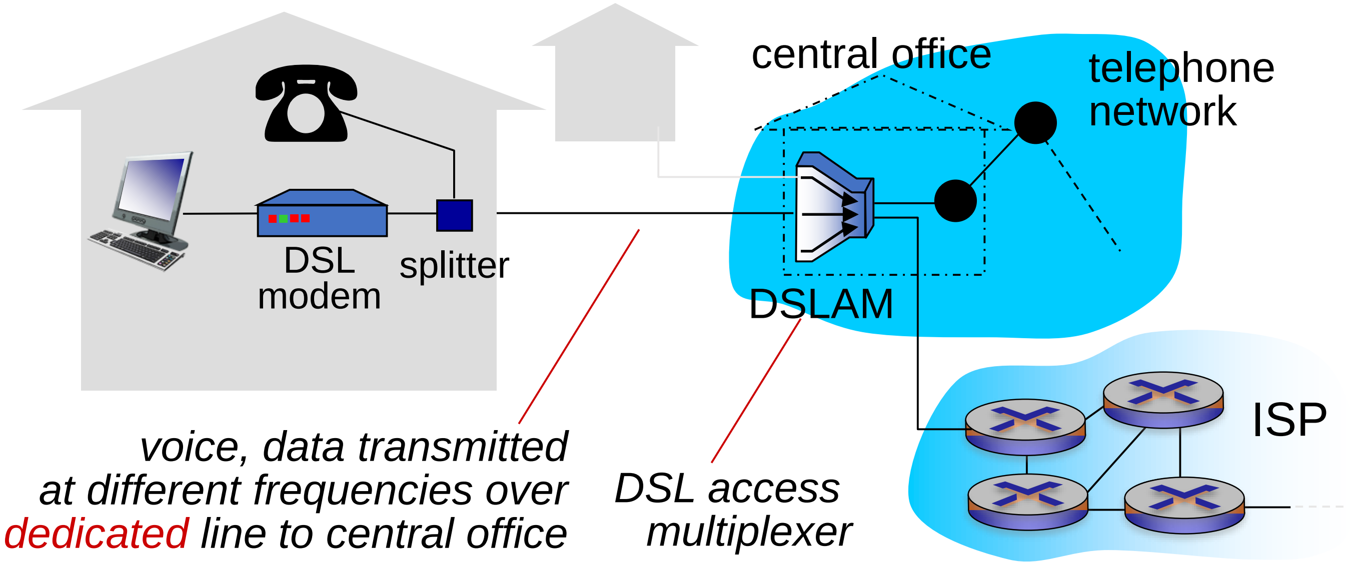
CS 3357 Fall 2019

Final Exam Review of Selected Topics

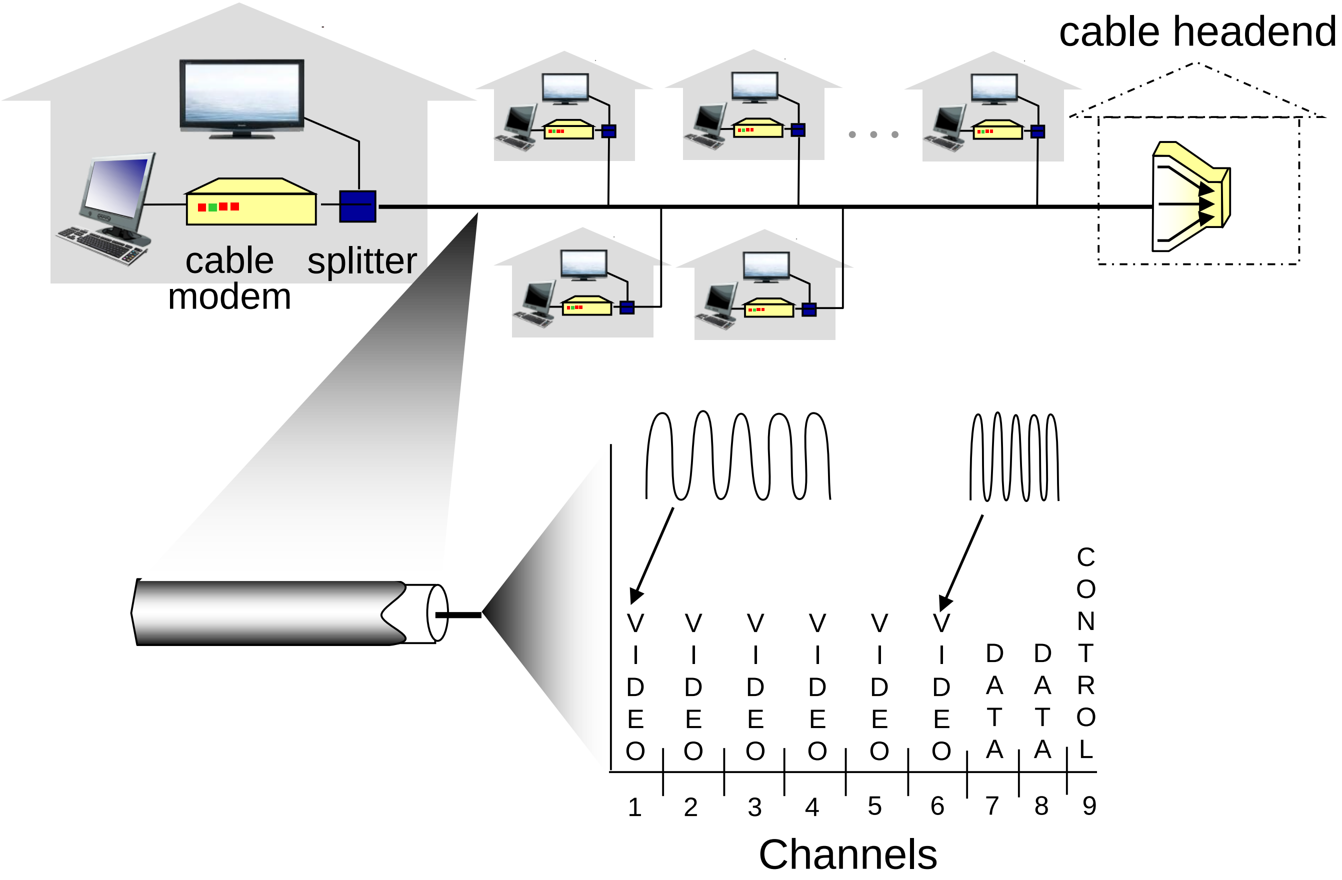
Wednesday Dec 4, 2019

Access network: digital subscriber line (DSL)



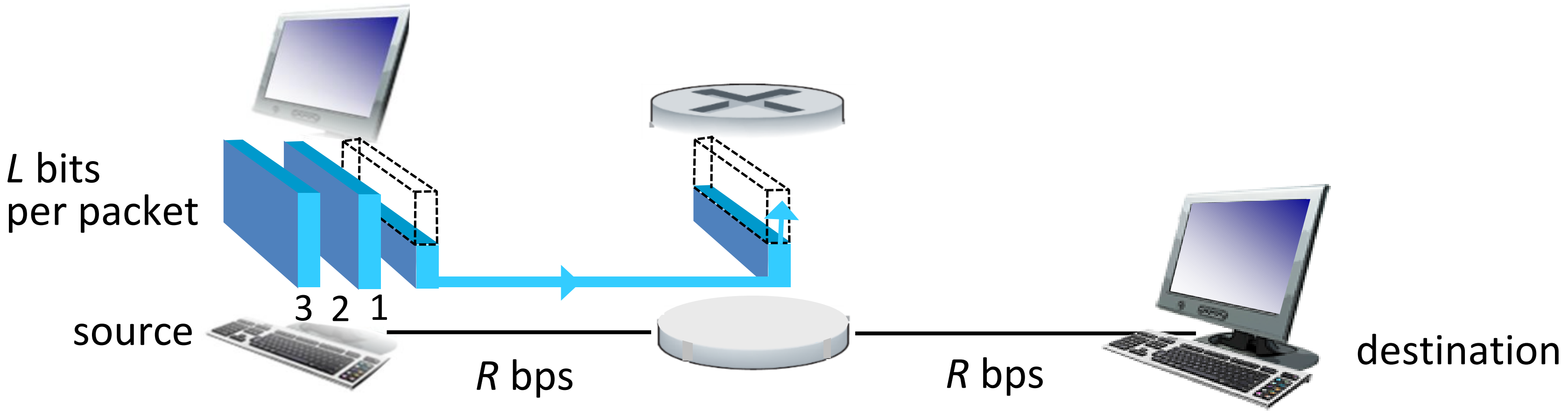
- use *existing* telephone line to central office DSLAM
  - data over DSL phone line goes to Internet
  - voice over DSL phone line goes to telephone net
- < 10 Mbps upstream transmission rate (typically < 5 Mbps)
- < 100 Mbps downstream transmission rate (typically < 15 Mbps)

Access network: cable network



frequency division multiplexing: different channels transmitted in different frequency bands

Packet-switching: store-and-forward



- takes L/R seconds to transmit (push out) L-bit packet into link at R bps
- *store and forward:* entire packet must arrive at router before it can be transmitted on next link
- end-end delay = $2L/R$ (assuming zero propagation delay)

more on delay shortly …

one-hop numerical example:

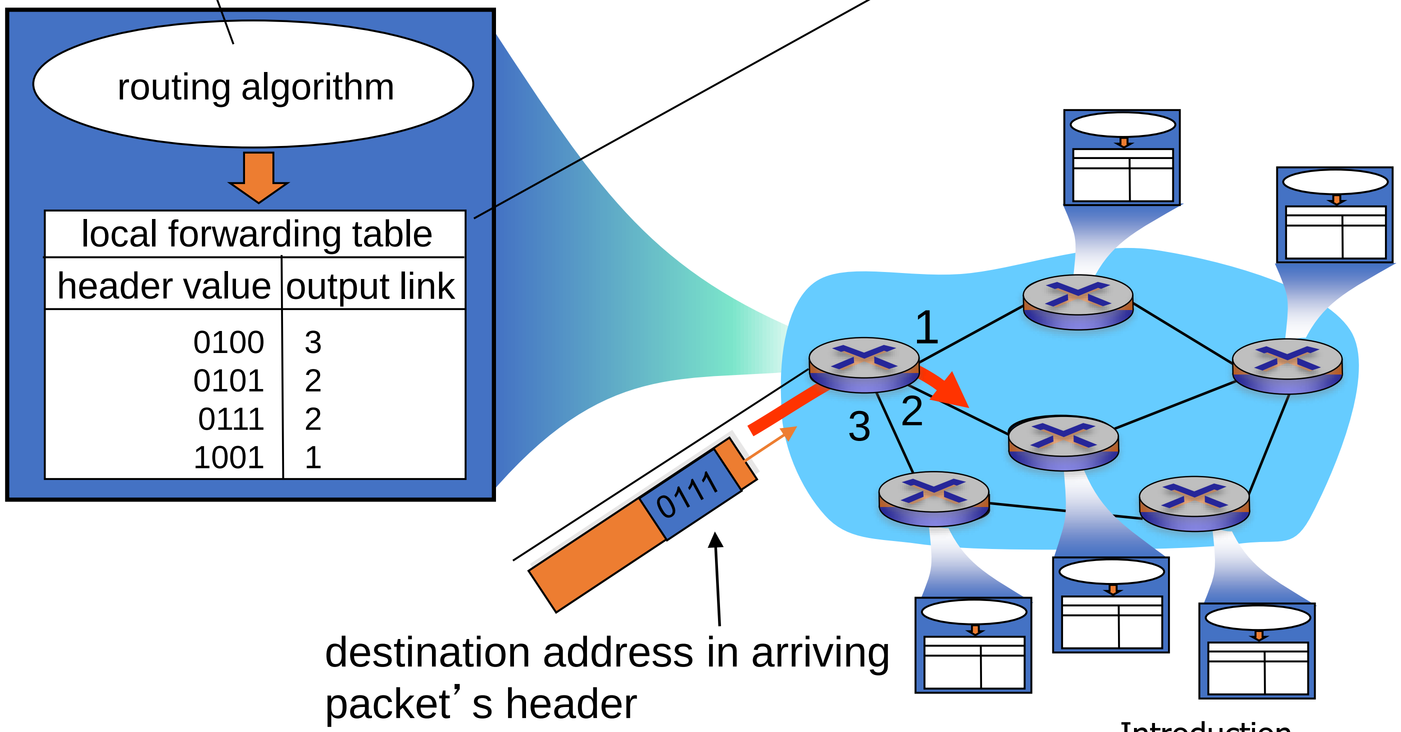
- L = 7.5 Mbits
- R = 1.5 Mbps
- one-hop transmission delay = 5 sec

Two key network-core functions

routing: determines source-destination route taken by packets

- *routing algorithms*

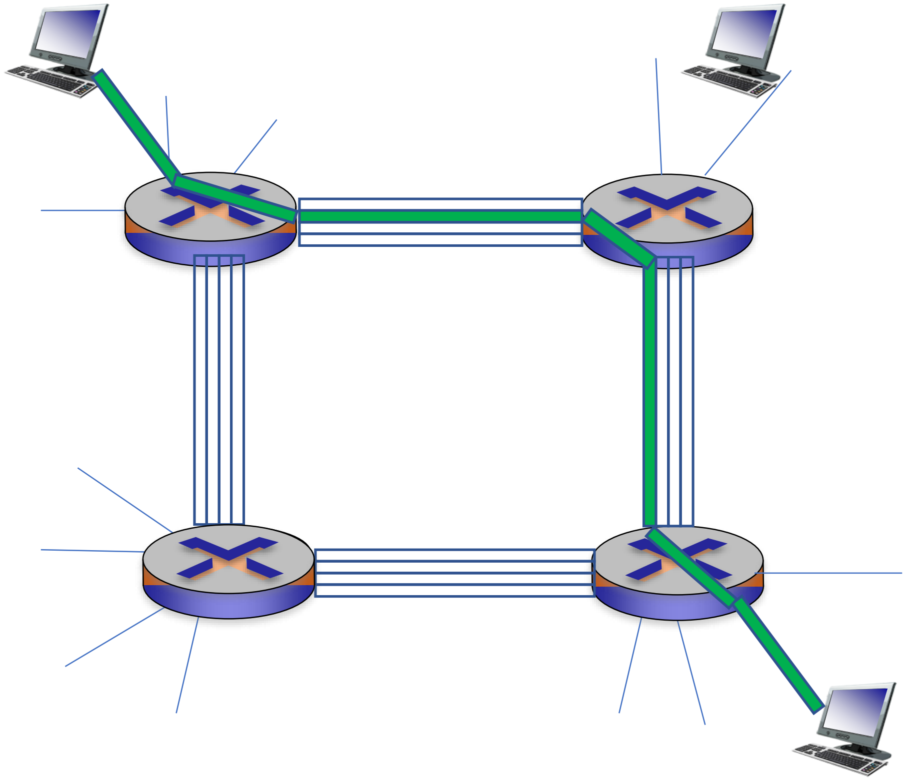
forwarding: move packets from router's input to appropriate router output



Alternative core: circuit switching

end-end resources allocated to, reserved for "call" between source & dest:

- in diagram, each link has four circuits.
  - call gets 2nd circuit in top link and 1st circuit in right link.
- dedicated resources: no sharing
  - circuit-like (guaranteed) performance
- circuit segment idle if not used by call *(no sharing)*
- commonly used in traditional telephone networks

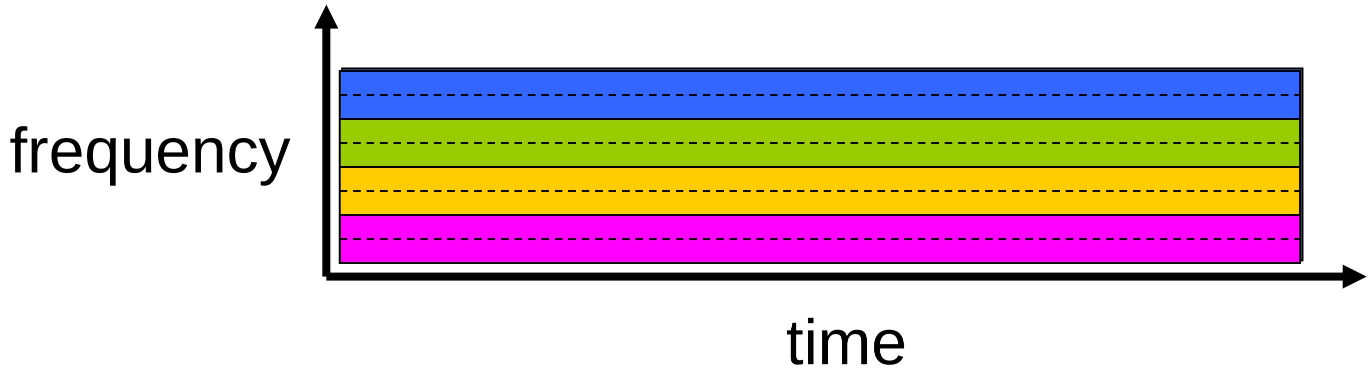


Circuit switching: FDM versus TDM

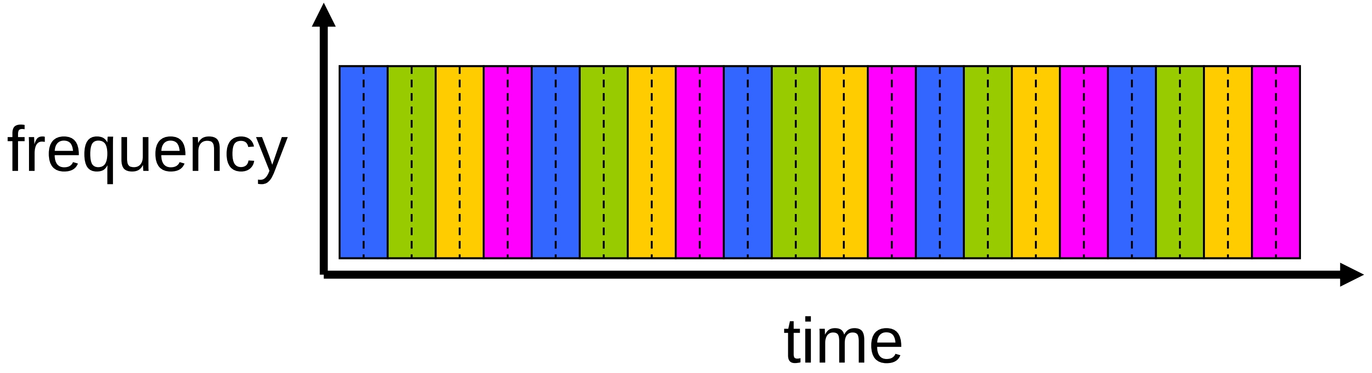
Example:

4 users

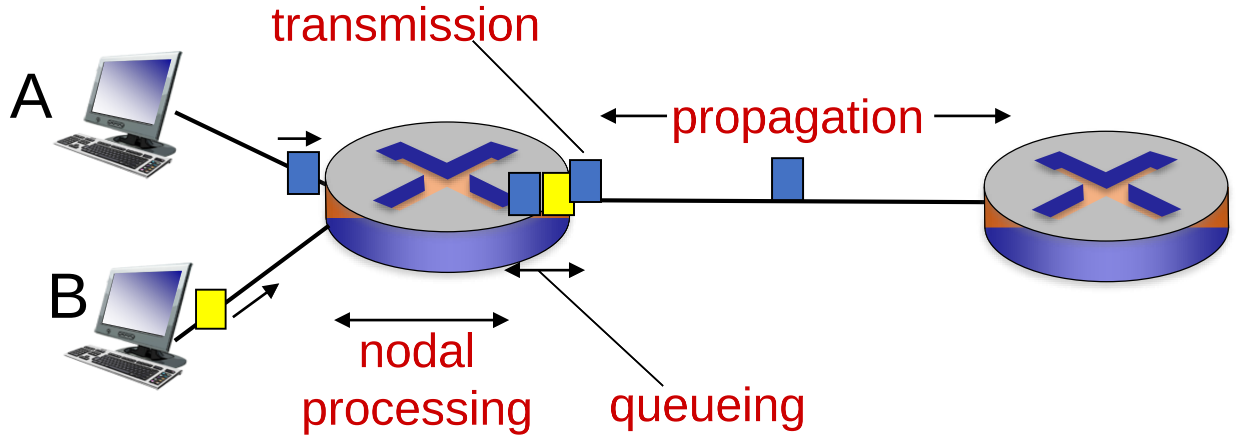
FDM



TDM



Four sources of packet delay



$$d_{nodal} = d_{proc} + d_{queue} + d_{trans} + d_{prop}$$

d_{trans}: transmission delay:

- L: packet length (bits)
- R: link *bandwidth (bps)*
- $d_{trans} = L/R$

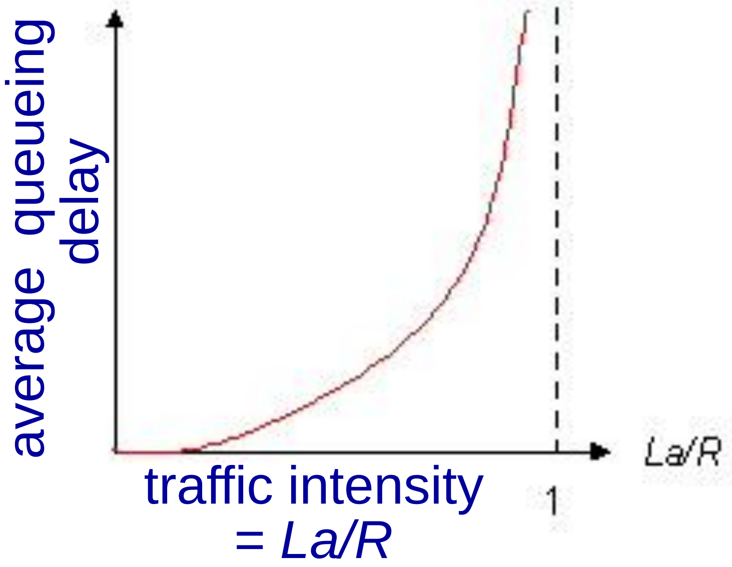
d_{trans} and d_{prop} *very* different

d_{prop}: propagation delay:

- d: length of physical link
- s: propagation speed (~2x10^8 m/sec)
- $d_{prop} = d/s$

Queueing delay (revisited)

- *R:* link bandwidth (bps)
- *L:* packet length (bits)
- a: average packet arrival rate (packets/sec)



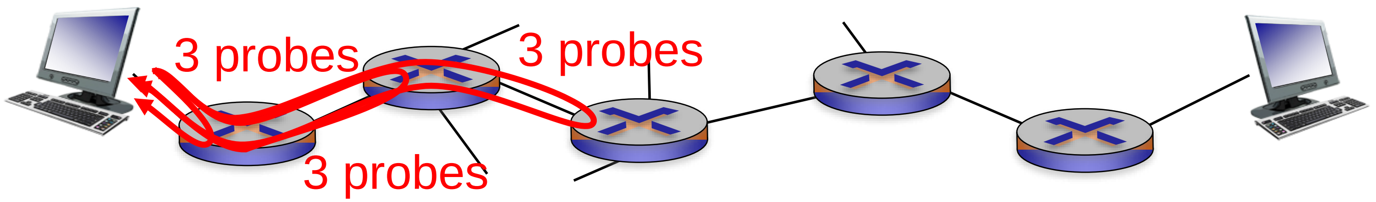
- *La/R* ~ 0: avg. queueing delay small
- *La/R* ≤ 1: avg. queueing delay varies
- *La/R* > 1: more "work" arriving than can be serviced, average delay infinite!

