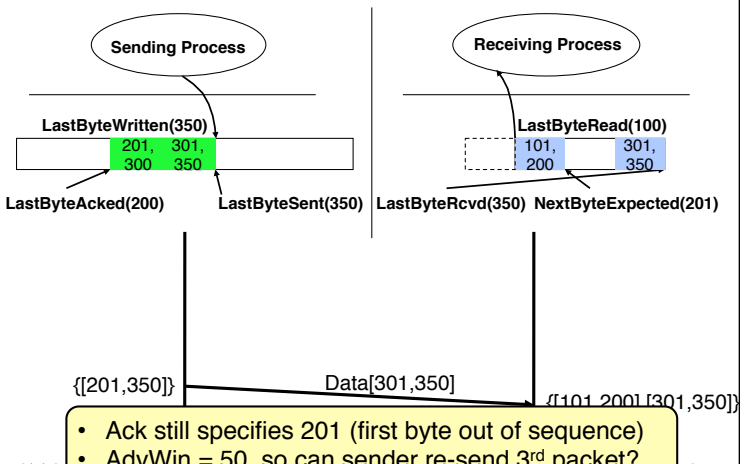
TCP Flow Control



TCP Flow Control

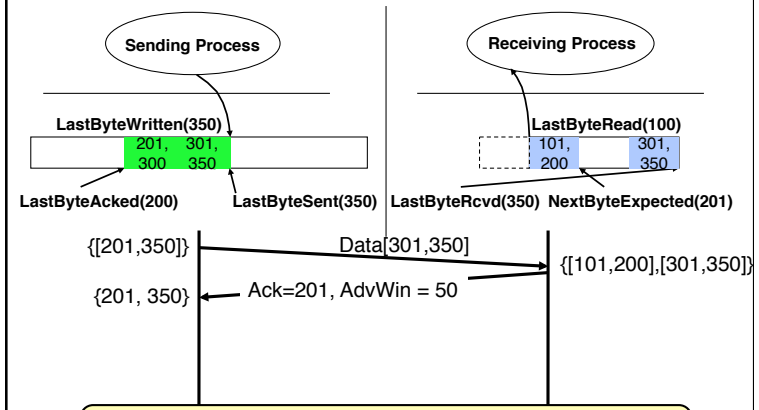


TCP Flow Control



- Ack still specifies 201 (first byte out of sequence)
- AdvWin = 50, so can sender re-send 3rd packet?

TCP Flow Control



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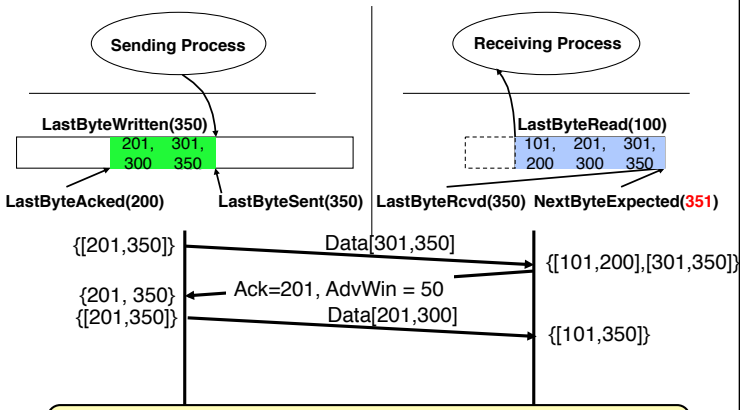
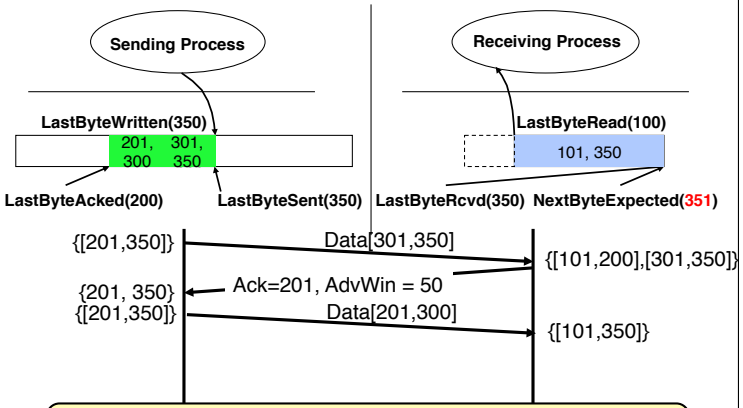
TCP Flow Control



Yes! Sender can re-send 2nd packet since it's in existing window – won't cause receiver window to grow

TCP Flow Control



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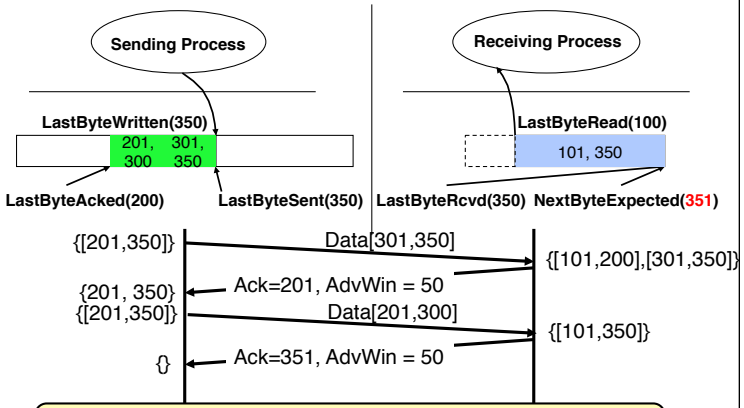
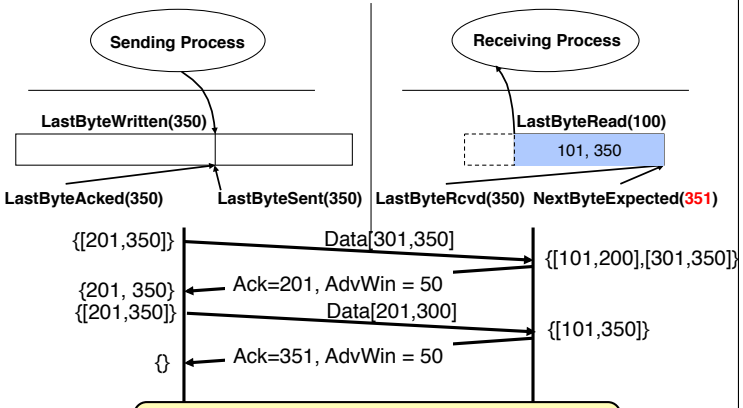
TCP Flow Control



- Sender gets 3rd packet and sends Ack for 351
- AdvWin = 50

TCP Flow Control



Sender DONE with sending all bytes!

Discussion

- Why not have a huge buffer at the receiver (memory is cheap!)?
- Sending window (SndWnd) also depends on network congestion
  - **Congestion control**: ensure that a fast sender doesn't overwhelm a router in the network (discussed in detail in EE122)
- In practice there is another set of buffers in the protocol stack, at the **link layer** (i.e., Network Interface Card)

Summary: Reliability & Flow Control

- Flow control: three pairs of producer consumers
  - Sending process → sending TCP
  - Sending TCP → receiving TCP
  - Receiving TCP → receiving process
- AdvertisedWindow: tells sender how much new data the receiver can buffer
- SenderWindow: specifies how many more bytes the sending application can send to the sending OS
  - Depends on AdvertisedWindow and on data sent since sender received AdvertisedWindow

Summary: Networking (Internet Layering)