

CS450 – Introduction to Networking Lecture 21 – Midterm review

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The midterm exam

- 6 questions = 28 points
 - Maximum point for midterm is 25 (you got 3 bonus points)
- Only one letter sheet of notes allowed
- Content
 - HTTP protocol
 - Email system
 - DNS
 - P2P
 - Transport layer
 - UDP/TCP
 - Flow control and congestion control

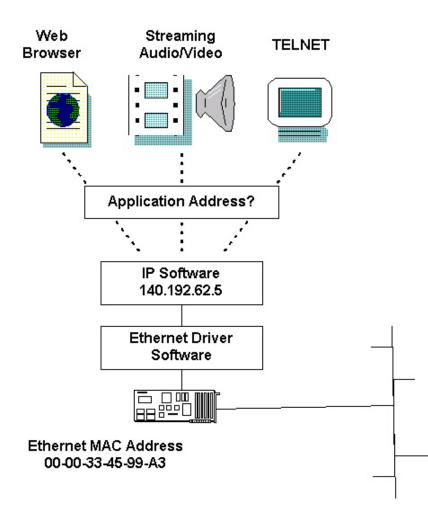
Internet protocol stack

- **application:** supporting network applications – FTP, SMTP, HTTP
- transport: process-process data transfer
 - TCP, UDP
- network: routing of datagrams from source to destination - IP, routing protocols
- link: data transfer between neighboring network elements – Ethernet, 802.111 (WiFi), PPP
 • physical: bits "on the wire"

	application
	transport
	network
	link
Γ	physical

Addressing processes

- identifier includes both IP address and port numbers associated with process on host.
- example port numbers:
 - HTTP server: 80
 - mail server: 25
- to send HTTP message to www.cs.uic.edu web server:
 - IP address: 131.193.32.29
 - port number: 80
- more shortly...



Internet transport protocols services

TCP service:

- reliable transport between sending and receiving process
- *flow control*: sender won't overwhelm receiver
- congestion control: throttle sender when network overloaded
- does not provide: timing, minimum throughput guarantee, security
- connection-oriented: setup required between client and server processes

UDP service:

- unreliable data transfer between sending and receiving process
- does not provide: reliability, flow control, congestion control, timing, throughput guarantee, security, or connection setup,

Q: why bother? Why is there a UDP?

HTTP connections

non-persistent HTTP

- at most one object sent over TCP connection
 - connection then closed
- downloading multiple objects required multiple connections

persistent HTTP

 multiple objects can be sent over single TCP connection between client, server

Non-persistent HTTP

suppose user enters URL:
www.someSchool.edu/someDepartment/home.index

(contains text, references to 10 jpeg images)

 Ia. HTTP client initiates TCP connection to HTTP server (process) at www.someSchool.edu on port 80

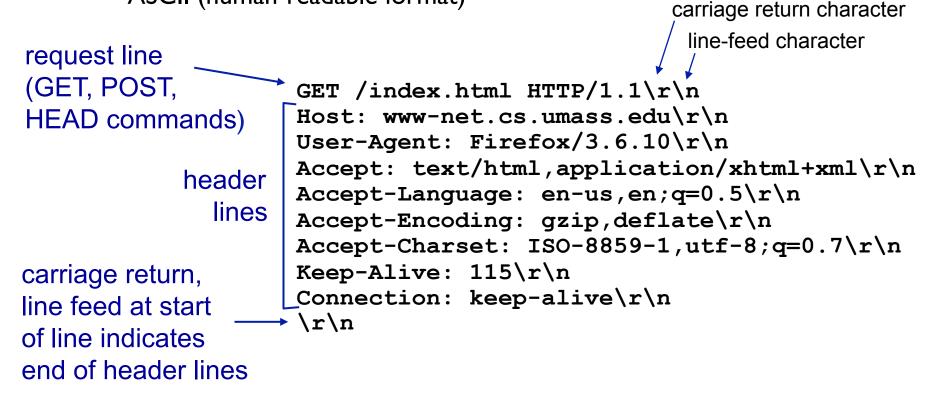
2. HTTP client sends HTTP request message (containing URL) into TCP connection socket. Message indicates that client wants object someDepartment/home.index Ib. HTTP server at host
 www.someSchool.edu waiting for
 TCP connection at port 80.
 "accepts" connection, notifying client