La/R ~ 0

La/R -> 1

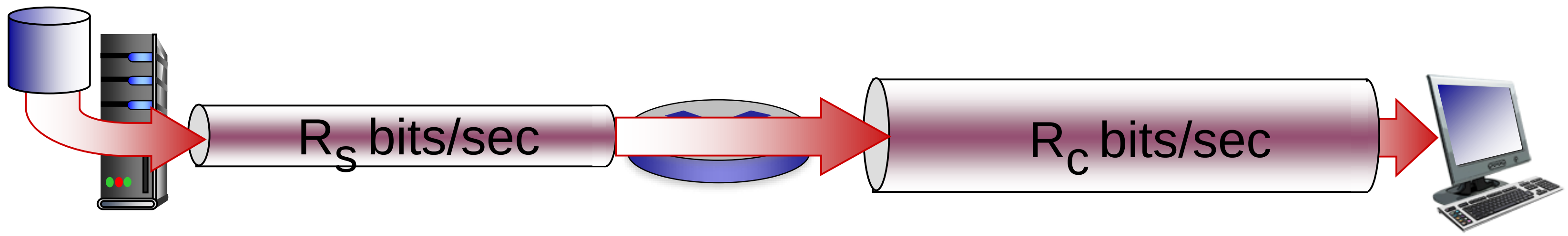
"Real" Internet delays and routes

- what do "real" Internet delay & loss look like?
- **traceroute** program: provides delay measurement from source to router along end-end Internet path towards destination. For all *i:*
  - sends three packets that will reach router *i* on path towards destination
  - router *i* will return packets to sender
  - sender times interval between transmission and reply.

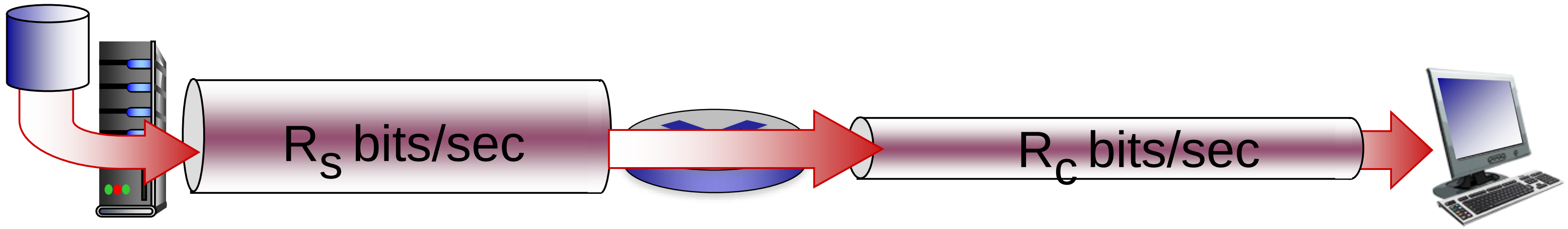


Throughput (more)

- $R_s < R_c$ What is average end-end throughput?



- $R_s > R_c$ What is average end-end throughput?

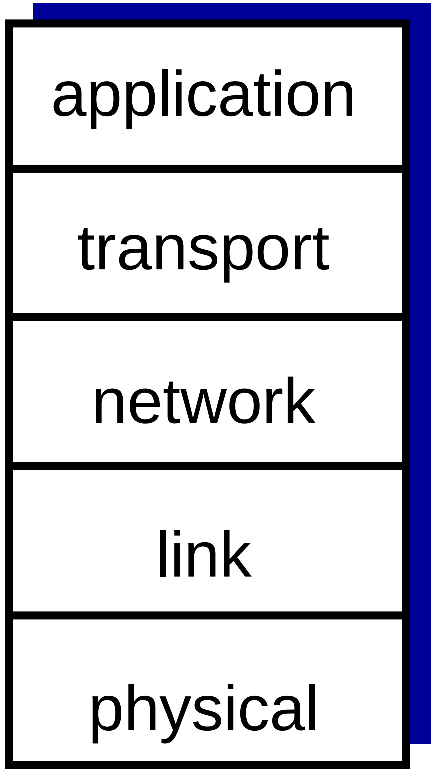


bottleneck link

link on end-end path that constrains end-end throughput

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"



Internet history

1980-1990: new protocols, a proliferation of networks

- 1983: deployment of TCP/IP
- 1982: smtp e-mail protocol defined
- 1983: DNS defined for name-to-IP-address translation
- 1985: ftp protocol defined
- 1988: TCP congestion control

- new national networks: CSnet, BITnet, NSFnet, Minitel
- 100,000 hosts connected to confederation of networks

Internet history

1990, 2000's: commercialization, the Web, new apps

- early 1990's: ARPAnet decommissioned
- 1991: NSF lifts restrictions on commercial use of NSFnet (decommissioned, 1995)
- early 1990s: Web
  - hypertext [Bush 1945, Nelson 1960's]
  - HTML, HTTP: Berners-Lee
  - 1994: Mosaic, later Netscape
  - late 1990's: commercialization of the Web

late 1990's – 2000's:

- more killer apps: instant messaging, P2P file sharing
- network security to forefront
- est. 50 million host, 100 million+ users
- backbone links running at Gbps

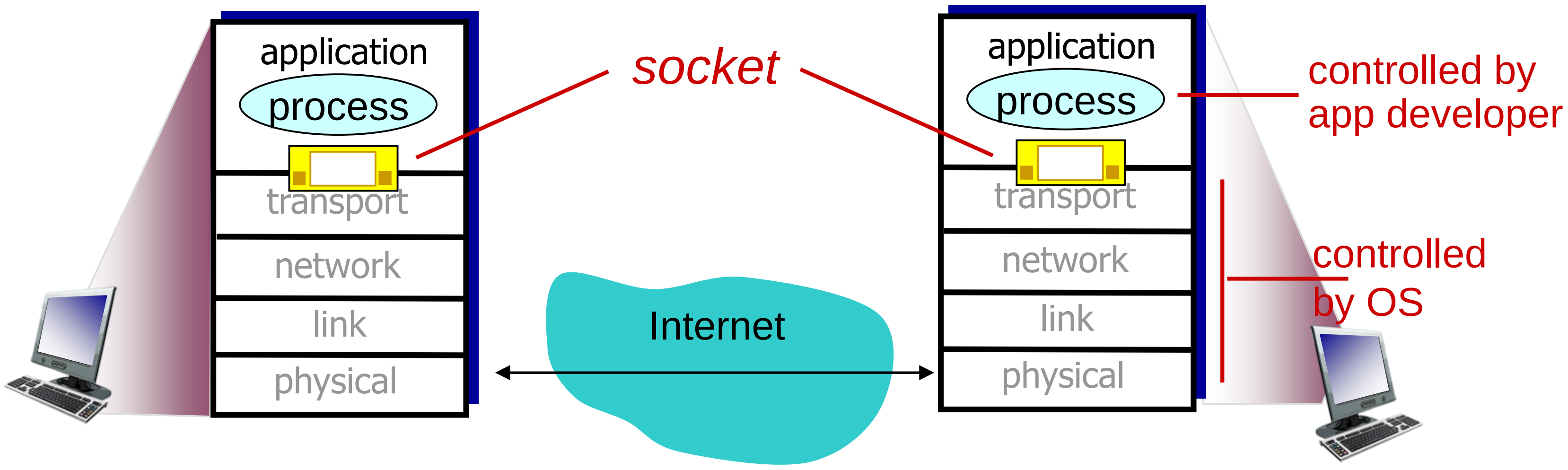
Internet history

2005-present

- ~5B devices attached to Internet (2016)
  - smartphones and tablets
- aggressive deployment of broadband access
- increasing ubiquity of high-speed wireless access
- emergence of online social networks:
  - Facebook: ~ one billion users
- service providers (Google, Microsoft) create their own networks
  - bypass Internet, providing "instantaneous" access to search, video content, email, etc.
- e-commerce, universities, enterprises running their services in "cloud" (e.g., Amazon EC2)

Sockets

- process sends/receives messages to/from its socket
- socket analogous to door
  - sending process shoves message out door
  - sending process relies on transport infrastructure on other side of door to deliver message to socket at receiving process



Addressing processes

- to receive messages, process must have *identifier*
- host device has unique 32-bit IP address
- *Q:* does IP address of host on which process runs suffice for identifying the process?
  - *A:* no, *many* processes can be running on same host

- *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 gaia.cs.umass.edu web server:
  - IP address: 128.119.245.12
  - 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?

Internet apps: application, transport protocols

application	application layer protocol	underlying transport protocol
e-mail	SMTP [RFC 2821]	TCP
remote terminal access	Telnet [RFC 854]	TCP
Web	HTTP [RFC 2616]	TCP
file transfer	FTP [RFC 959]	TCP
streaming multimedia	HTTP (e.g., YouTube), RTP [RFC 1889]	TCP or UDP
Internet telephony	SIP, RTP, proprietary (e.g., Skype)	TCP or UDP

Securing TCP

TCP & UDP

- no encryption
- cleartext passwds sent into socket traverse Internet in cleartext

SSL

- provides encrypted TCP connection
- data integrity
- end-point authentication

SSL is at app layer

- apps use SSL libraries, that "talk" to TCP

SSL socket API

- cleartext passwords sent into socket traverse Internet encrypted
- see Chapter 8

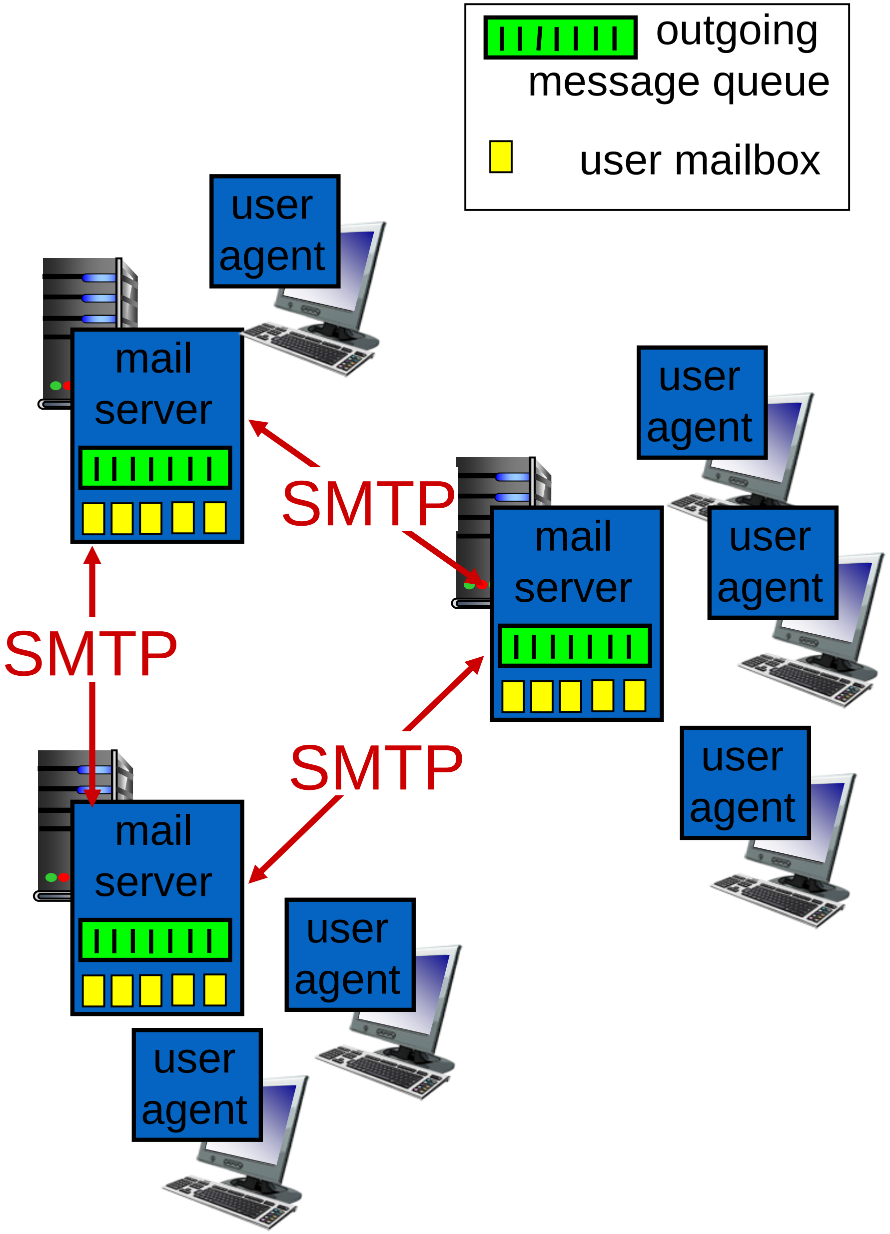
Electronic mail

Three major components:

- user agents
- mail servers
- simple mail transfer protocol: SMTP

User Agent

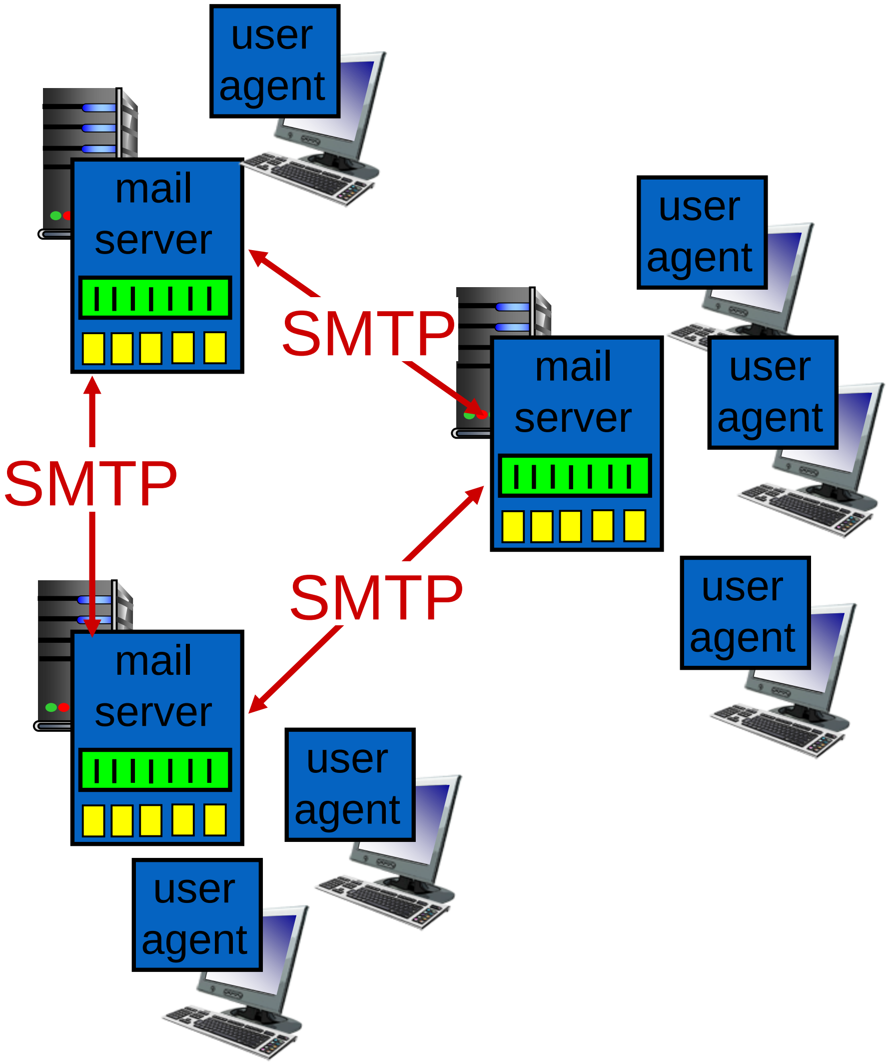
- a.k.a. "mail reader"
- composing, editing, reading mail messages
- e.g., Outlook, Thunderbird, iPhone mail client
- outgoing, incoming messages stored on server



Electronic mail: mail servers

mail servers:

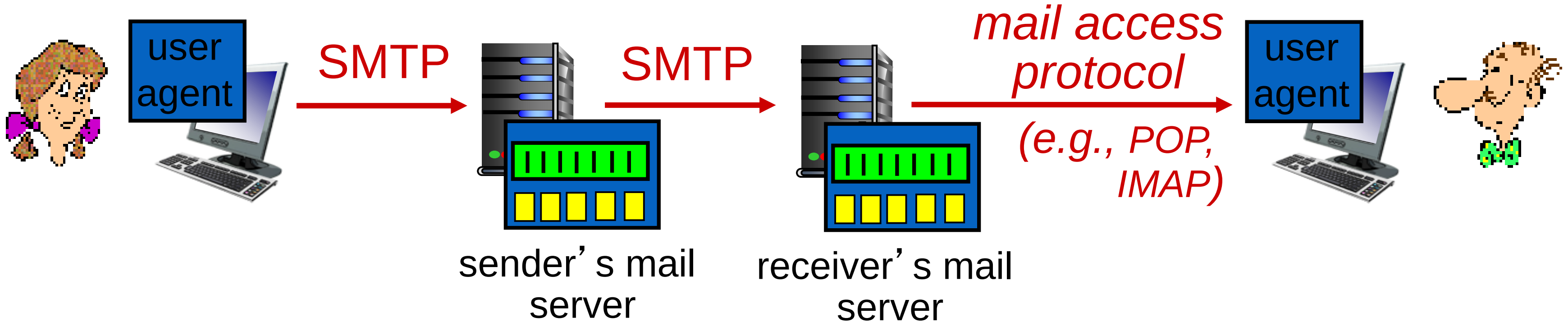
- *mailbox* contains incoming messages for user
- *message queue* of outgoing (to be sent) mail messages
- *SMTP protocol* between mail servers to send email messages
  - client: sending mail server
  - "server": receiving mail server



Electronic Mail: SMTP [RFC 2821]

- uses TCP to reliably transfer email message from client to server, port 25
- direct transfer: sending server to receiving server
- three phases of transfer
  - handshaking (greeting)
  - transfer of messages
  - closure
- command/response interaction (like HTTP)
  - commands: ASCII text
  - response: status code and phrase
- messages must be in 7-bit ASCI

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 messages on server
  - HTTP: gmail, Hotmail, Yahoo! Mail, etc.

DNS: domain name system

people: many identifiers:

- SSN, name, passport #

Internet hosts, routers:

- IP address (32 bit) - used for addressing datagrams
- "name", e.g., www.yahoo.com - used by humans

Q: how to map between IP address and name, and vice versa ?

Domain Name System:

- *distributed database* implemented in hierarchy of many *name servers*
- *application-layer protocol:* hosts, name servers communicate to *resolve* names (address/name translation)
  - note: core Internet function, implemented as application-layer protocol
  - complexity at network's "edge"

DNS: services, structure

DNS services

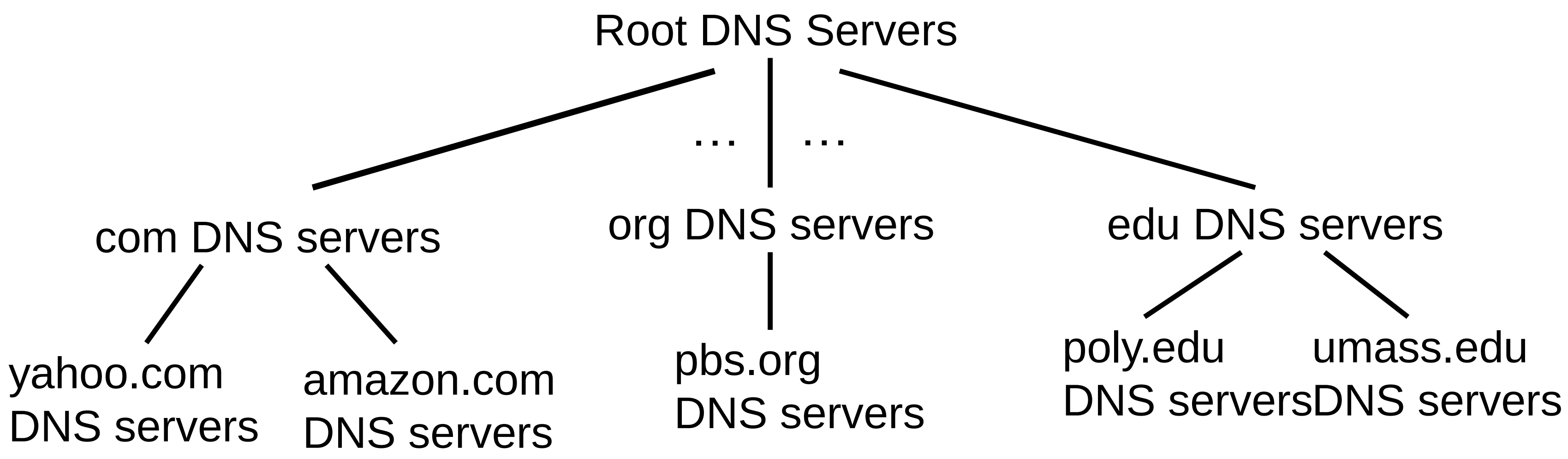
- hostname to IP address translation
- host aliasing
  - canonical, alias names
- mail server aliasing
- load distribution
  - replicated Web servers: many IP addresses correspond to one name

why not centralize DNS?

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

A: *doesn't scale!*

DNS: a distributed, hierarchical database

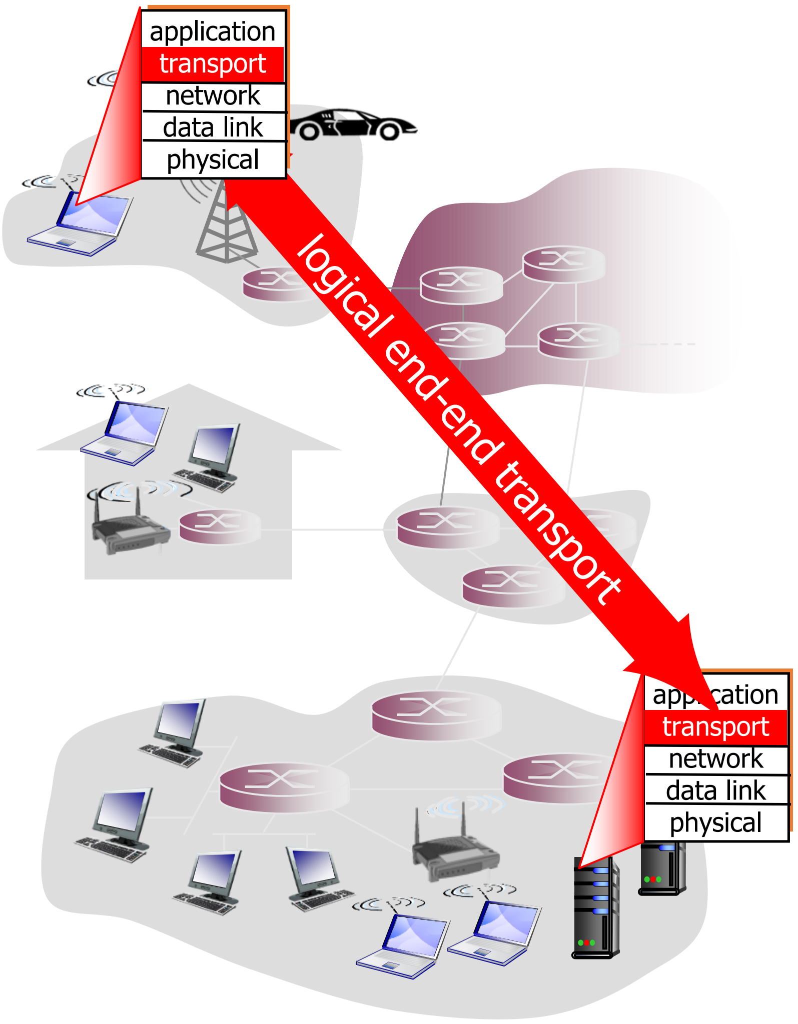


client wants IP for www.amazon.com; 1st approximation:

- client queries root server to find com DNS server
- client queries .com DNS server to get amazon.com DNS server
- client queries amazon.com DNS server to get IP address for www.amazon.com

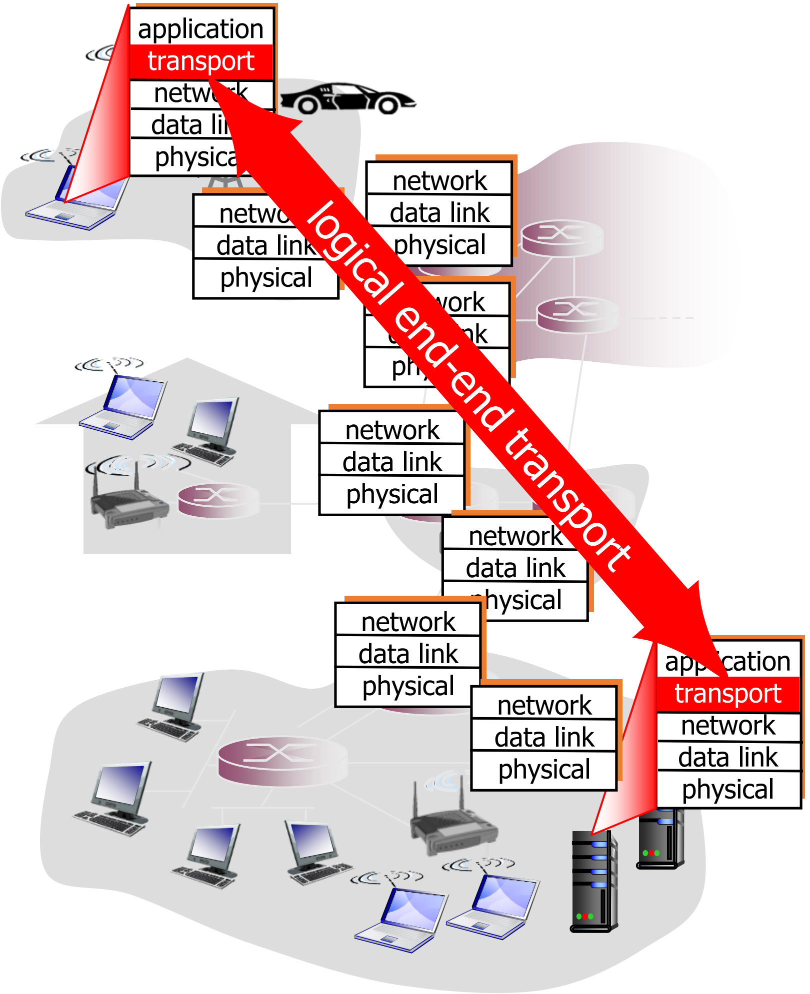
Transport services and protocols

- 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

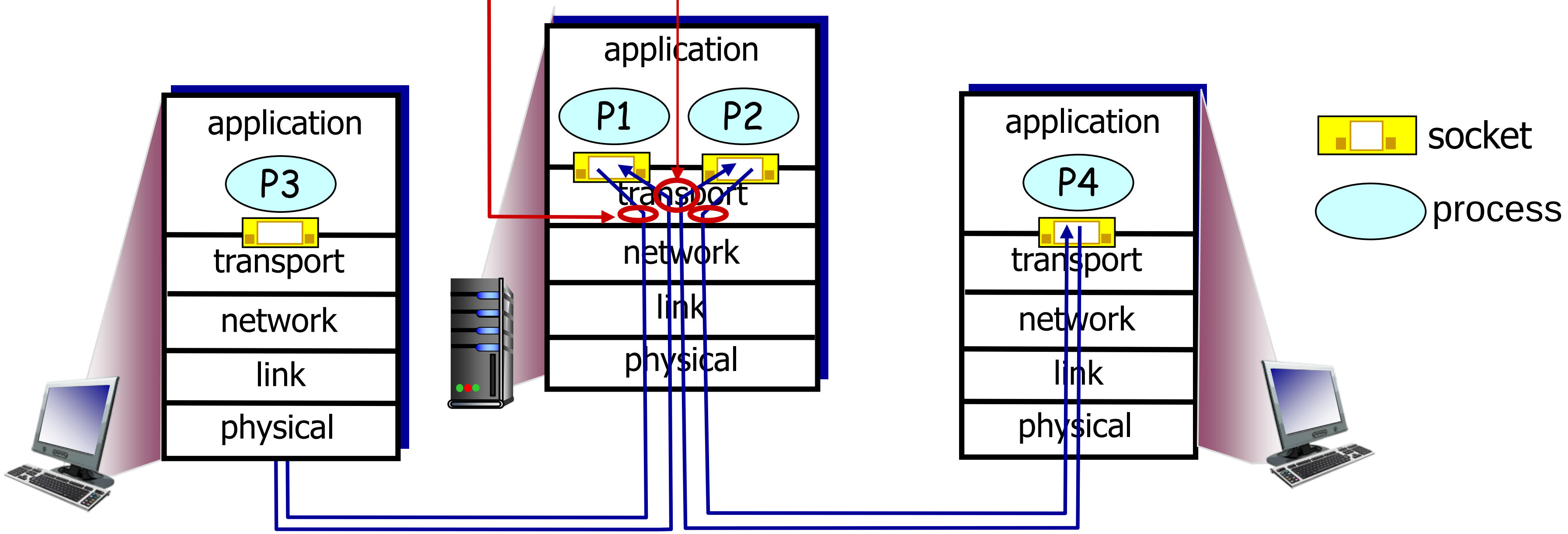
- reliable, in-order delivery (TCP)
  - congestion control
  - flow control
  - connection setup
- unreliable, unordered delivery: UDP
  - no-frills extension of "best-effort" IP
- services not available:
  - delay guarantees
  - bandwidth guarantees



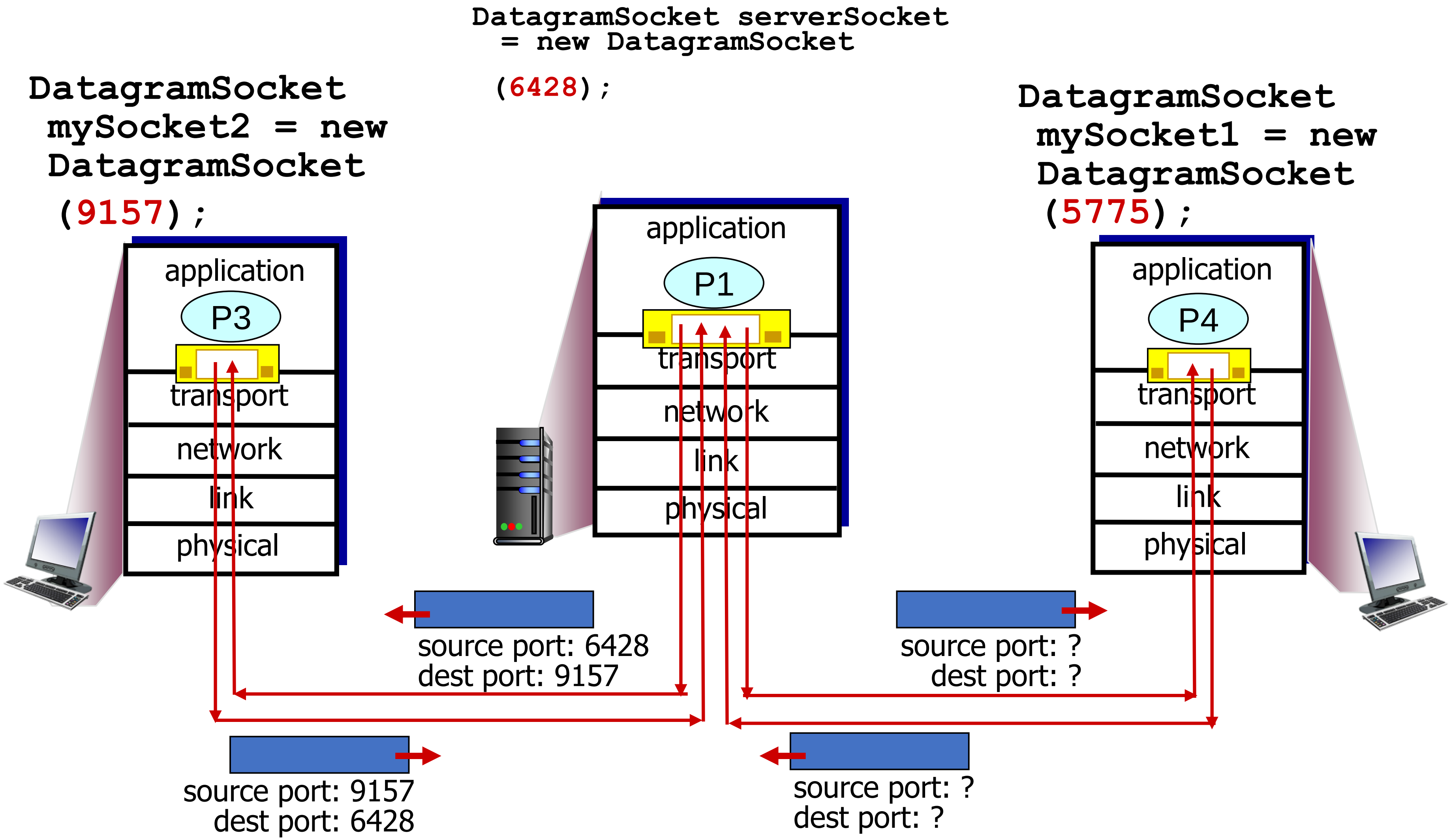
Multiplexing/demultiplexing

multiplexing at sender:
handle data from multiple sockets, add transport header (later used for demultiplexing)

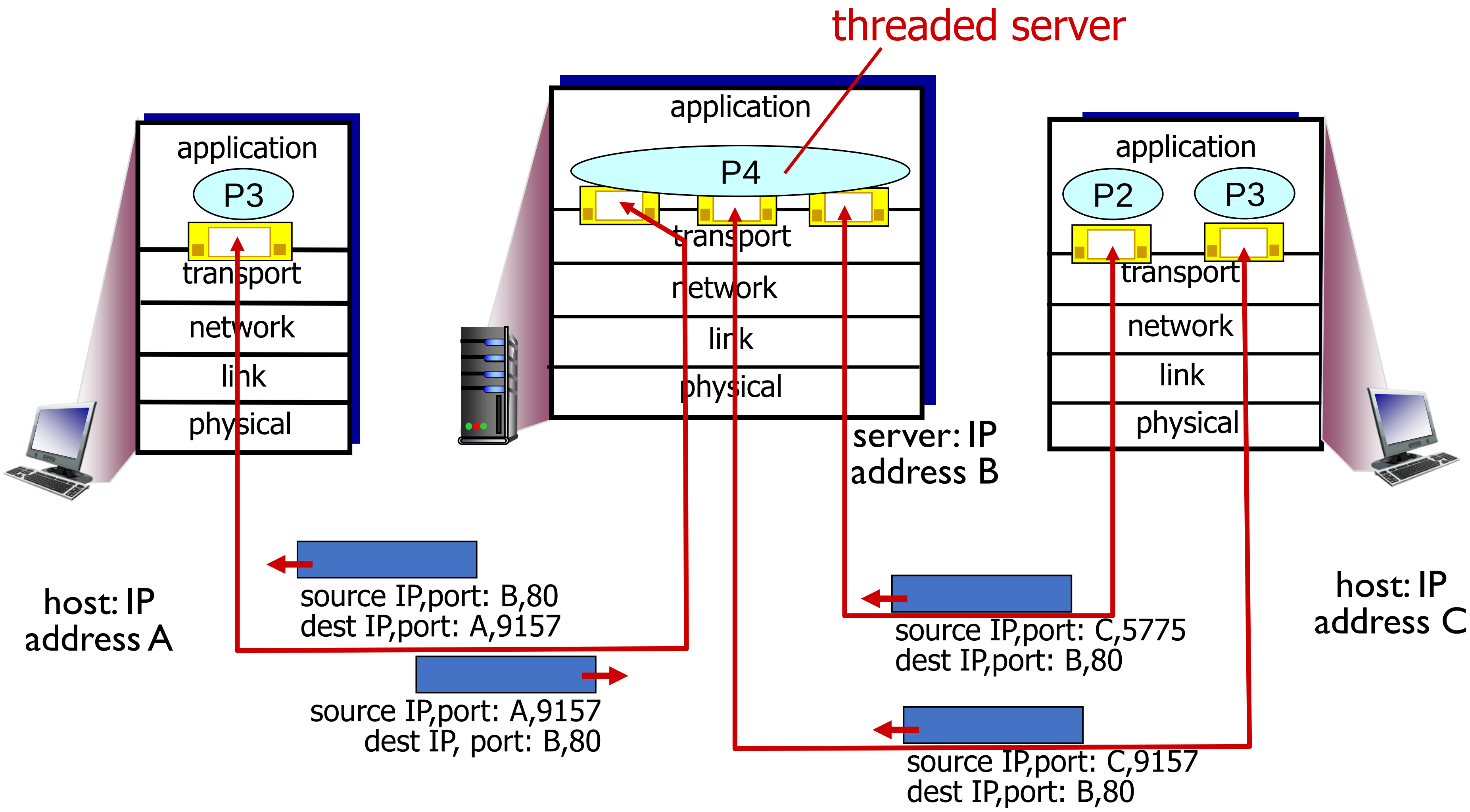
demultiplexing at receiver:
use header info to deliver received segments to correct socket



Connectionless demux: example



Connection-oriented demux: example



UDP: User Datagram Protocol [RFC 768]

- "no frills," "bare bones" Internet transport protocol
- "best effort" service, UDP segments may be:
  - lost
  - delivered out-of-order to app
- *connectionless:*
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently of others

- UDP use:
  - streaming multimedia apps (loss tolerant, rate sensitive)
  - DNS
  - SNMP
- reliable transfer over UDP:
  - add reliability at application layer
  - application-specific error recovery!