 3. HTTP server receives request message, forms response message containing requested object, and sends message into its socket

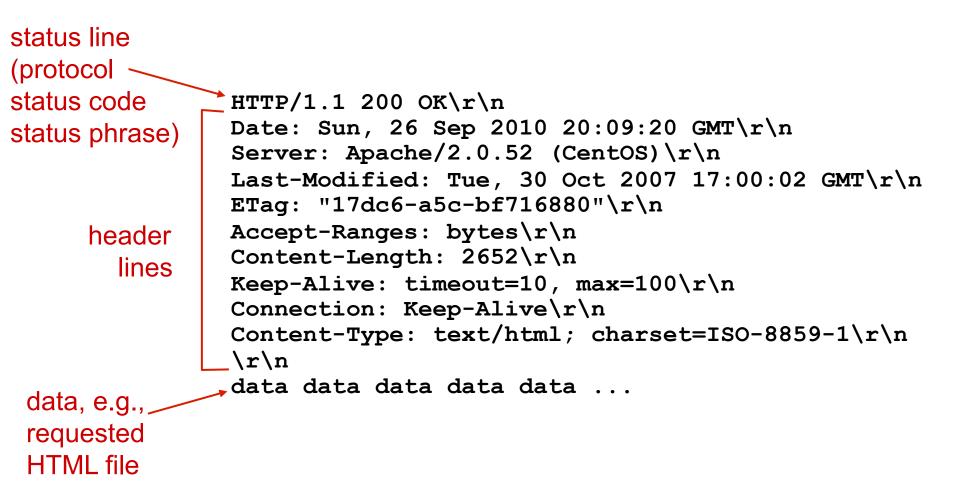
Non-persistent HTTP (cont.) 4. HTTP server closes TCP connection. 5. HTTP client receives response message containing html file, displays html. Parsing html file, finds 10 referenced jpeg objects 6. Steps 1-5 repeated for each of 10 jpeg objects

HTTP request message

- two types of HTTP messages: request, response
- HTTP request message:
 - ASCII (human-readable format)



HTTP response message



User-server state: cookies

many Web sites use cookies

four components:

- I) cookie header line of HTTP response message
- 2) cookie header line in next HTTP *request* message
- cookie file kept on user's host, managed by user's browser
- 4) back-end database at Web site

example:

- Susan always access Internet from PC
- visits specific e-commerce site for first time
- when initial HTTP requests arrives at site, site creates:
 - unique ID
 - entry in backend
 database for ID

Sample question

Consider the following HTTP request, issued using telnet:

~/> telnet www.somedomain.com 80

GET / HTTP/1.0

(empty line here)

Say the domain name `www.somedomain.com' points to address 12.13.14.15. Another domain name, `www.someother.com' also points to 12.13.14.15.

Say the owner of this web server wants to display different pages for the two different domains, even though they lead to the same IP address, and consequently to the same server. This is a very common occurrence today.

Question: Is it possible to do what the owner wants, given a request issued as above? If so, how does the web server distinguish between the two? If it is not possible, why not?

Solution:

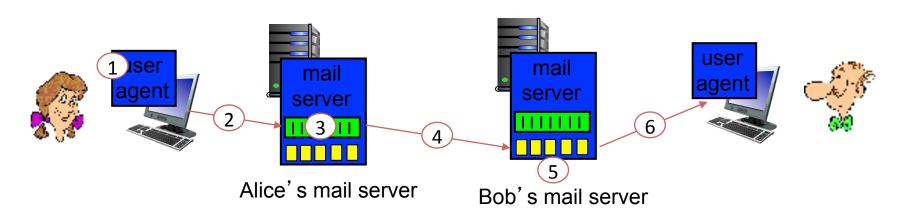
It is not possible with the request as provided above, since the domain name is not revealed to the web server at any point.