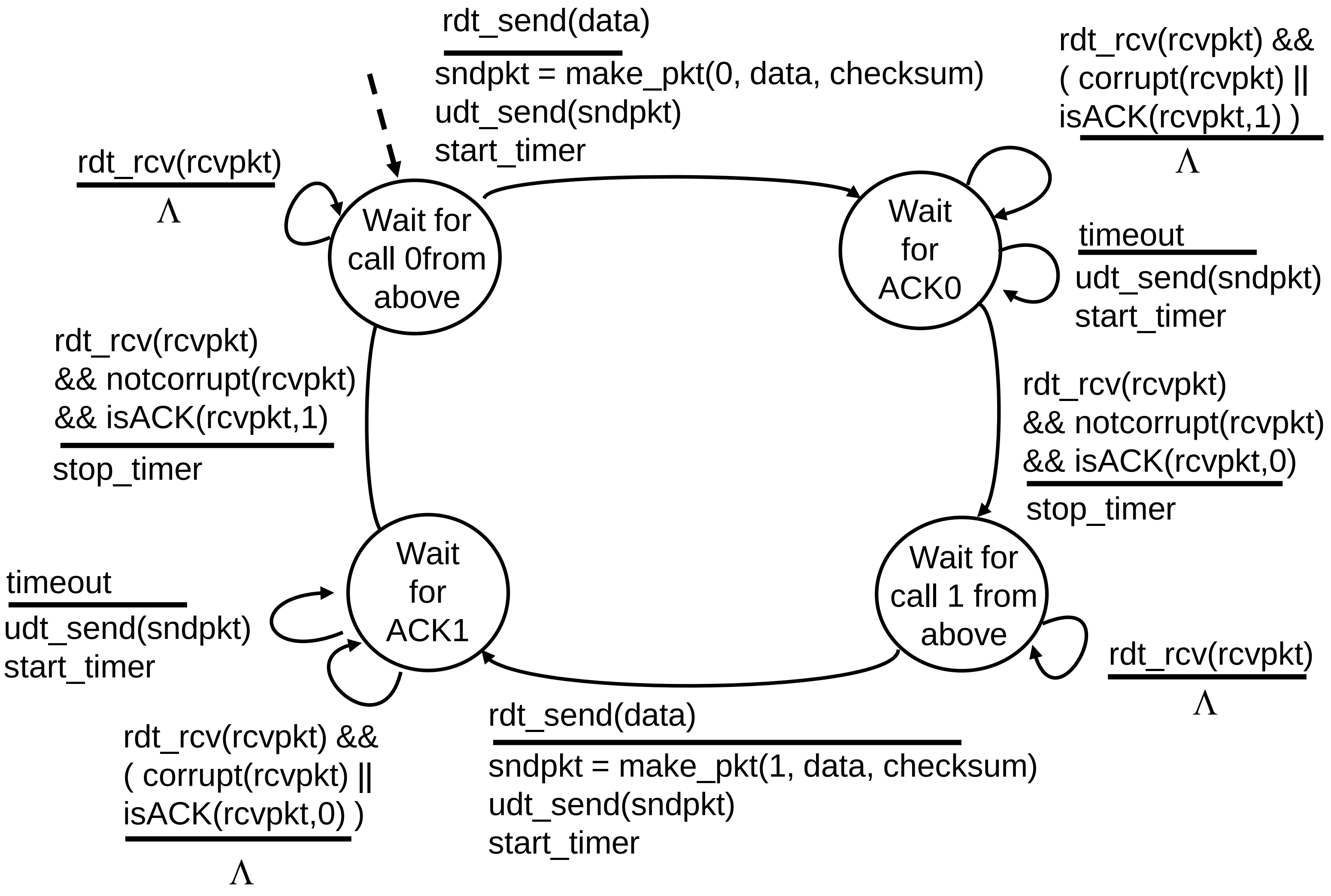
Internet checksum: example

example: add two 16-bit integers

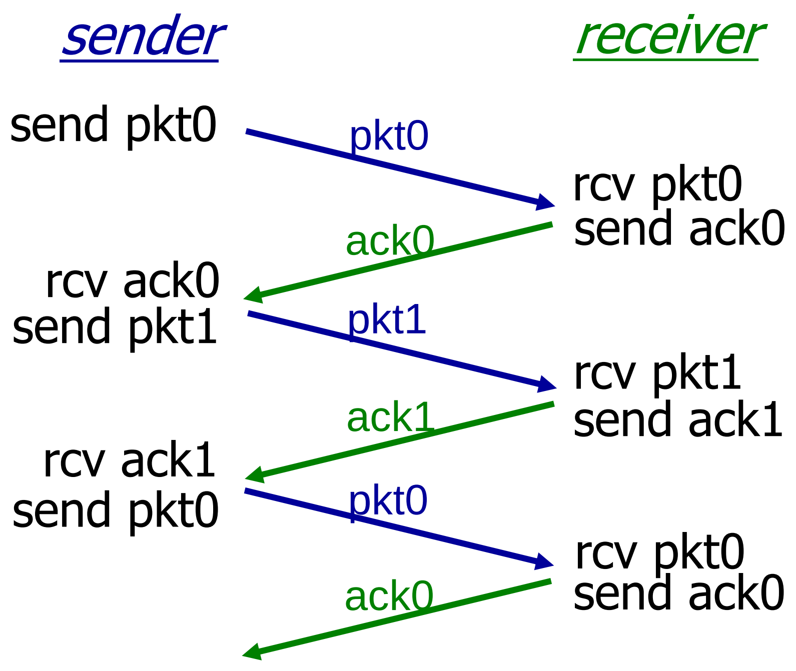
		1	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0
		1	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1
wraparound	1	1	0	1	1	1	0	1	1	1	0	1	1	1	0	1	1
sum		1	0	1	1	1	0	1	1	1	0	1	1	1	1	0	0
checksum		0	1	0	0	0	1	0	0	0	1	0	0	0	0	1	1

Note: when adding numbers, a carryout from the most significant bit needs to be added to the result

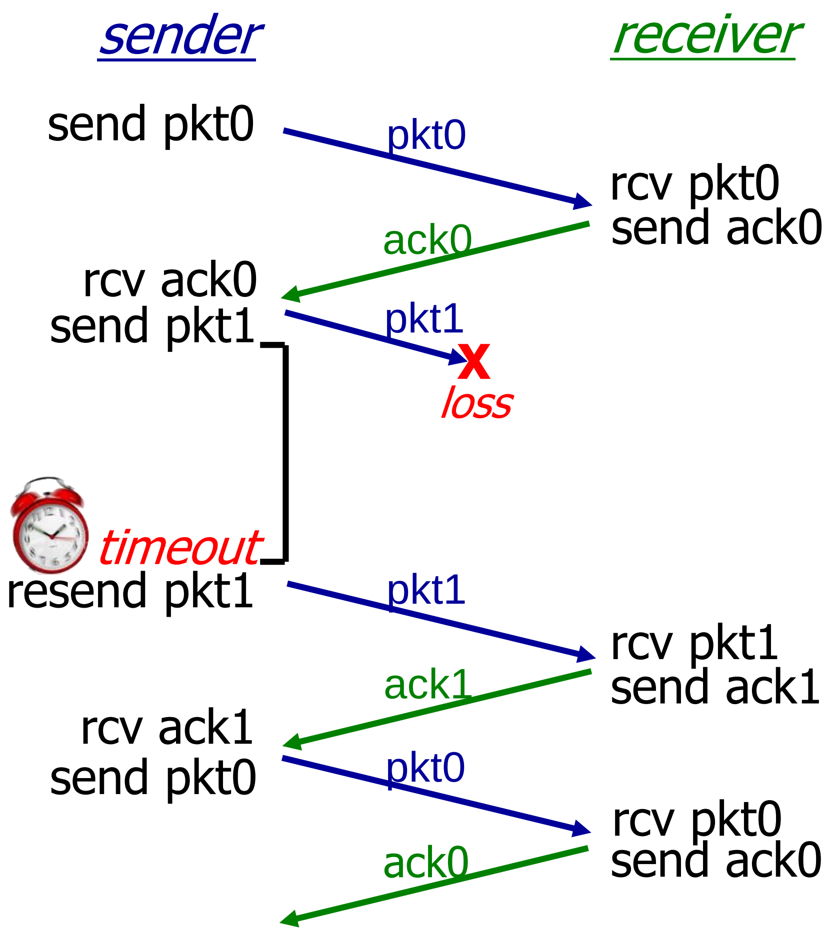
rdt3.0 sender



rdt3.0 in action

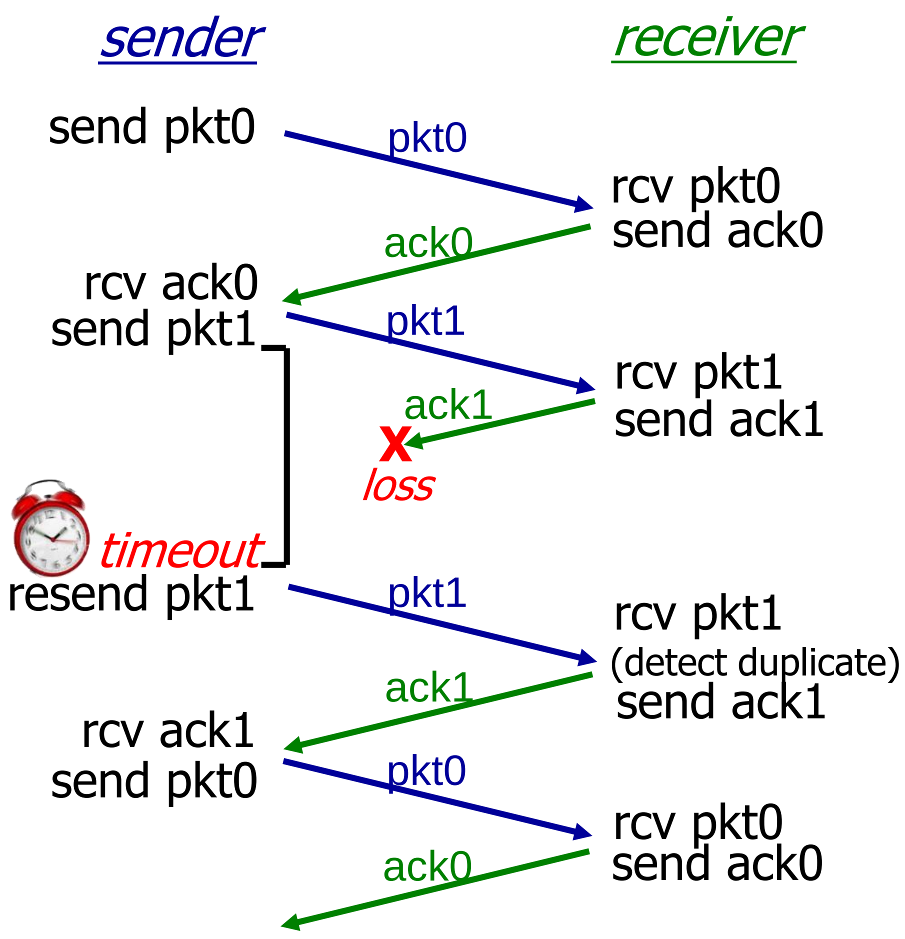


(a) no loss

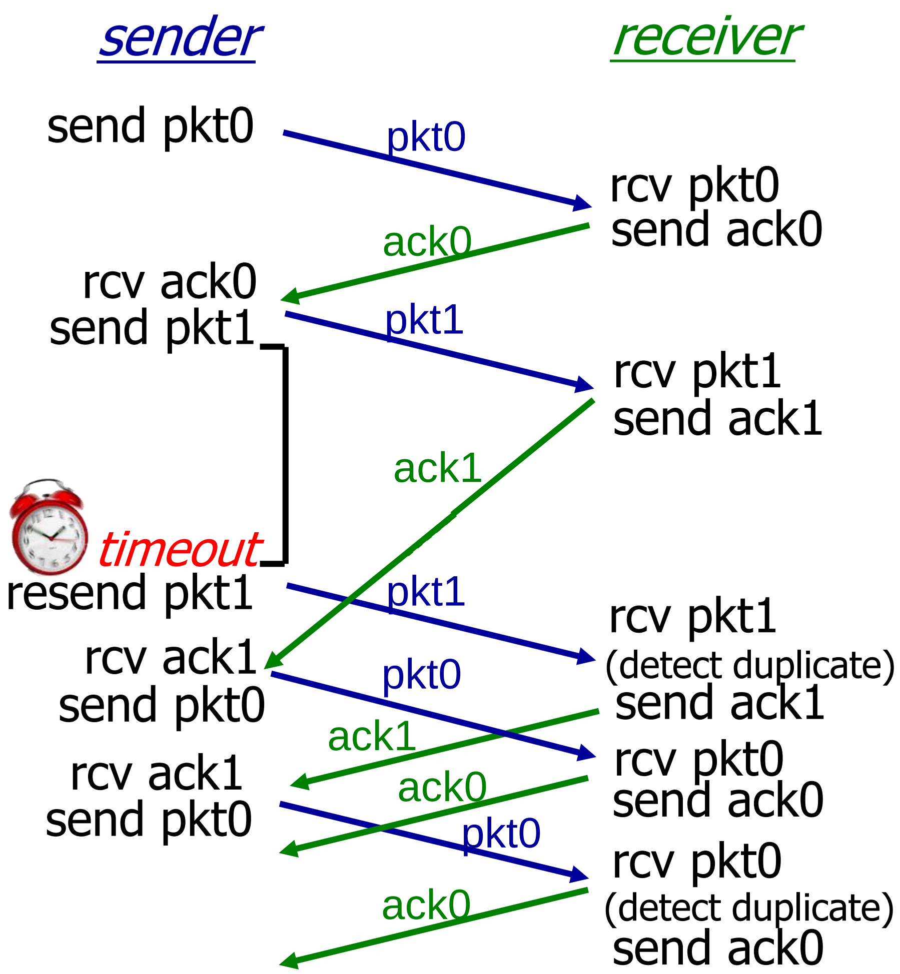


(b) packet loss

rdt3.0 in action



(c) ACK loss



(d) premature timeout/ delayed ACK

TCP seq. numbers, ACKs

sequence numbers:

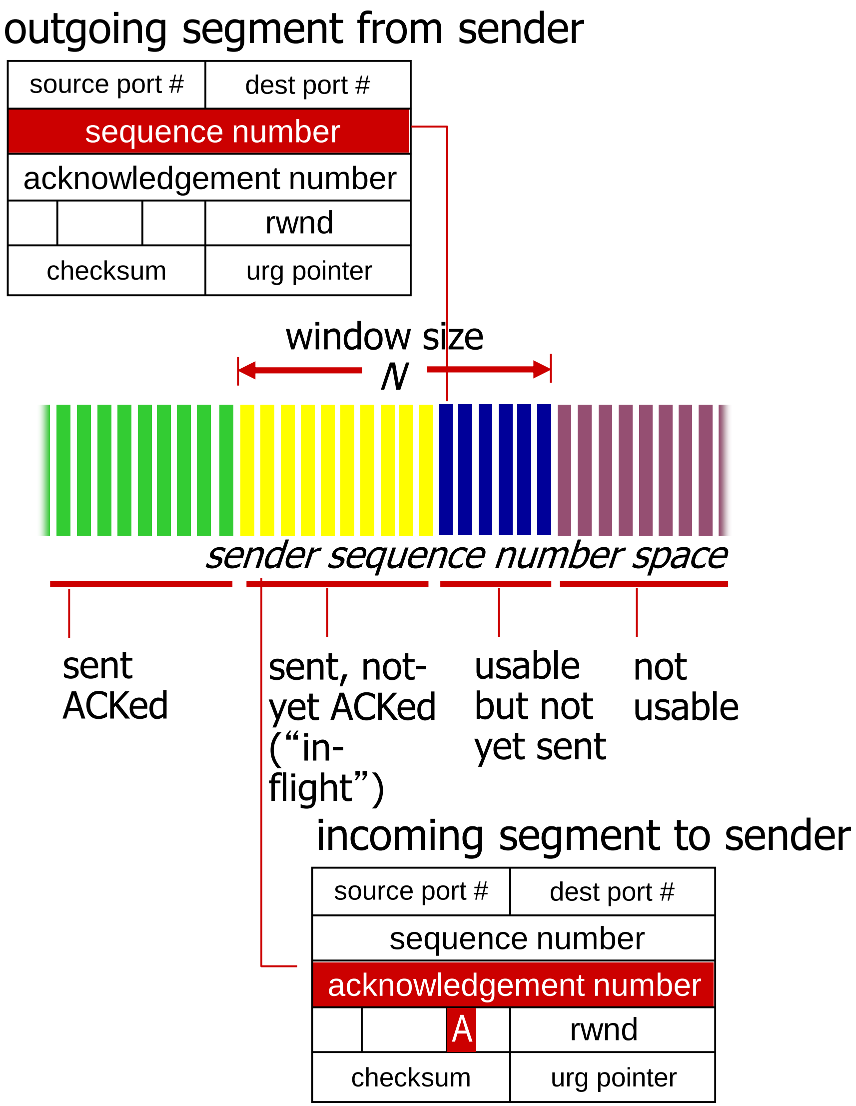
- byte stream "number" of first byte in segment's data

acknowledgements:

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

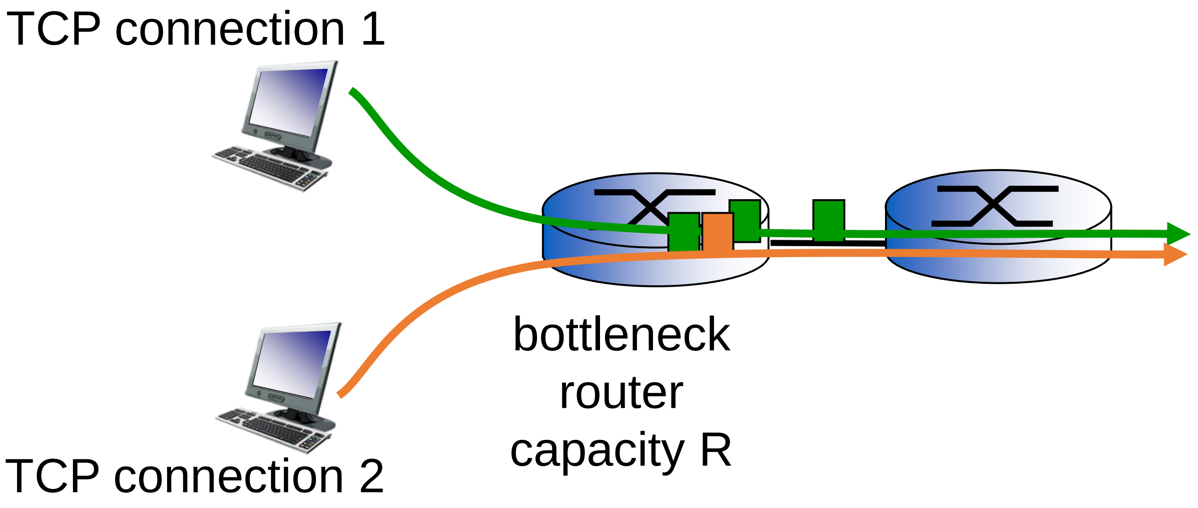
Q: how receiver handles out-of-order segments

- A: TCP spec doesn't say, - up to implementor



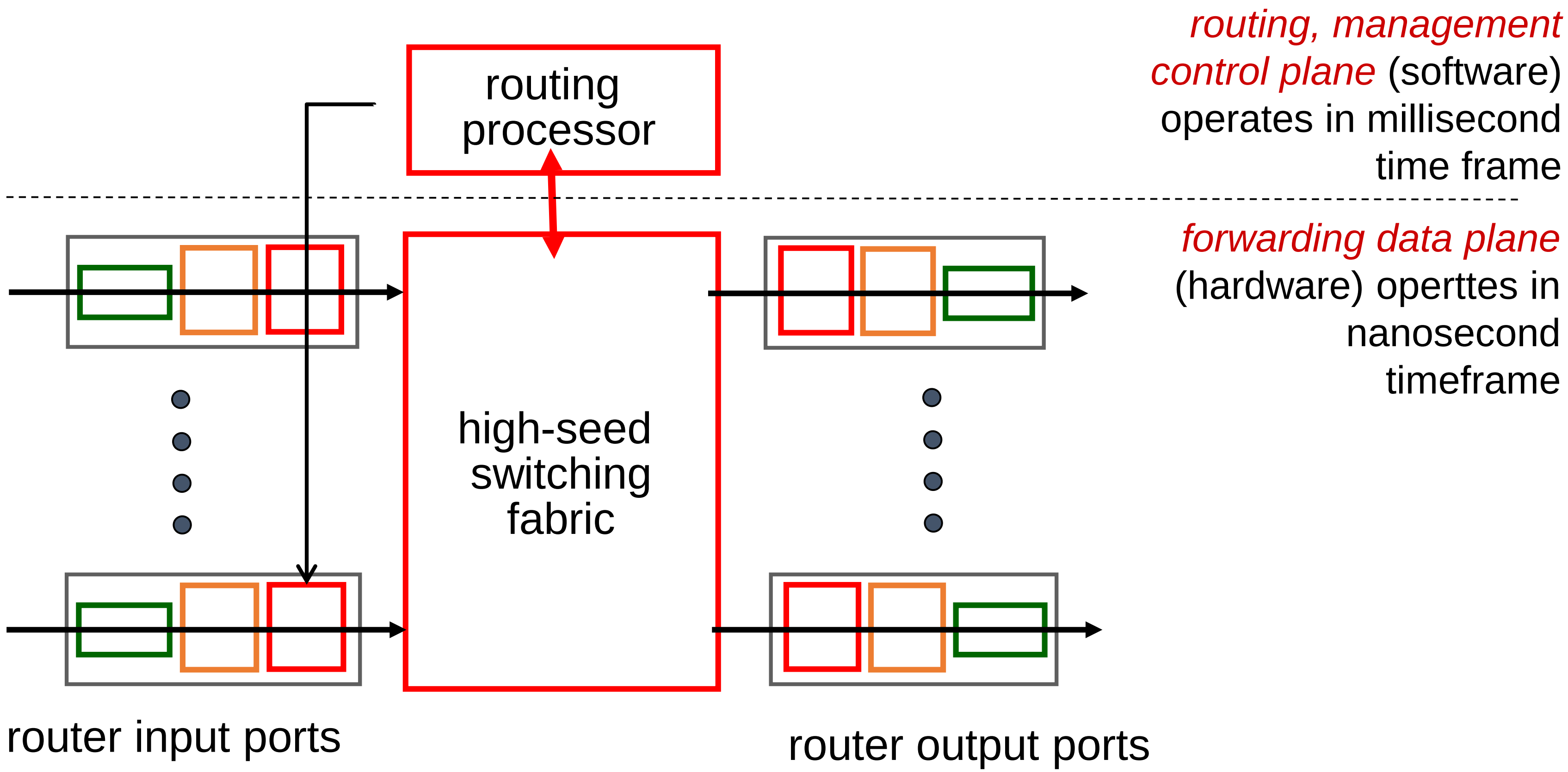
TCP Fairness

fairness goal: if K TCP sessions share same bottleneck link of bandwidth R, each should have average rate of R/K



Router architecture overview

- high-level view of generic router architecture:



Longest prefix matching

longest prefix matching

when looking for forwarding table entry for given destination address, use *longest* address prefix that matches destination address.