Email system in the Internet

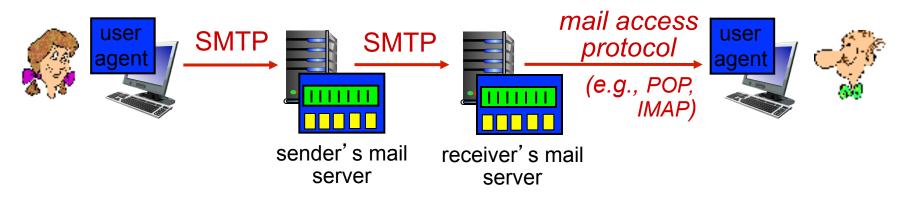
Scenario: Alice sends message to Bob

- I) Alice uses MUA to compose message "to" bob@someschool.edu
- 2) Alice's MUA sends message to her mail server; message placed in message queue
- 3) client side of SMTP opens TCP connection with Bob's mail server

- 4) SMTP client sends Alice's message over the TCP connection
- 5) Bob's mail server places the message in Bob's mailbox
- 6) Bob invokes his user agent to read message



Mail access protocols



- SMTP: delivery/storage to receiver's server
- mail access protocol: retrieval from server
 - POP: Post Office Protocol [RFC 1939]: authorization, download
 - IMAP: Internet Mail Access Protocol [RFC 1730]: more features, including manipulation of stored msgs on server
 - HTTP: gmail, Hotmail, Yahoo! Mail, etc.

Try SMTP interaction for yourself:

- dig mx gmail.com
- telnet gmail-smtp-in.l.google.com 25
- see 220 reply from server
- enter HELO, MAIL FROM, RCPT TO, DATA, QUIT commands

above lets you send email without using email client (reader)

Sample question

Overheard in the campus cafeteria: ``Sure Dr. Johnson, just email your account number to jimbob@gmail.com and I'll transfer the money." How could someone exploit this information to get some free cash? What if every outgoing email server required a username and password - would this solve the problem? Would it be better if both incoming and outgoing email servers required user credentials?

Solution

If I can find out Dr. Johnson's email address, I can easily exploit the unauthenticated SMTP protocol to create a fake email from him, with my own account number in it. I don't need an outgoing email server to deliver the email - I can connect directly to the remote server. Requiring username and password on the remote server is not feasible - every email server would need to have an account for every Internet user!

Domain Name Service (DNS)

DNS: services, structure

DNS services

- hostname to IP address translation
- host aliasing
 - canonical, alias names
- mail server aliasing
- load distribution
 - replicated Web servers: many IP addresses correspond to one name

why not centralize DNS?

- single point of failure
- traffic volume
- distant centralized database
- maintenance

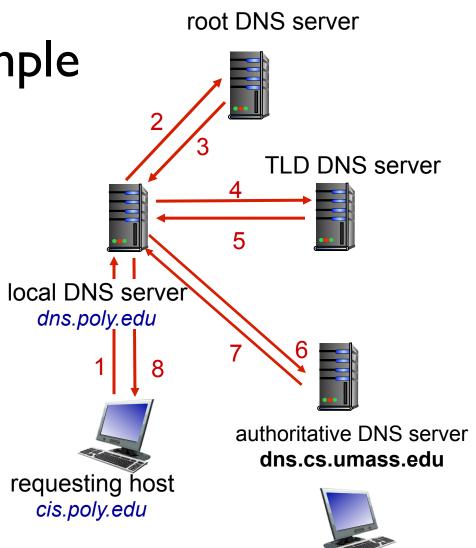
A: doesn't scale!

DNS name resolution example

 host at cis.poly.edu wants IP address for gaia.cs.umass.edu

iterated query:

- contacted server replies with name of server to contact
- "I don't know this name, but ask this server"

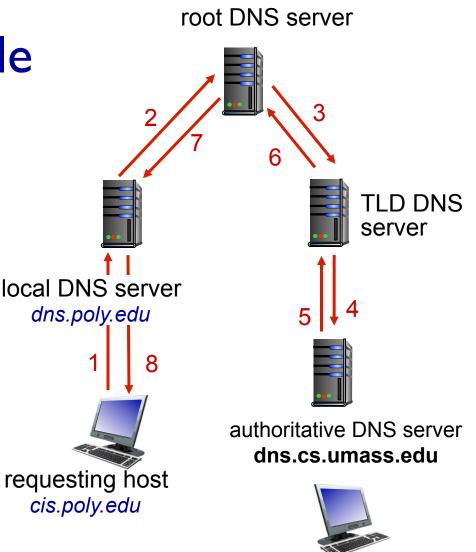


gaia.cs.umass.edu

DNS name resolution example

recursive query:

- puts burden of name resolution on contacted name server
- heavy load at upper levels of hierarchy?



gaia.cs.umass.edu

DNS records

DNS: distributed db storing resource records (RR)

RR format: (name, value, type, ttl)



- name is hostname
- value is IP address

type=NS

- name is domain (e.g., foo.com)
- value is hostname of authoritative name server for this domain

type=CNAME

- name is alias name for some "canonical" (the real) name
- www.ibm.com is really
 servereast.backup2.ibm.com
- value is canonical name

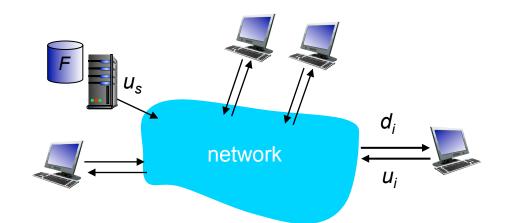
type=MX

 value is name of mailserver associated with name

Peer-to-Peer System

File distribution time: P2P

- server transmission: must upload at least one copy
 - time to send one copy: F/u_s
- client: each client must download file copy
 - min client download time: F/d_{min}



- clients: as aggregate must download NF bits
 - max upload rate (limting max download rate) is $u_s + \Sigma u_i$

time to distribute F to N clients using P2P approach

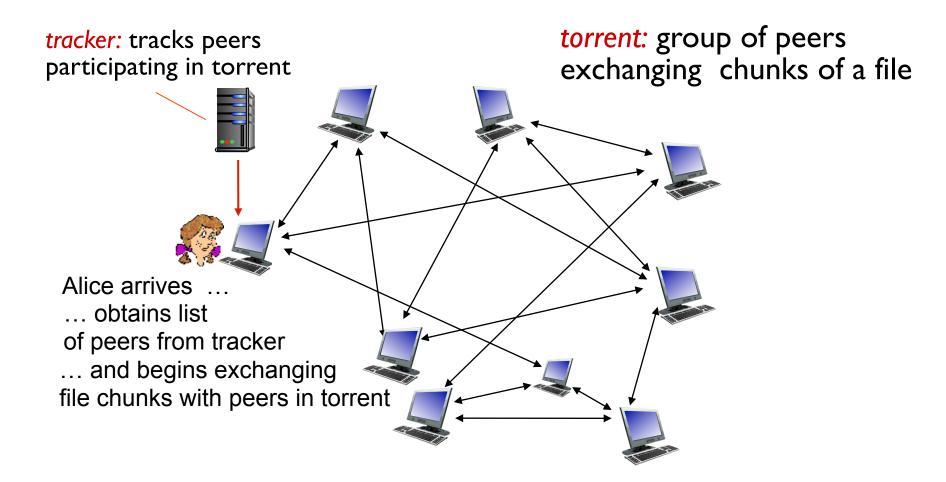
 $D_{P2P} \ge max\{F/u_{s.}, F/d_{min.}, NF/(u_s + \Sigma u_i)\}$

increases linearly in N ...