Destination Address Range	Link interface
11001000 00010111 00010*** *********	0
11001000 00010111 00011000 *********	1
11001000 00010111 00011*** *********	2
otherwise	3

examples:

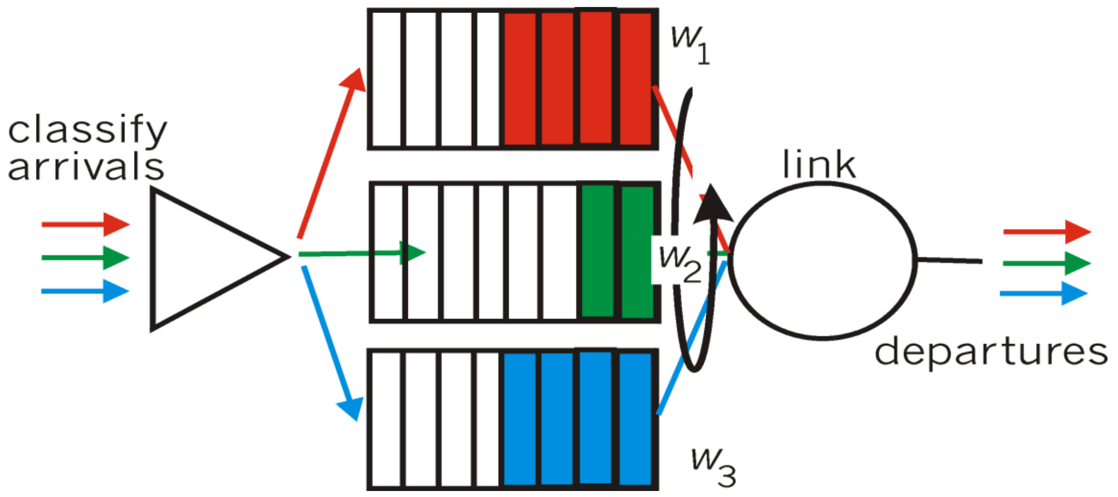
DA: 11001000 00010111 00010110 10100001 which interface?

DA: 11001000 00010111 00011000 10101010 which interface?

Scheduling policies: still more

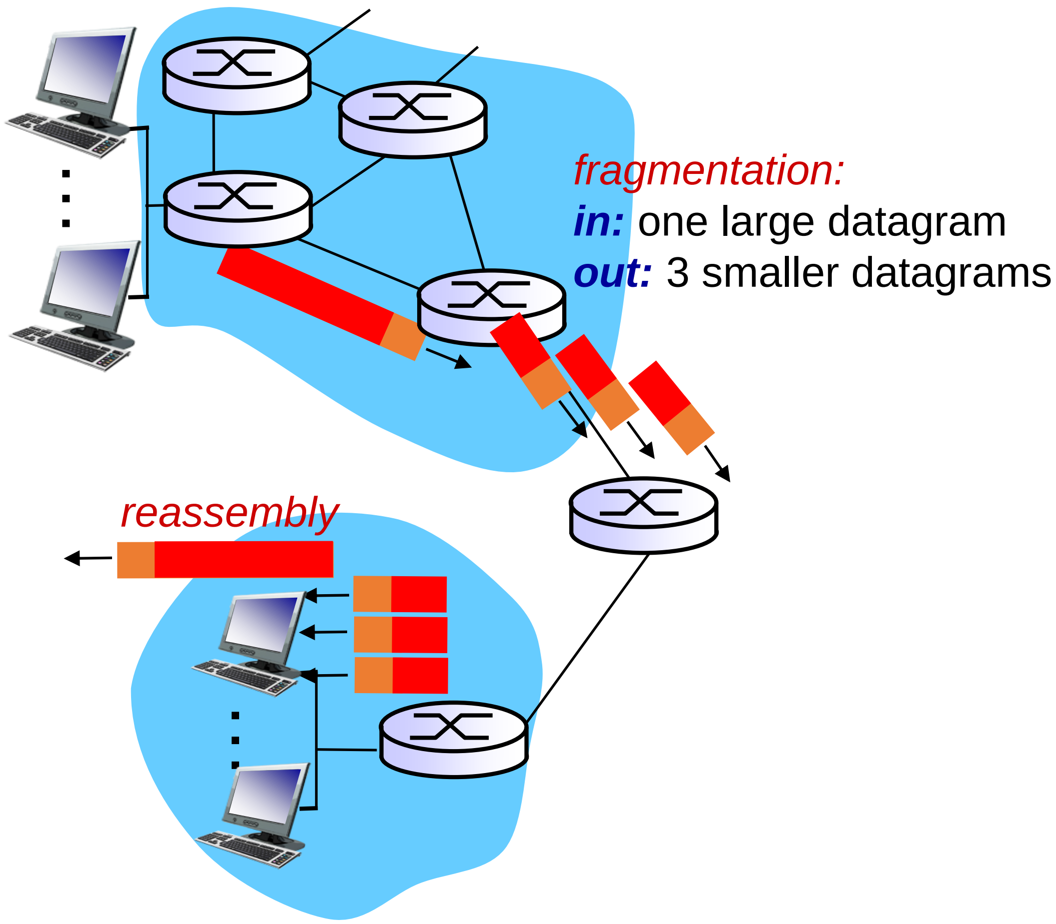
Weighted Fair Queuing (WFQ):

- generalized Round Robin
- each class gets weighted amount of service in each cycle



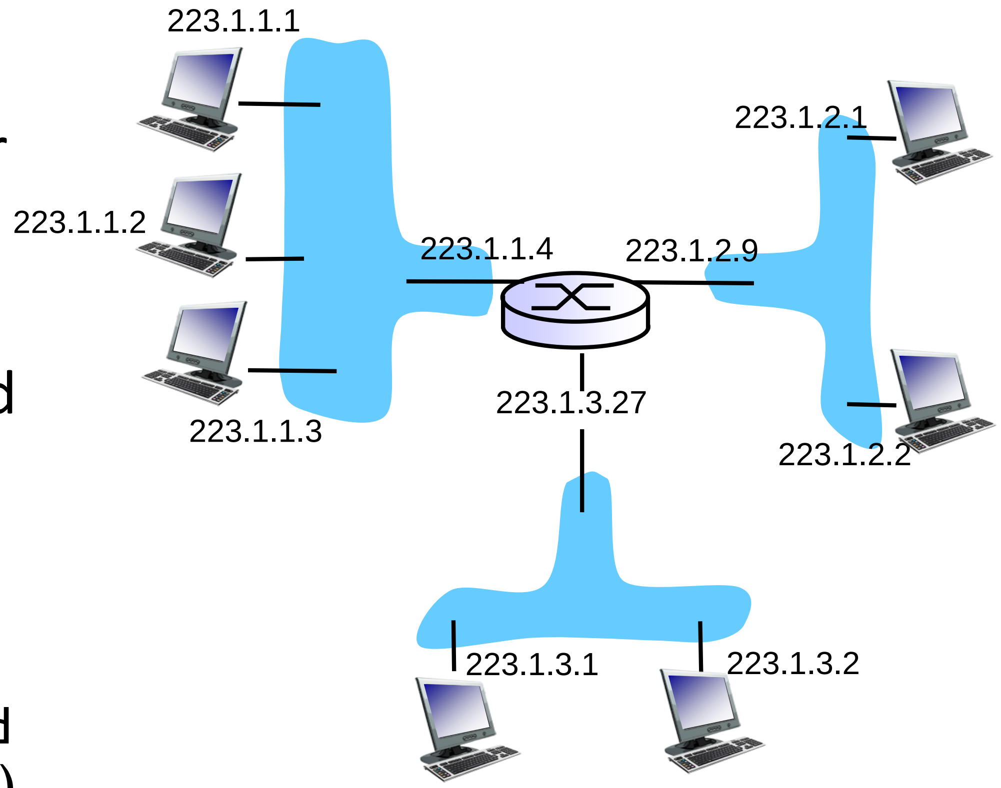
IP fragmentation, reassembly

- network links have MTU (max.transfer size) - largest possible link-level frame
  - different link types, different MTUs
- large IP datagram divided ("fragmented") within net
  - one datagram becomes several datagrams
  - "reassembled" only at final destination
  - IP header bits used to identify, order related fragments



IP addressing: introduction

- *IP address:* 32-bit identifier for host, router *interface*
- *interface:* connection between host/router and physical link
  - router's typically have multiple interfaces
  - host typically has one or two interfaces (e.g., wired Ethernet, wireless 802.11)
- *IP addresses associated with each interface*



$$223.1.1.1 = \underbrace{11011111}_{223}\ \underbrace{00000001}_{1}\ \underbrace{00000001}_{1}\ \underbrace{00000001}_{1}$$

IP addressing: CIDR

CIDR: Classless InterDomain Routing

- subnet portion of address of arbitrary length
- address format: a.b.c.d/x, where x is # bits in subnet portion of address

subnet part ←→ host part

11001000 00010111 00010000 00000000

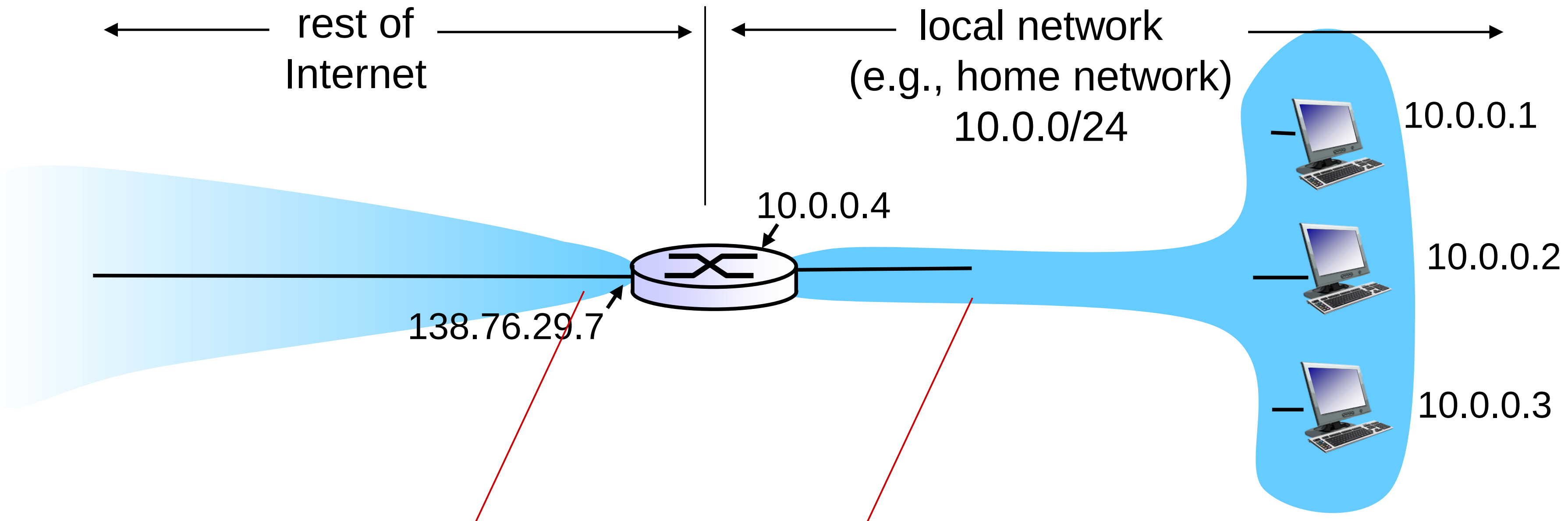
200.23.16.0/23

DHCP: more than IP addresses

DHCP can return more than just allocated IP address on subnet:

- address of first-hop router for client
- name and IP address of DNS sever
- network mask (indicating network versus host portion of address)

NAT: network address translation



all datagrams *leaving* local network have *same* single source NAT IP address: 138.76.29.7,different source port numbers

datagrams with source or destination in this network have 10.0.0/24 address for source, destination (as usual)

IPv6: motivation

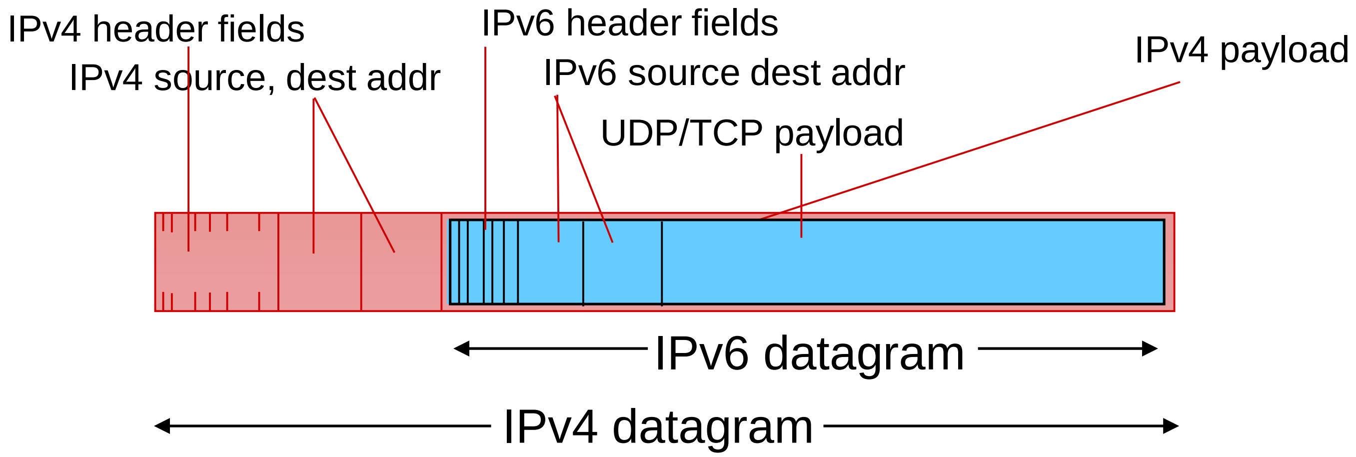
- *initial motivation:* 32-bit address space soon to be completely allocated.
- additional motivation:
  - header format helps speed processing/forwarding
  - header changes to facilitate QoS

IPv6 datagram format:

- fixed-length 40 byte header
- no fragmentation allowed

Transition from IPv4 to IPv6

- not all routers can be upgraded simultaneously
  - no "flag days"
  - how will network operate with mixed IPv4 and IPv6 routers?
- *tunneling:* IPv6 datagram carried as *payload* in IPv4 datagram among IPv4 routers



A link-state routing algorithm

Dijkstra's algorithm

- net topology, link costs known to all nodes
  - accomplished via "link state broadcast"
  - all nodes have same info
- computes least cost paths from one node ('source") to all other nodes
  - gives *forwarding table* for that node
- iterative: after k iterations, know least cost path to k dest.'s

notation:

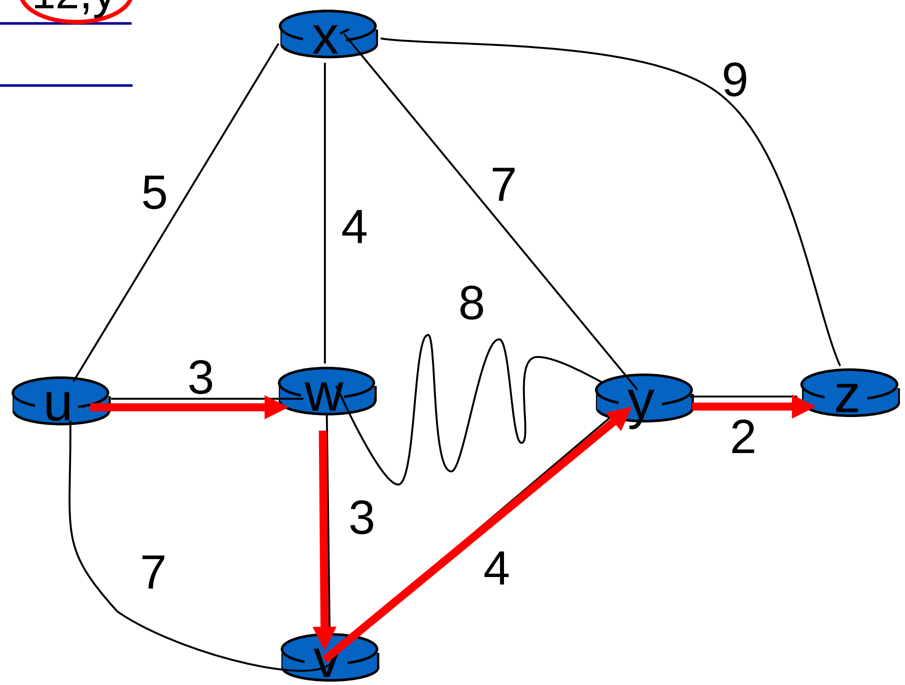
- c(x,y): link cost from node x to y; = ∞ if not direct neighbors
- D(v): current value of cost of path from source to dest. v
- p(v): predecessor node along path from source to v
- N': set of nodes whose least cost path definitively known

Dijkstra's algorithm: example

Step	N'	D(v) p(v)	D(w) p(w)	D(x) p(x)	D(y) p(y)	D(z) p(z)
0	u	7,u	3,u	5,u	∞	∞
1	uw	6,w		5,u	11,w	∞
2	uwx	6,w			11,w	14,x
3	uwxv				10,v	14,x
4	uwxvy					12,y
5	uwxvyz					

notes:

- construct shortest path tree by tracing predecessor nodes
- ties can exist (can be broken arbitrarily)



Distance vector algorithm

Bellman-Ford equation (dynamic programming)

let

$d_x(y)$:= cost of least-cost path from x to y

then

$$d_x(y) = \min_v \{c(x,v) + d_v(y)\}$$

cost from neighbor v to destination y

cost to neighbor v

min taken over all neighbors v of x

Distance vector algorithm

iterative, asynchronous:
each local iteration caused by:

- local link cost change
- DV update message from neighbor

distributed:

- each node notifies neighbors *only* when its DV changes
  - neighbors then notify their neighbors if necessary

each node:

wait for (change in local link cost or msg from neighbor)

recompute estimates

if DV to any dest has changed, *notify* neighbors

Comparison of LS and DV algorithms

message complexity

- ***LS:*** with n nodes, E links, O(nE) msgs sent
- ***DV:*** exchange between neighbors only
  - convergence time varies

speed of convergence

- ***LS:*** $O(n^2)$ algorithm requires O(nE) msgs
  - may have oscillations
- ***DV:*** convergence time varies
  - may be routing loops
  - count-to-infinity problem

robustness: what happens if router malfunctions?