... but so does this, as each peer brings service capacity

P2P file distribution: BitTorrent

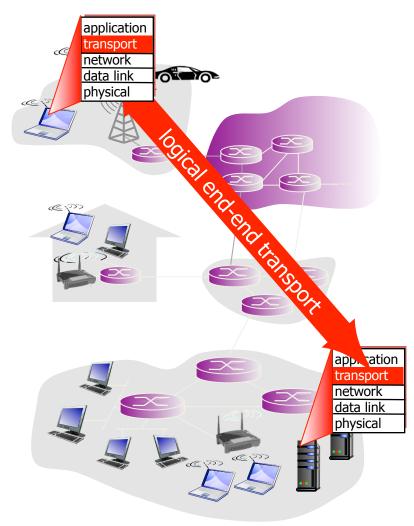
file divided into 256Kb chunks
peers in torrent send/receive file chunks



Transport Layer

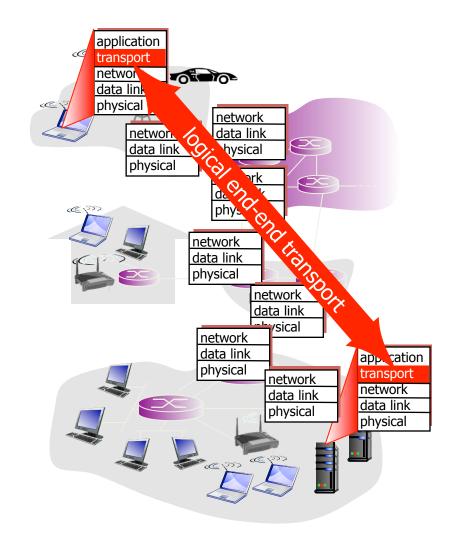
Transport services and protocols

- provide logical communication between app processes running on different hosts
- transport protocols run in end systems
 - send side: breaks app messages into segments, passes to network layer
 - rcv side: reassembles segments into messages, passes to app layer
- more than one transport protocol available to apps
 - Internet: TCP and UDP

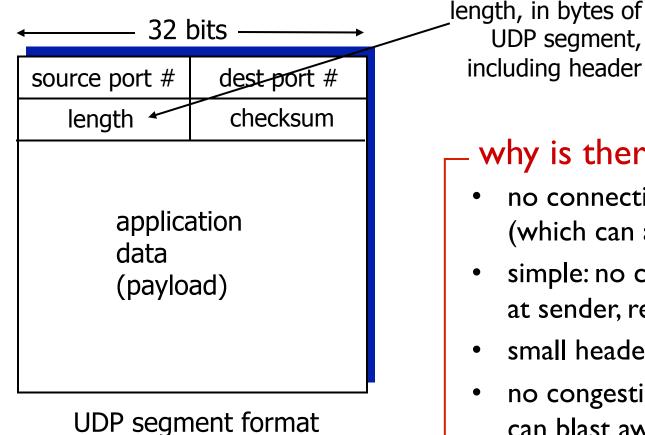


Internet transport-layer protocols

- reliable, in-order delivery (TCP)
 - congestion control
 - flow control
 - connection setup
- unreliable, unordered delivery: UDP
 - no-frills extension of "best-effort" IP
- services not available:
 - delay guarantees
 - bandwidth guarantees



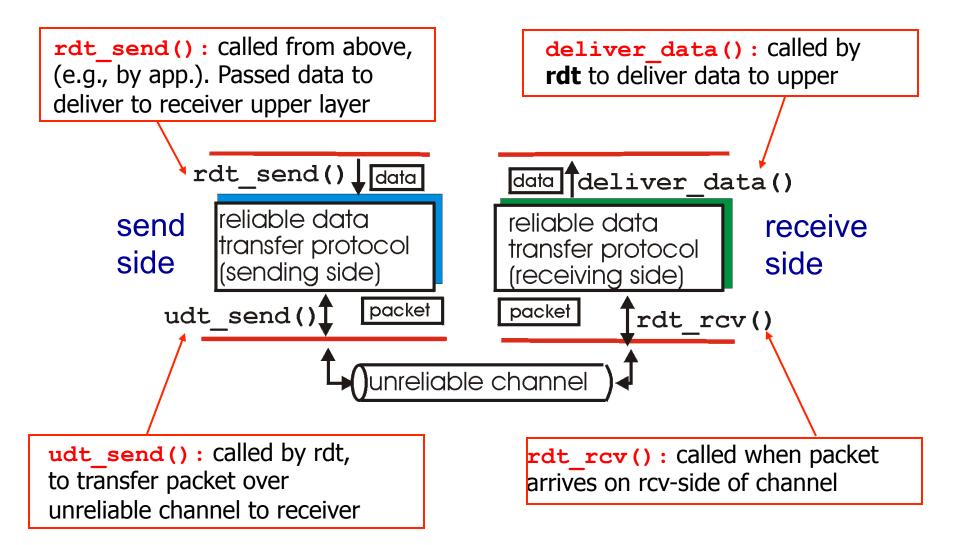
UDP: segment header



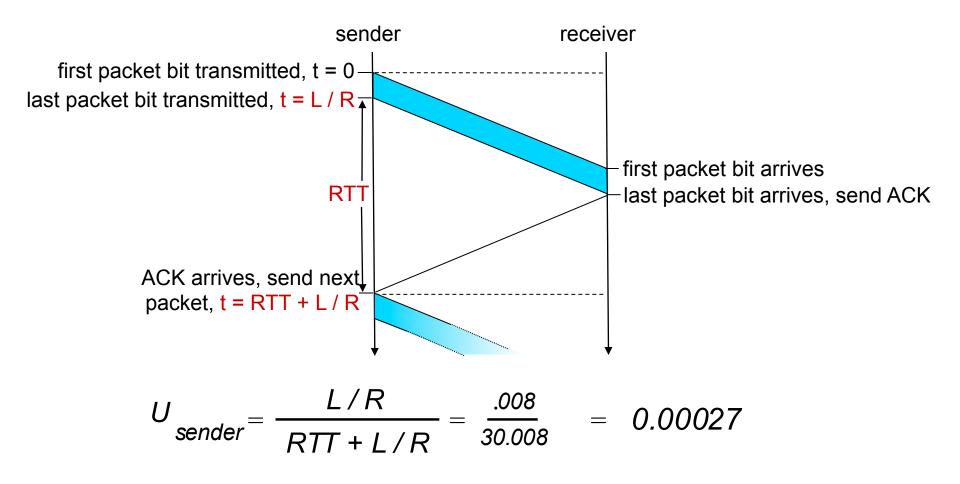
why is there a UDP? -

- no connection establishment (which can add delay)
- simple: no connection state at sender. receiver
- small header size
- no congestion control: UDP can blast away as fast as desired

Reliable data transfer: getting started



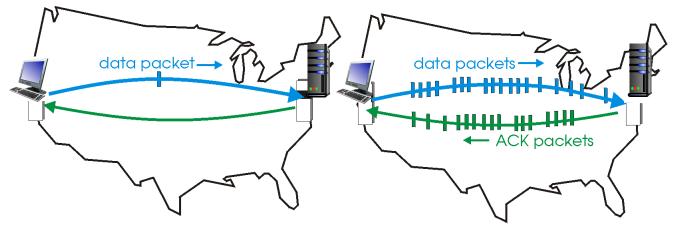
rdt3.0: stop-and-wait operation



Pipelined protocols

pipelining: sender allows multiple, "in-flight", yet-tobe-acknowledged pkts

- range of sequence numbers must be increased
- buffering at sender and/or receiver

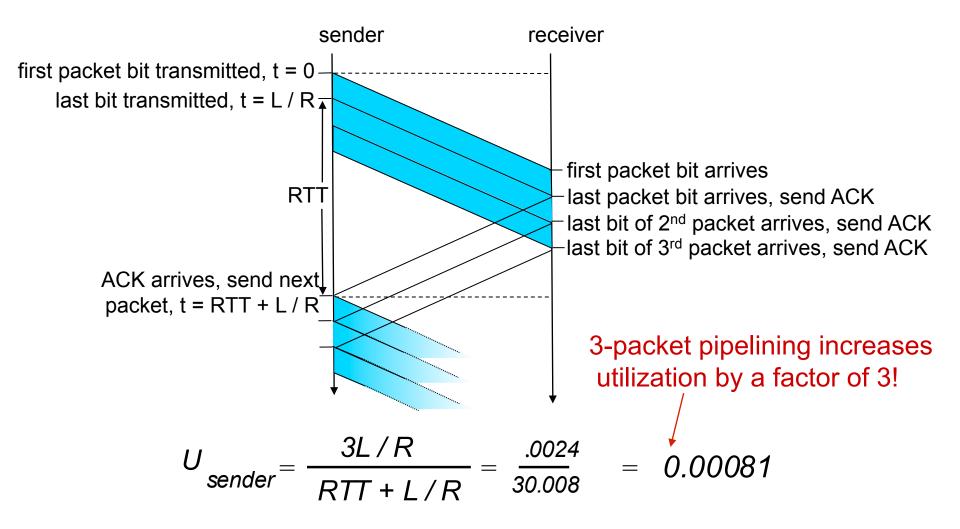


(a) a stop-and-wait protocol in operation

(b) a pipelined protocol in operation

 two generic forms of pipelined protocols: go-Back-N, selective repeat

Pipelining: increased utilization



Pipelined protocols: overview

Go-back-N:

- sender can have up to N unacked packets in pipeline
- receiver only sends cumulative ack
 - doesn't ack packet if there's a gap
- sender has timer for oldest unacked packet
 - when timer expires, retransmit *all* unacked packets

Selective Repeat:

- sender can have up to N unack' ed packets in pipeline
- rcvr sends *individual ack* for each packet

- sender maintains timer for each unacked packet
 - when timer expires, retransmit only that unacked packet

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

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

TCP round trip time, timeout

- timeout interval: EstimatedRTT plus "safety margin"
 - large variation in EstimatedRTT –> larger safety margin
- estimate SampleRTT deviation from EstimatedRTT:

```
DevRTT = (1-\beta) *DevRTT +
\beta*|SampleRTT-EstimatedRTT|
(typically, \beta = 0.25)
```

```
TimeoutInterval = EstimatedRTT + 4*DevRTT
```

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:

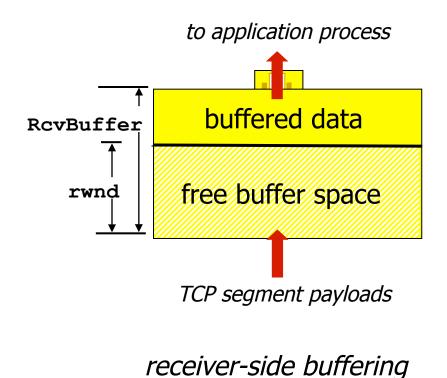
- retransmit segment that caused timeout
- restart timer
 ack rcvd:
- if ack acknowledges previously unacked segments
 - update what is known to be ACKed
 - start timer if there are still unacked segments

TCPACK generation [RFC 1122, RFC 2581]

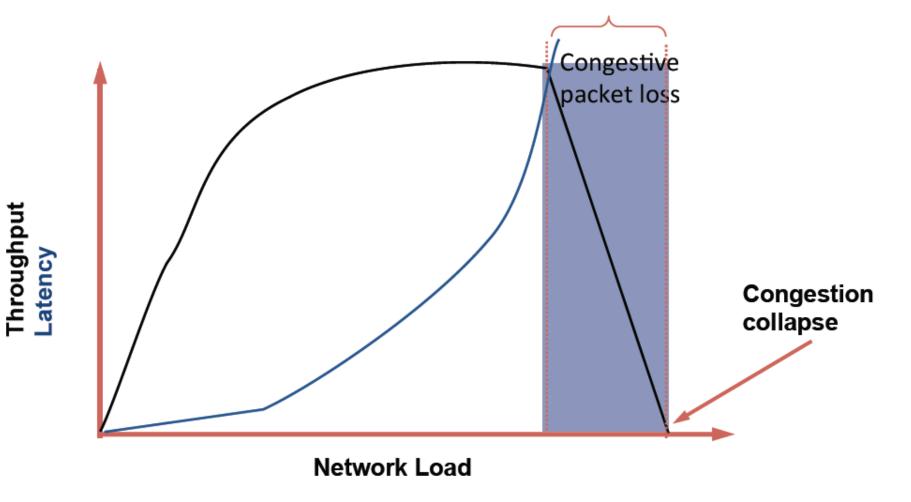
event at receiver	TCP receiver action
arrival of in-order segment with expected seq #. All data up to expected seq # already ACKed	delayed ACK. Wait up to 500ms for next segment. If no next segment, send ACK
arrival of in-order segment with expected seq #. One other segment has ACK pending	immediately send single cumulative ACK, ACKing both in-order segments
arrival of out-of-order segment higher-than-expect seq. # . Gap detected	immediately send <i>duplicate ACK</i> , indicating seq. # of next expected byte
arrival of segment that partially or completely fills gap	immediate send ACK, provided that segment starts at lower end of gap

TCP flow control

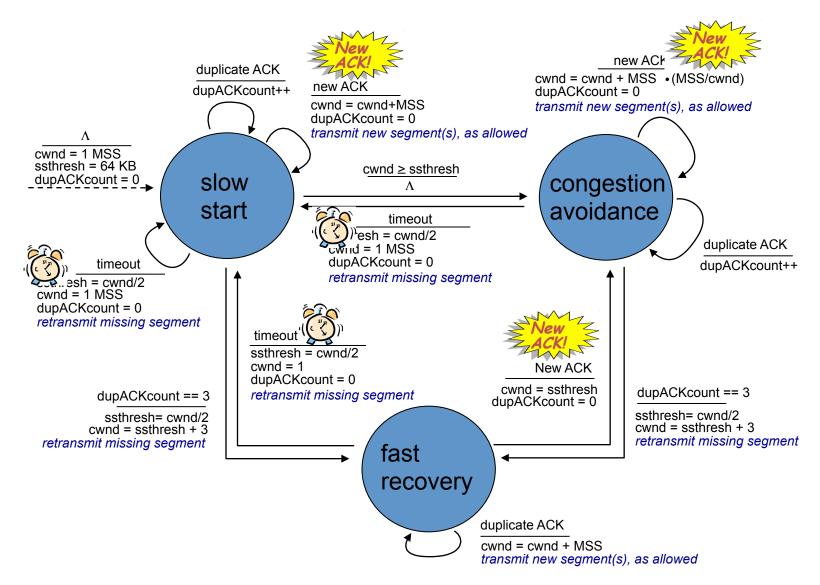
- receiver "advertises" free buffer space by including **rwnd** value in TCP header of receiver-to-sender segments
 - RcvBuffer size set via socket options (typical default is 4096 bytes)
 - many operating systems autoadjust RcvBuffer
- sender limits amount of unacked ("in-flight") data to receiver's rwnd value
- guarantees receive buffer will not overflow



Congestion collapse revisit



Summary: TCP Congestion Control



Exam: 1 PM Friday March 6, 2015 (in class - SES 238)

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 - Transport layer
 - UDP/TCP
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Next lecture – Wed, March 4

• Network Layer Intro

- Readings 4.1.1-4.2.2