LS:

- node can advertise incorrect *link* cost
- each node computes only its *own* table

DV:

- DV node can advertise incorrect *path* cost
- each node's table used by others
  - error propagate thru network

Internet approach to scalable routing

aggregate routers into regions known as "autonomous systems" (AS) (a.k.a. "domains")

intra-AS routing

- routing among hosts, routers in same AS ("network")
- all routers in AS must run *same* intra-domain protocol
- routers in *different* AS can run *different* intra-domain routing protocol
- gateway router: at "edge" of its own AS, has link(s) to router(s) in other AS'es

inter-AS routing

- routing among AS'es
- gateways perform inter-domain routing (as well as intra-domain routing)

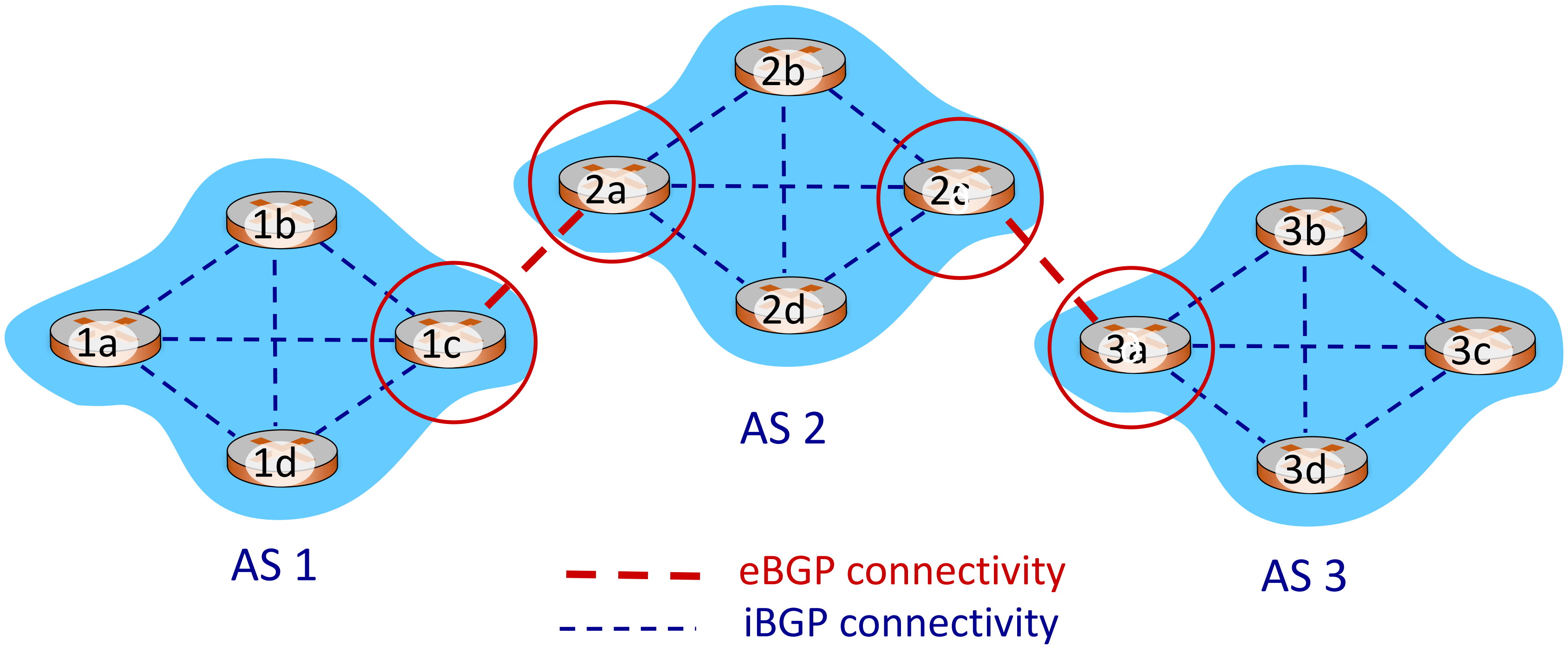
OSPF (Open Shortest Path First)

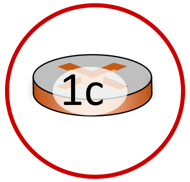
- "open": publicly available
- uses link-state algorithm
  - link state packet dissemination
  - topology map at each node
  - route computation using Dijkstra's algorithm
- router floods OSPF link-state advertisements to all other routers in *entire* AS
  - carried in OSPF messages directly over IP (rather than TCP or UDP
  - link state: for each attached link
- *IS-IS routing* protocol: nearly identical to OSPF

Internet inter-AS routing: BGP

- BGP (Border Gateway Protocol): *the* de facto inter-domain routing protocol
  - "glue that holds the Internet together"
- BGP provides each AS a means to:
  - eBGP: obtain subnet reachability information from neighboring ASes
  - iBGP: propagate reachability information to all AS-internal routers.
  - determine "good" routes to other networks based on reachability information and *policy*
- allows subnet to advertise its existence to rest of Internet: *"I am here"*

eBGP, iBGP connections





gateway routers run both eBGP and iBGP protocols

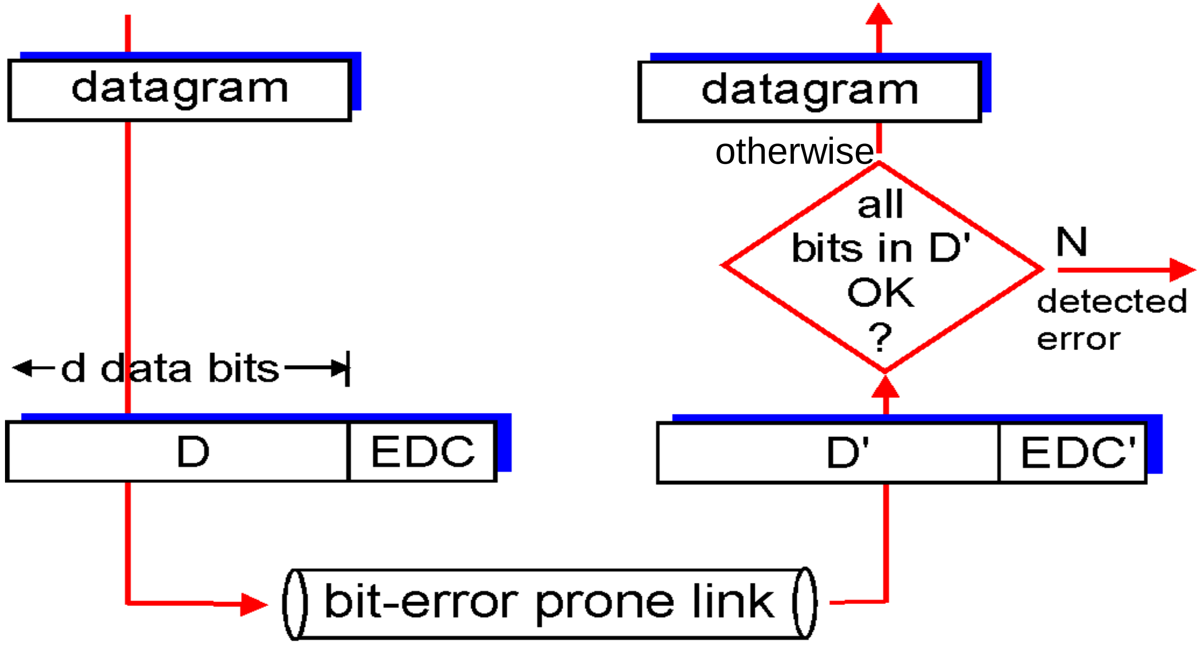
Link layer services

- *framing, link access:*
  - encapsulate datagram into frame, adding header, trailer
  - channel access if shared medium
  - "MAC" addresses used in frame headers to identify source, destination
    - different from IP address!
- *reliable delivery between adjacent nodes*
  - we learned how to do this already (chapter 3)!
  - seldom used on low bit-error link (fiber, some twisted pair)
  - wireless links: high error rates
    - *Q:* why both link-level and end-end reliability?

Error detection

EDC= Error Detection and Correction bits (redundancy)
D = Data protected by error checking, may include header fields

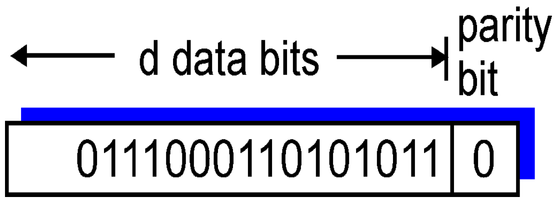
- Error detection not 100% reliable!
  - protocol may miss some errors, but rarely
  - larger EDC field yields better detection and correction



Parity checking

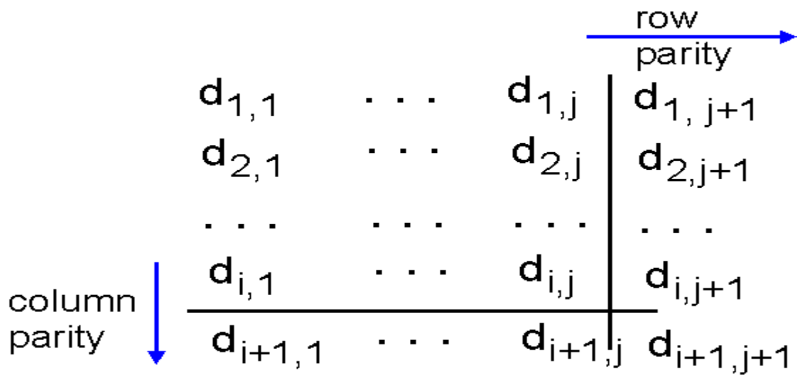
single bit parity:

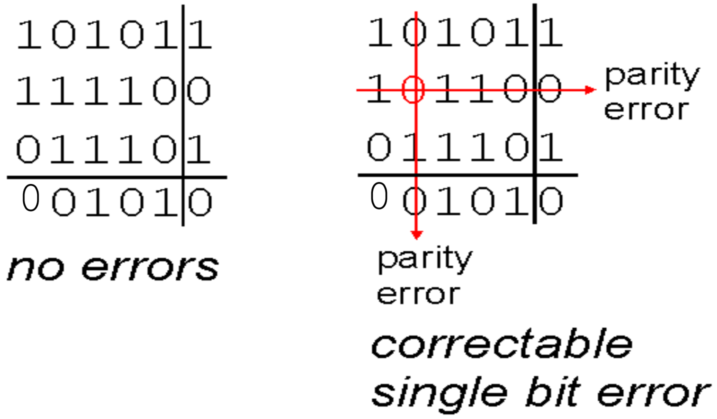
- *d*etect single bit errors



two-dimensional bit parity:

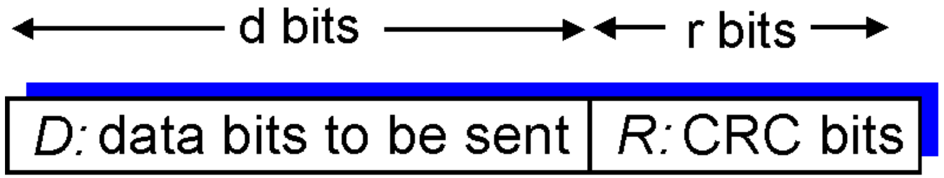
- detect and correct single bit errors





Cyclic redundancy check

- more powerful error-detection coding
- view data bits, D, as a binary number
- choose r+1 bit pattern (generator), G
- goal: choose r CRC bits, R, such that
  - <D,R> exactly divisible by G (modulo 2)
  - receiver knows G, divides <D,R> by G. If non-zero remainder: error detected!
  - can detect all burst errors less than r+1 bits
- widely used in practice (Ethernet, 802.11 WiFi, ATM)



bit pattern

$$D * 2^r \text{ XOR } R$$

mathematical formula

CRC example

want:

$D \cdot 2^r \text{ XOR } R = nG$

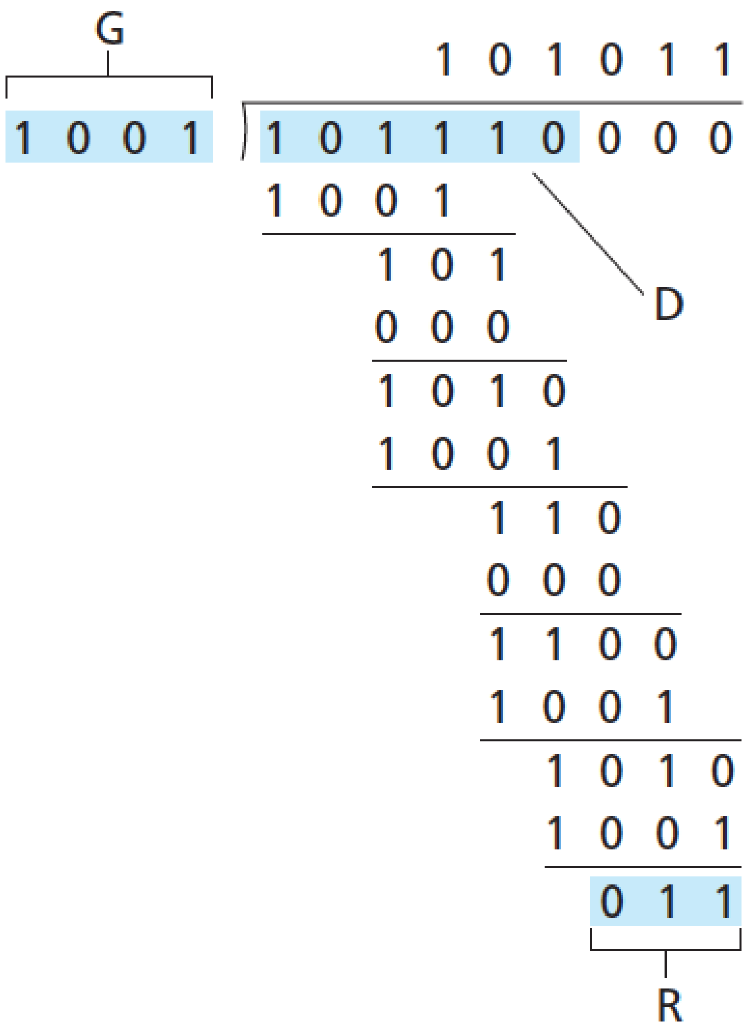
equivalently:

$D \cdot 2^r = nG \text{ XOR } R$

equivalently:

if we divide $D \cdot 2^r$ by G, want remainder R to satisfy:

$$R = remainder\left[\frac{D \cdot 2^r}{G}\right]$$



Multiple access protocols

- single shared broadcast channel
- two or more simultaneous transmissions by nodes: interference
  - *collision* if node receives two or more signals at the same time

multiple access protocol

- distributed algorithm that determines how nodes share channel, i.e., determine when node can transmit
- communication about channel sharing must use channel itself!
  - no out-of-band channel for coordination

- MAC – Medium(or Media) Access Control ← **Important Term!**

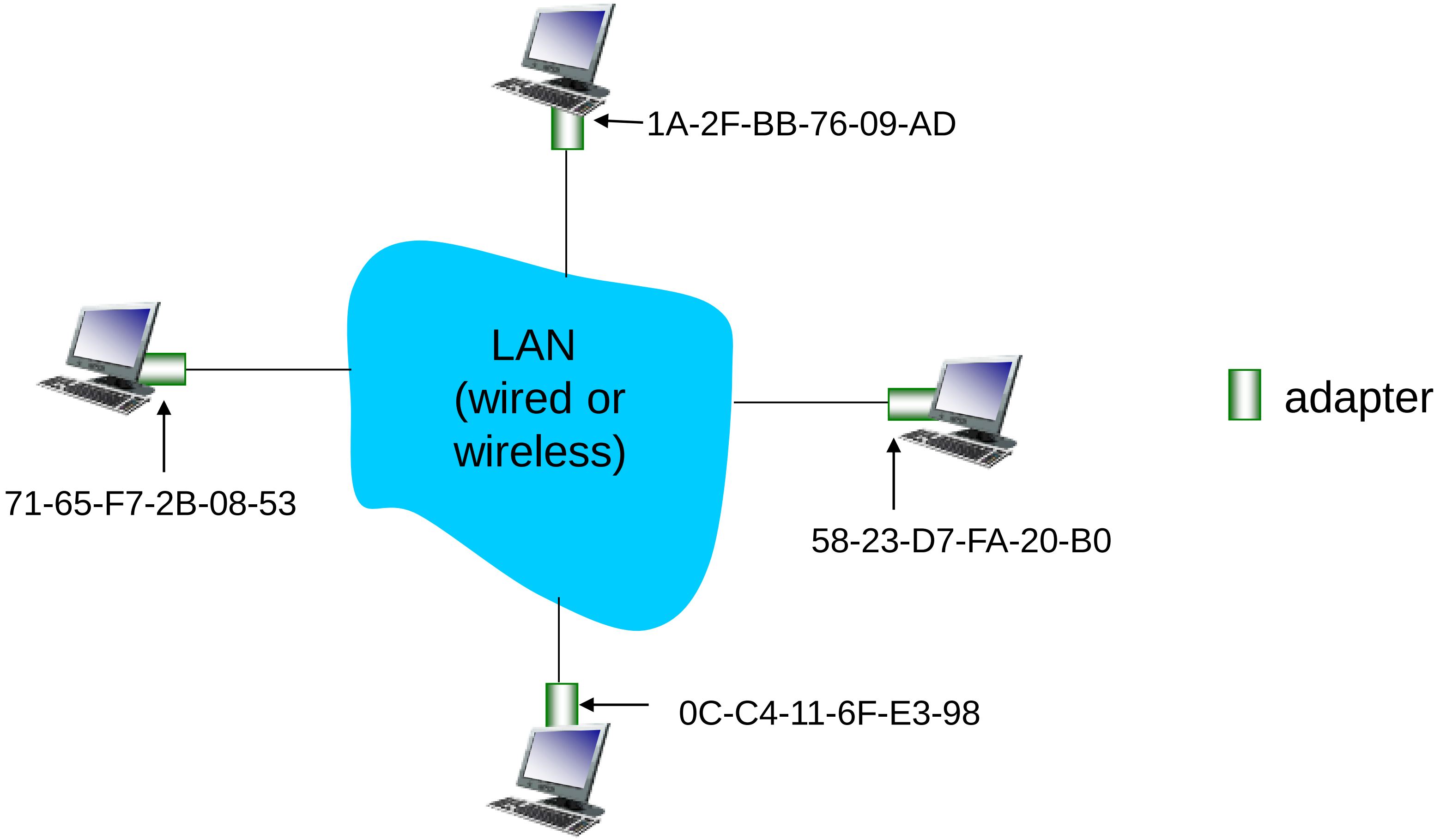
MAC addresses and ARP

- 32-bit IP address:
  - *network-layer* address for interface
  - used for layer 3 (network layer) forwarding
- MAC (or LAN or physical or Ethernet) address:
  - function: *used 'locally" to get frame from one interface to another physically-connected interface (same network, in IP-addressing sense)*
  - 48 bit MAC address (for most LANs) burned in NIC ROM, also sometimes software settable
  - e.g.: 1A-2F-BB-76-09-AD

hexadecimal (base 16) notation
(each "numeral" represents 4 bits)

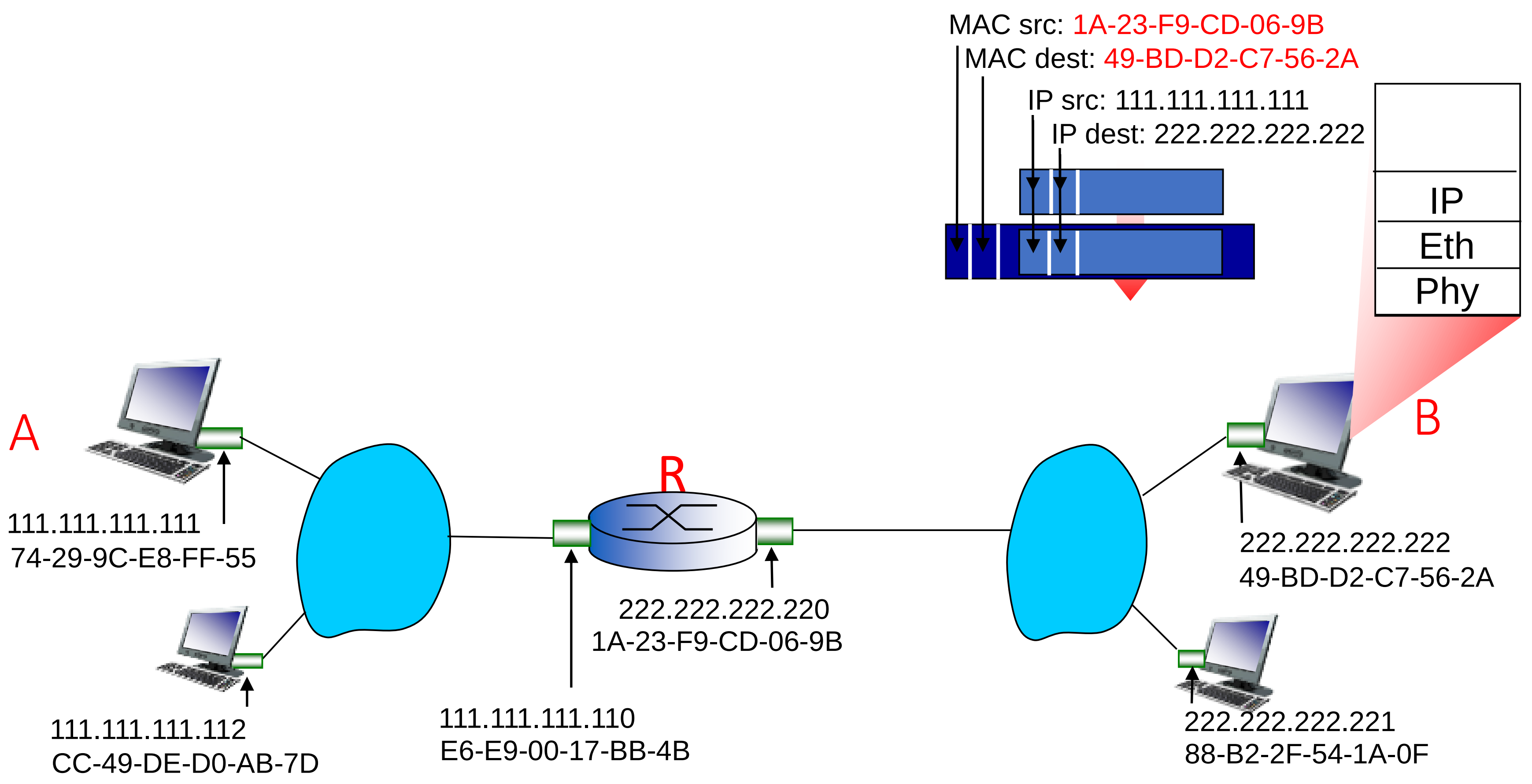
LAN addresses and ARP

each adapter on LAN has unique *LAN* address



Addressing: routing to another LAN

- R forwards datagram to destination B



Ethernet frame structure

sending adapter encapsulates IP datagram (or other network layer protocol packet) in Ethernet frame

preamble:

- 7 bytes with pattern 10101010 followed by one byte with pattern 10101011
- used to synchronize receiver, sender clock rates

Ethernet frame structure (more)

- *addresses:* 6 byte source, destination MAC addresses
  - if adapter receives frame with matching destination address, or with broadcast address (e.g. ARP packet), it passes data in frame to network layer protocol
  - otherwise, adapter discards frame
- *type:* indicates higher layer protocol (mostly IP but others possible, e.g., Novell IPX, AppleTalk)
- *CRC:* cyclic redundancy check at receiver
  - error detected: frame is dropped

Ethernet: unreliable, connectionless

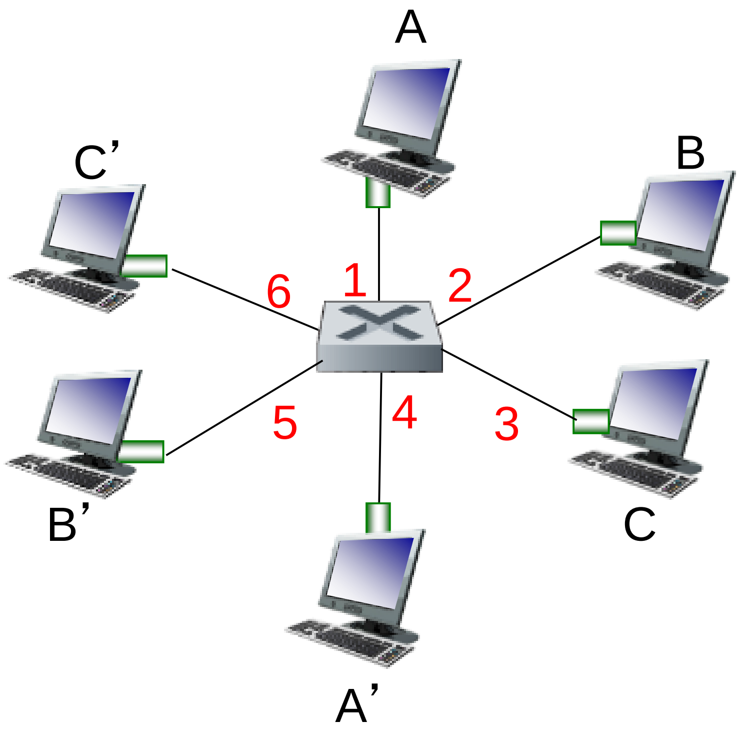
- *connectionless:* no handshaking between sending and receiving NICs
- *unreliable:* receiving NIC doesn't send acks or nacks to sending NIC
  - data in dropped frames recovered only if initial sender uses higher layer rdt (e.g., TCP), otherwise dropped data lost
- Ethernet's MAC protocol: unslotted *CSMA/CD with binary backoff*

Ethernet switch

- link-layer device: takes an *active* role
  - store, forward Ethernet frames
  - examine incoming frame's MAC address, selectively forward frame to one-or-more outgoing links when frame is to be forwarded on segment, uses CSMA/CD to access segment
- *transparent*
  - hosts are unaware of presence of switches
- *plug-and-play, self-learning*
  - switches do not need to be configured

Switch: *multiple* simultaneous transmissions

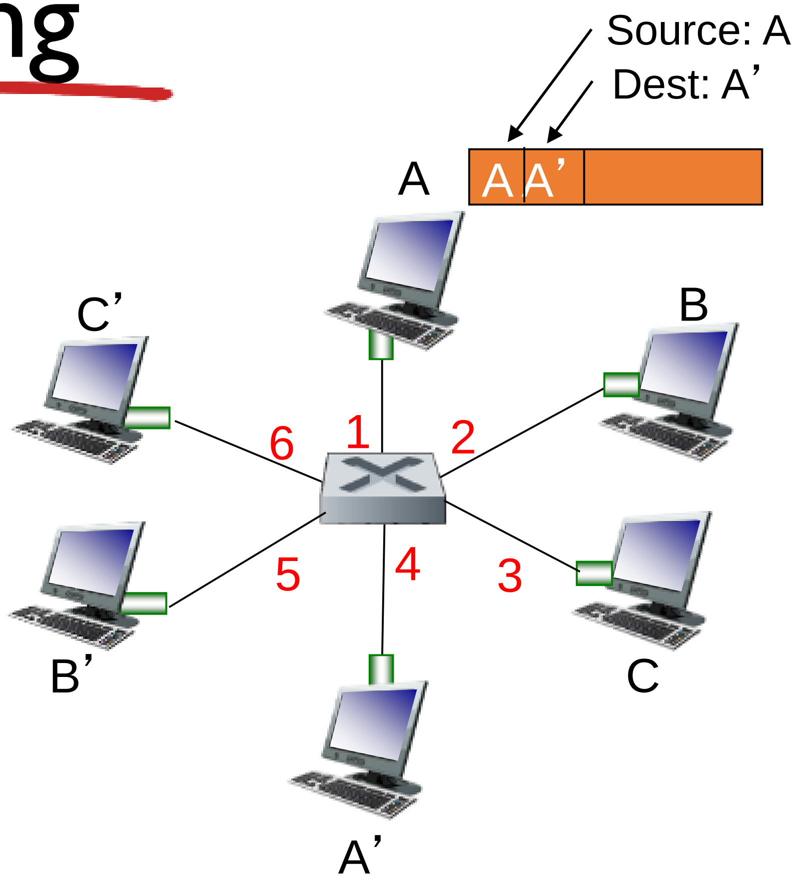
- hosts have dedicated, direct connection to switch
- switches buffer packets
- Ethernet protocol used on *each* incoming link, but no collisions; full duplex
  - each link is its own collision domain
- *switching:* A-to-A' and B-to-B' can transmit simultaneously, without collisions



switch with six interfaces (1,2,3,4,5,6)

Switch: self-learning

- switch *learns* which hosts can be reached through which interfaces
  - when frame received, switch "learns" location of sender: incoming LAN segment
  - records sender/location pair in switch table



MAC addr	interface	TTL
A	*1*	*60*

Switch table (initially empty)

Switch: frame filtering/forwarding

when frame received at switch:

1. record incoming link, MAC address of sending host
2. index switch table using MAC destination address
3. if entry found for destination

```
   then {
   if destination on segment from which frame arrived
       then drop frame
        else forward frame on interface indicated by entry
    }
    else flood  /* forward on all interfaces except arriving
                   interface */
```

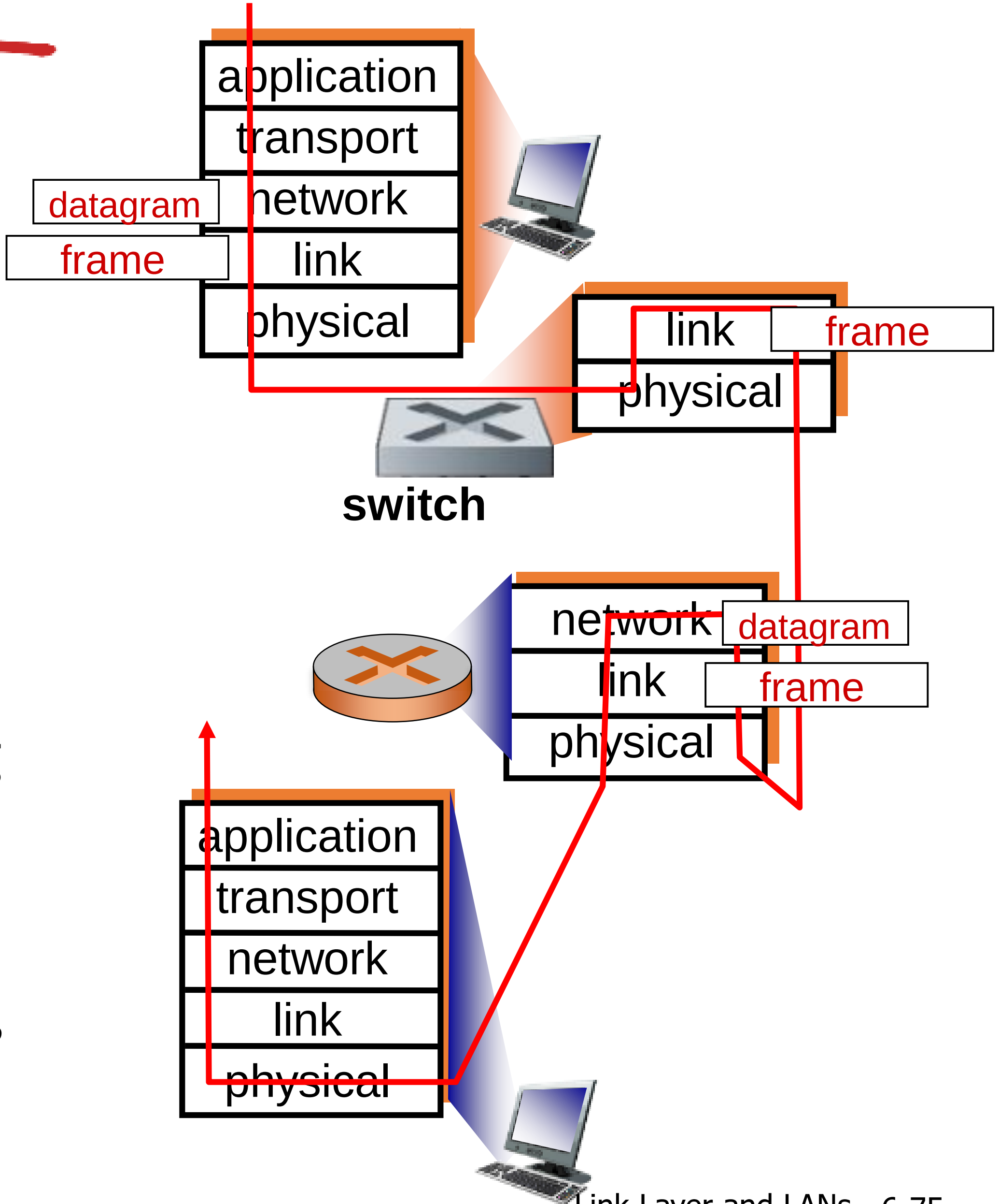
Switches vs. routers

both are store-and-forward:

- *routers:* network-layer devices (examine network-layer headers)
- *switches:* link-layer devices (examine link-layer headers)

both have forwarding tables:

- *routers:* compute tables using routing algorithms, IP addresses
- *switches:* learn forwarding table using flooding, learning, MAC addresses

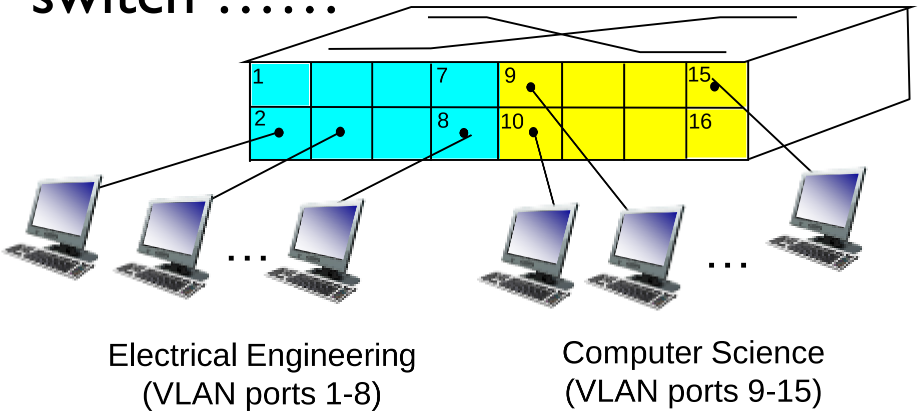


VLANs

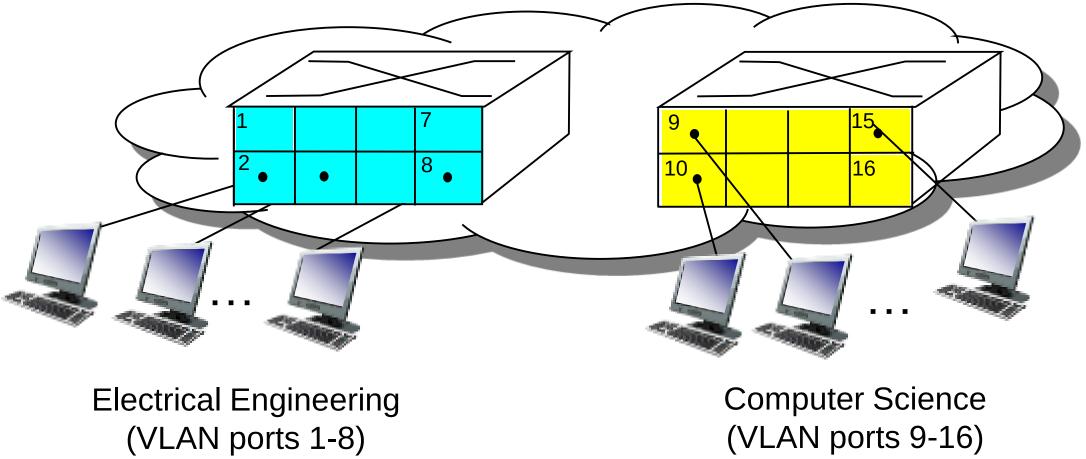
Virtual Local Area Network

switch(es) supporting VLAN capabilities can be configured to define multiple ***virtual*** LANS over single physical LAN infrastructure.

port-based VLAN: switch ports grouped (by switch management software) so that *single* physical switch ……



… operates as multiple virtual switches



Synthesis: a day in the life of a web request

- journey down protocol stack complete!
  - application, transport, network, link
- putting-it-all-together: synthesis!
  - *goal:* identify, review, understand protocols (at all layers) involved in seemingly simple scenario: requesting www page
  - *scenario:* student attaches laptop to campus network, requests/receives www.google.com

Wireless Link Characteristics (1)

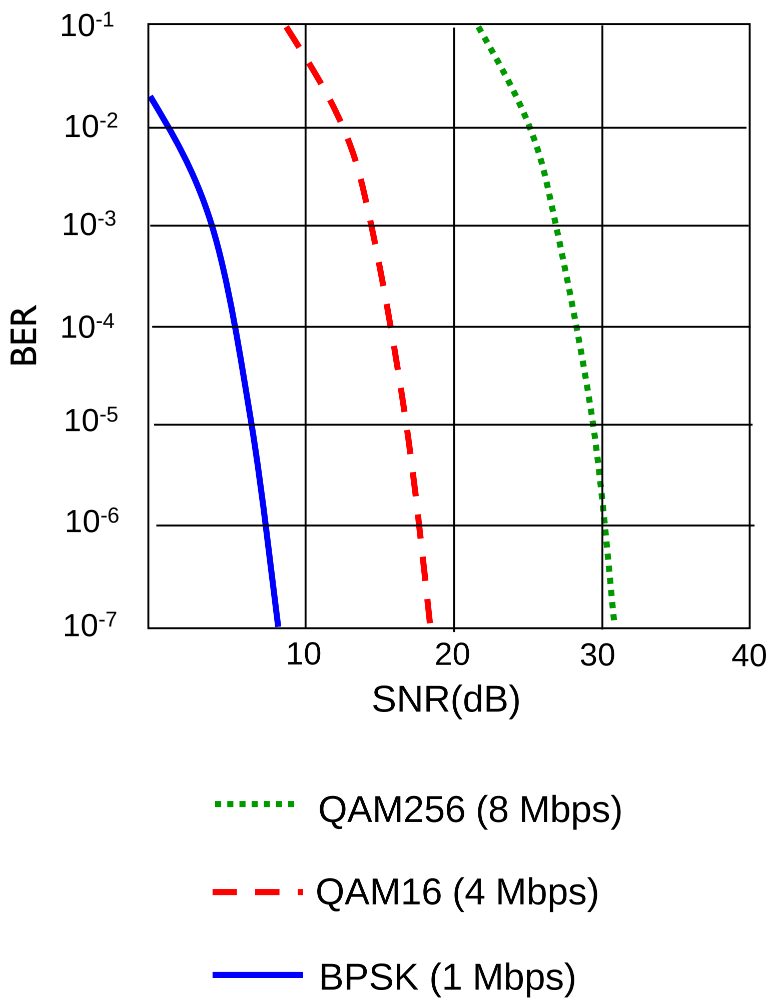
important differences from wired link ….

- *decreased signal strength:* radio signal attenuates as it propagates through matter (path loss)
- *interference from other sources:* standardized wireless network frequencies (e.g., 2.4 GHz) shared by other devices (e.g., phone); devices (motors) interfere as well
- *multipath propagation:* radio signal reflects off objects ground, arriving ad destination at slightly different times

…. make communication across (even a point to point) wireless link much more "difficult"

Wireless Link Characteristics (2)

- SNR: signal-to-noise ratio
  - larger SNR – easier to extract signal from noise (a "good thing")
- *SNR versus BER tradeoffs*
  - *given physical layer:* increase power -> increase SNR->decrease BER
  - *given SNR:* choose physical layer that meets BER requirement, giving highest thruput
    - SNR may change with mobility: dynamically adapt physical layer (modulation technique, rate)



IEEE 802.11 Wireless LAN

802.11b

- 2.4-5 GHz unlicensed spectrum
- up to 11 Mbps
- direct sequence spread spectrum (DSSS) in physical layer
  - all hosts use same chipping code

802.11a

- 5-6 GHz range
- up to 54 Mbps

802.11g

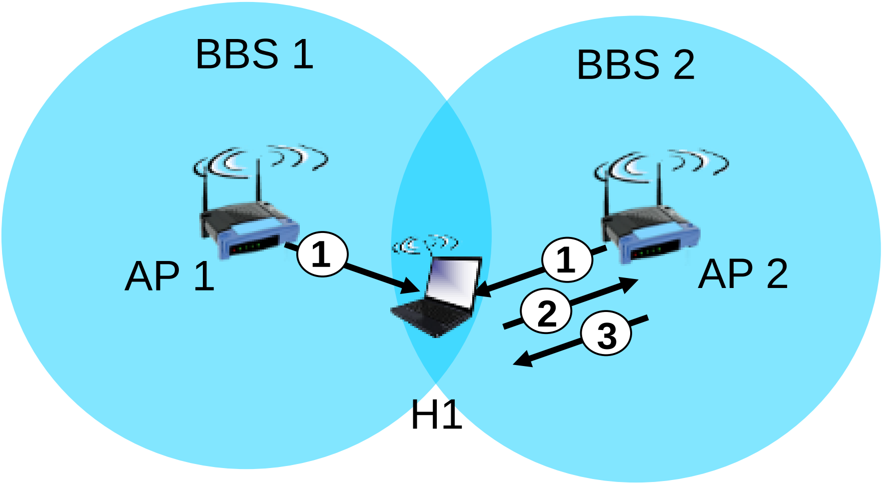
- 2.4-5 GHz range
- up to 54 Mbps

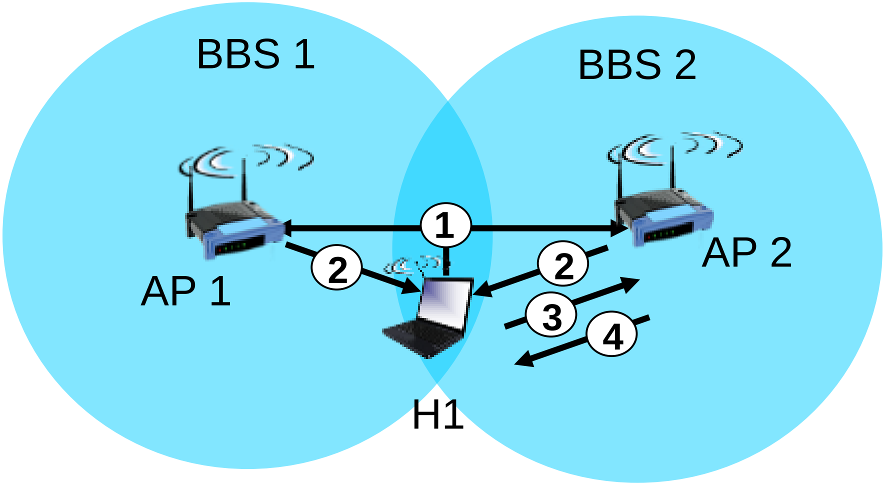
802.11n: multiple antennae

- 2.4-5 GHz range
- up to 200 Mbps

- all use CSMA/CA for multiple access
- all have base-station and ad-hoc network versions

802.11: passive/active scanning





passive scanning:

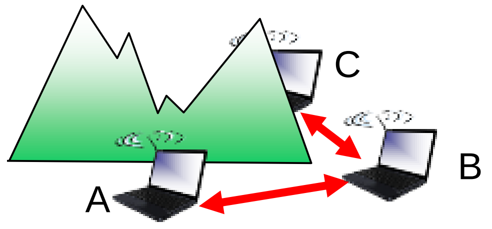
(1) beacon frames sent from APs
(2) association Request frame sent: H1 to selected AP
(3) association Response frame sent from selected AP to H1

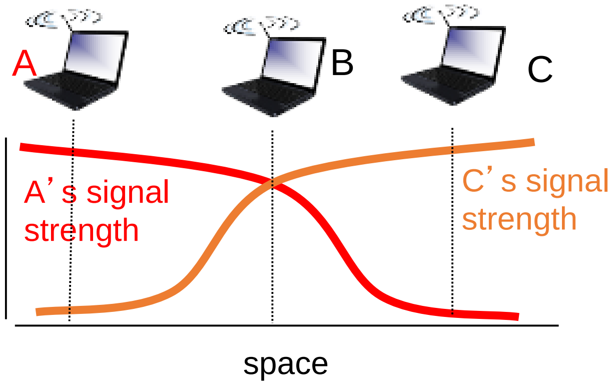
active scanning:

(1) Probe Request frame broadcast from H1
(2) Probe Response frames sent from APs
(3) Association Request frame sent: H1 to selected AP
(4) Association Response frame sent from selected AP to H1

IEEE 802.11: multiple access

- avoid collisions: 2$^+$ nodes transmitting at same time
- 802.11: CSMA - sense before transmitting
  - don't collide with ongoing transmission by other node
- 802.11: *no* collision detection!
  - difficult to receive (sense collisions) when transmitting due to weak received signals (fading)
  - can't sense all collisions in any case: hidden terminal, fading
  - goal: *avoid collisions:* CSMA/C(ollision)A(voidance)





What is network security?

confidentiality: only sender, intended receiver should "understand" message contents

- sender encrypts message
- receiver decrypts message

authentication: sender, receiver want to confirm identity of each other

message integrity: sender, receiver want to ensure message not altered (in transit, or afterwards) without detection

access and availability: services must be accessible and available to users

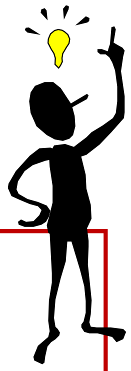
AES: Advanced Encryption Standard

- symmetric-key NIST standard, replaced DES (Nov 2001)
- processes data in 128 bit blocks
- 128, 192, or 256 bit keys
- brute force decryption (try each key) taking 1 sec on DES, takes 149 trillion years for AES

Public Key Cryptography

symmetric key crypto

- requires sender, receiver know shared secret key
- Q: how to agree on key in first place (particularly if never "met")?



public key crypto

- radically different approach [Diffie-Hellman76, RSA78]
- sender, receiver do *not* share secret key
- *public* encryption key known to *all*
- *private* decryption key known only to receiver

Public key encryption algorithms

requirements:

1. need $K_B^+(\cdot)$ and $K_B^-(\cdot)$ such that

$$K_B^-(K_B^+(m)) = m$$

2. given public key K_B^+, it should be impossible to compute private key K_B^-

RSA: Rivest, Shamir, Adelson algorithm

RSA: important property!!

The following property will be *very* useful later:

$$\underbrace{K_B^-(K_B^+(m))}_{\text{}} = m = \underbrace{K_B^+(K_B^-(m))}_{\text{}}$$

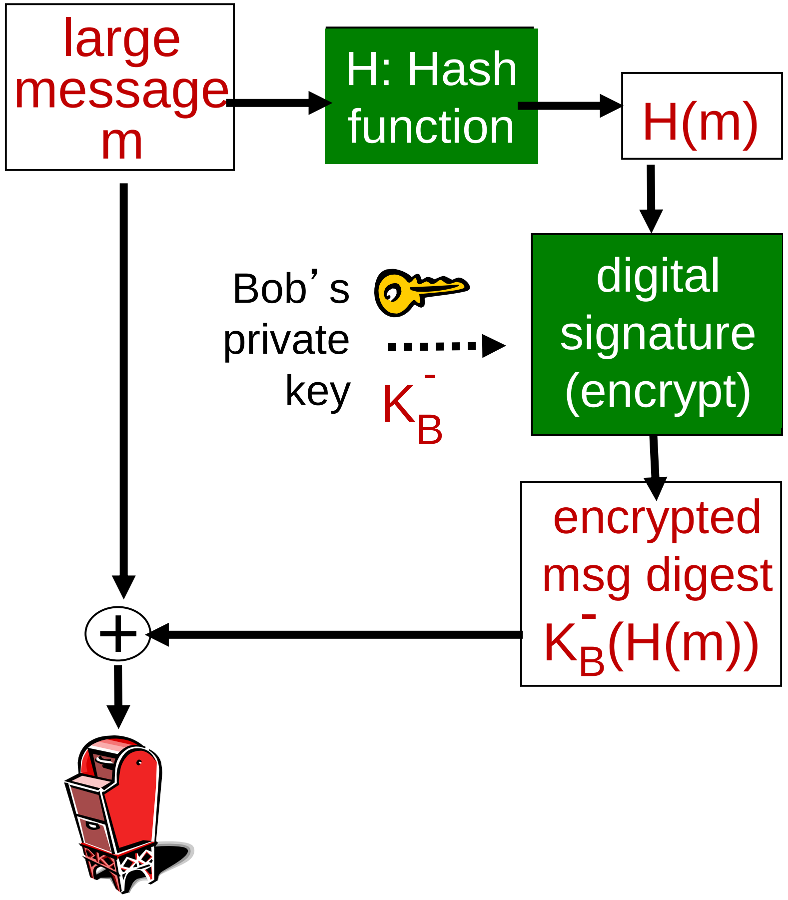
use public key first, followed by private key

use private key first, followed by public key

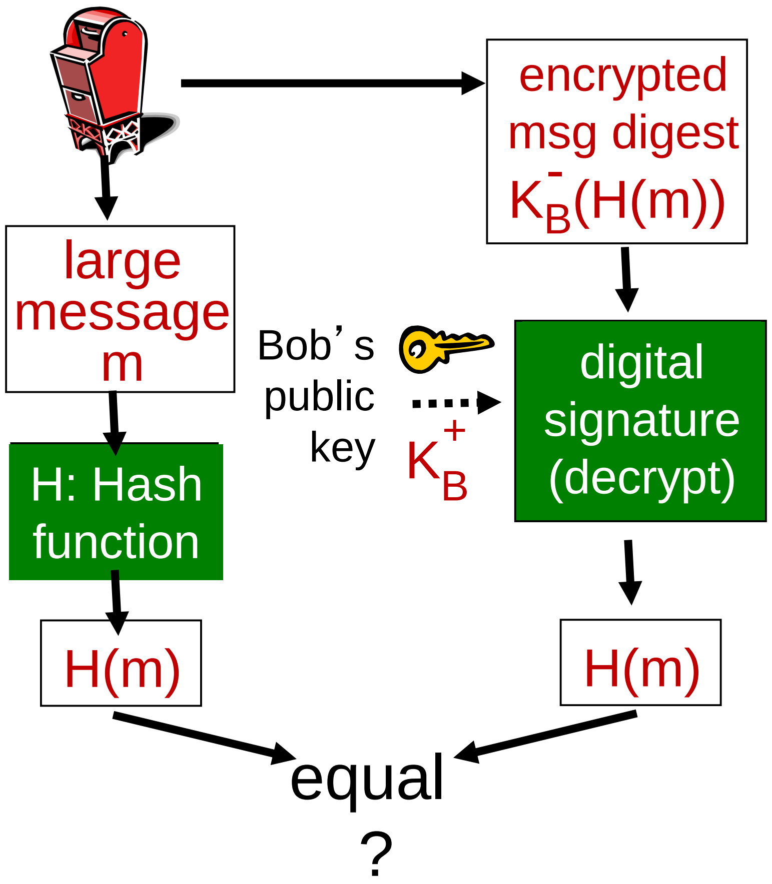
result is the same!

Digital signature = signed message digest

Bob sends digitally signed message:



Alice verifies signature, integrity of digitally signed message:

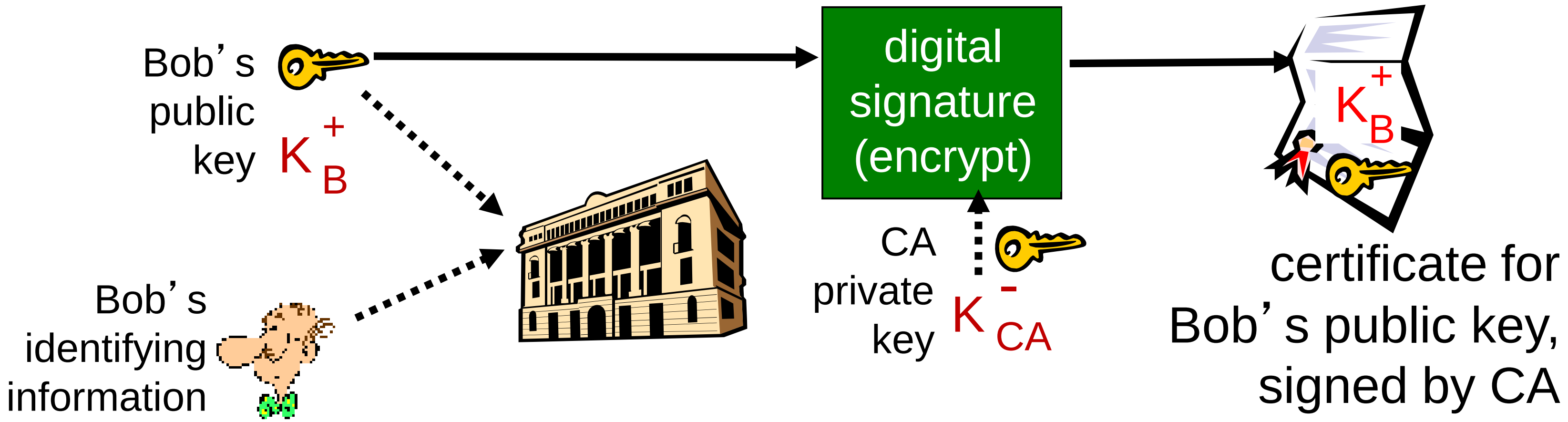


Hash function algorithms

- MD5 hash function widely used (RFC 1321)
  - computes 128-bit message digest in 4-step process.
  - arbitrary 128-bit string x, appears difficult to construct msg m whose MD5 hash is equal to x
- SHA-1 is also used
  - US standard [NIST, FIPS PUB 180-1]
  - 160-bit message digest

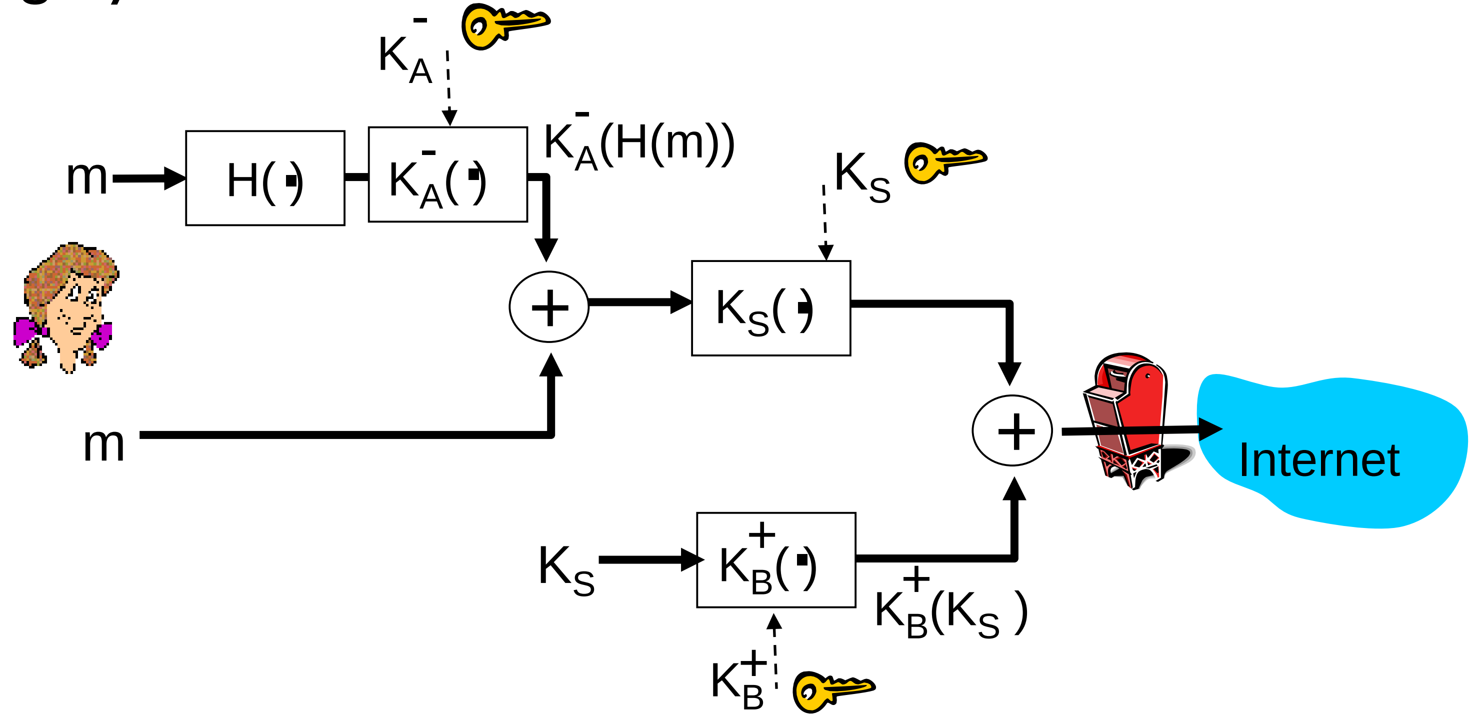
Certification authorities

- *certification authority (CA):* binds public key to particular entity, E.
- E (person, router) registers its public key with CA.
  - E provides "proof of identity" to CA.
  - CA creates certificate binding E to its public key.
  - certificate containing E's public key digitally signed by CA – CA says "this is E's public key"



Secure e-mail (continued)

Alice wants to provide secrecy, sender authentication, message integrity.



Alice uses three keys: her private key, Bob's public key, newly created symmetric key

Key derivation

- client nonce, server nonce, and pre-master secret input into pseudo random-number generator.
  - produces master secret
- master secret and new nonces input into another random-number generator: "key block"
  - because of resumption: TBD
- key block sliced and diced:
  - client MAC key
  - server MAC key
  - client encryption key
  - server encryption key
  - client initialization vector (IV)
  - server initialization vector (IV)

Breaking 802.11 WEP encryption

security hole:

- 24-bit initialization vector (IV), one IV per frame, -> IV's eventually reused
- IV transmitted in plaintext -> IV reuse detected

attack:

- Trudy causes Alice to encrypt known plaintext d_1 d_2 d_3 d_4 …
- Trudy sees: $c_i = d_i$ XOR k_i^{IV}
- Trudy knows c_i d_i, so can compute k_i^{IV}
- Trudy knows encrypting key sequence k_1^{IV} k_2^{IV} k_3^{IV} …
- Next time IV is used, Trudy can decrypt!

802.11i: improved security

- numerous (stronger) forms of encryption possible
- provides mechanism for key distribution (EAP)
- Can use authentication server separate from access point (Enterprise mode)
- WEP<WPA<WPA2
- WPA still uses RC4 but has larger IVs and uses a 256 bit key. TKIP makes sure each client gets a new key. Fixes major holes in WEP, but only meant as a stop-gap before WPA2
- WPA2 – 256 bit keys, TKIP & RC4 replaced with CCMP & AES. More secure, but still not 100%

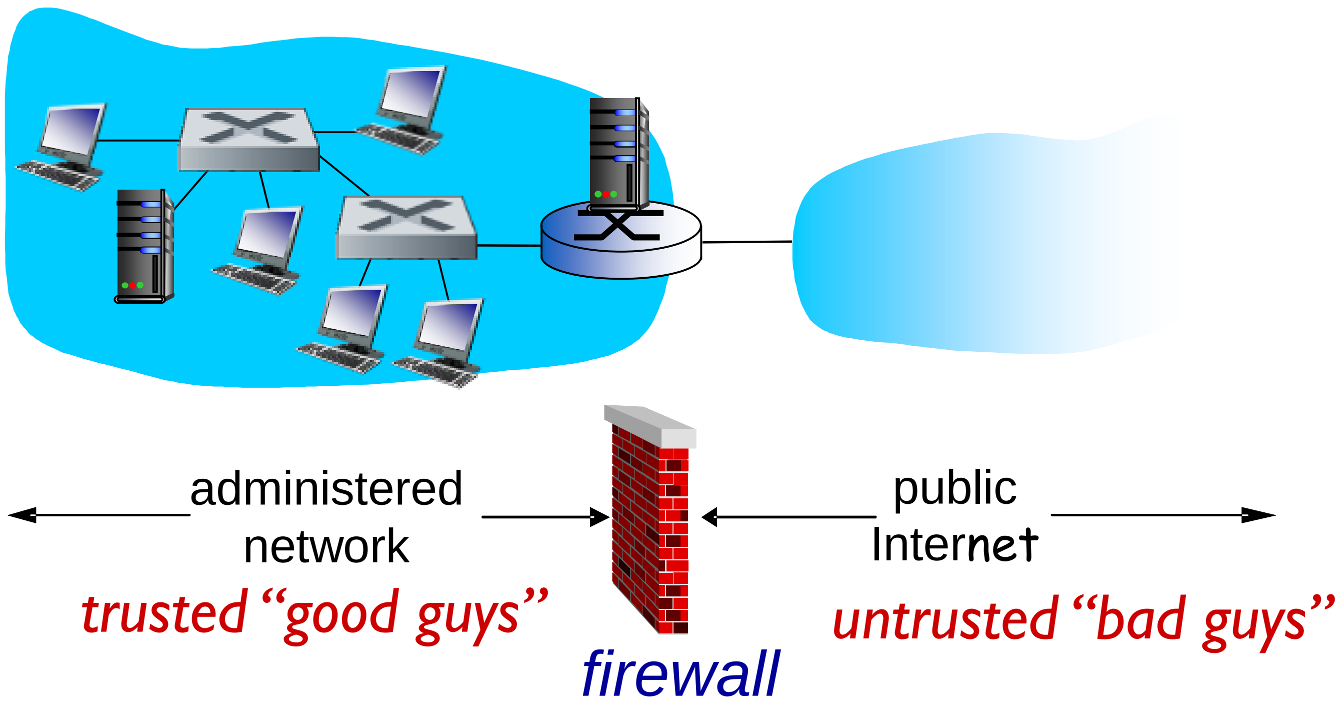
802.11i: continued

- Security issues mainly related to enterprise attacks, home networks can be considered secure with WPA2
- WPS on home routers – don’t do it.

Firewalls

firewall

isolates organization's internal net from larger Internet, allowing some packets to pass, blocking others



Firewalls: why

prevent denial of service attacks:

- SYN flooding: attacker establishes many bogus TCP connections, no resources left for "real" connections

prevent illegal modification/access of internal data

- e.g., attacker replaces CIA's homepage with something else

allow only authorized access to inside network

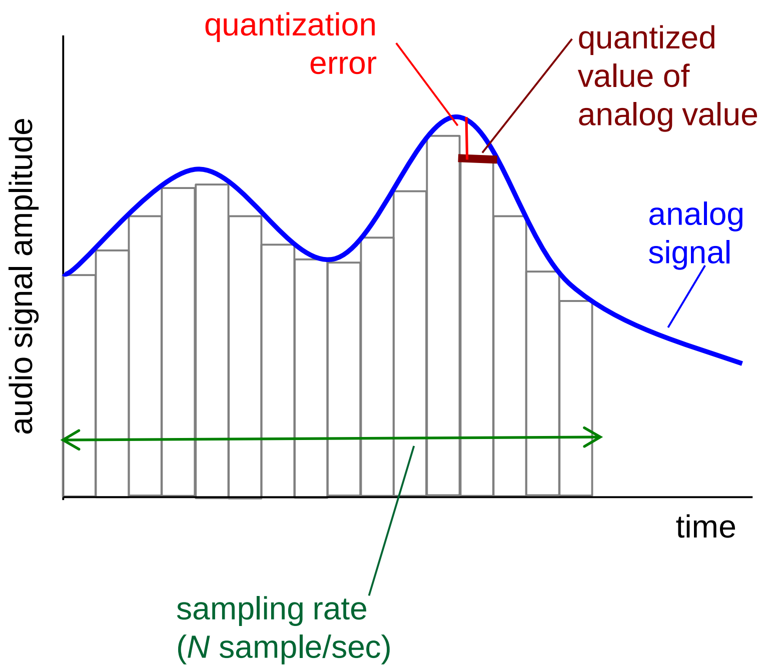
- set of authenticated users/hosts

three types of firewalls:

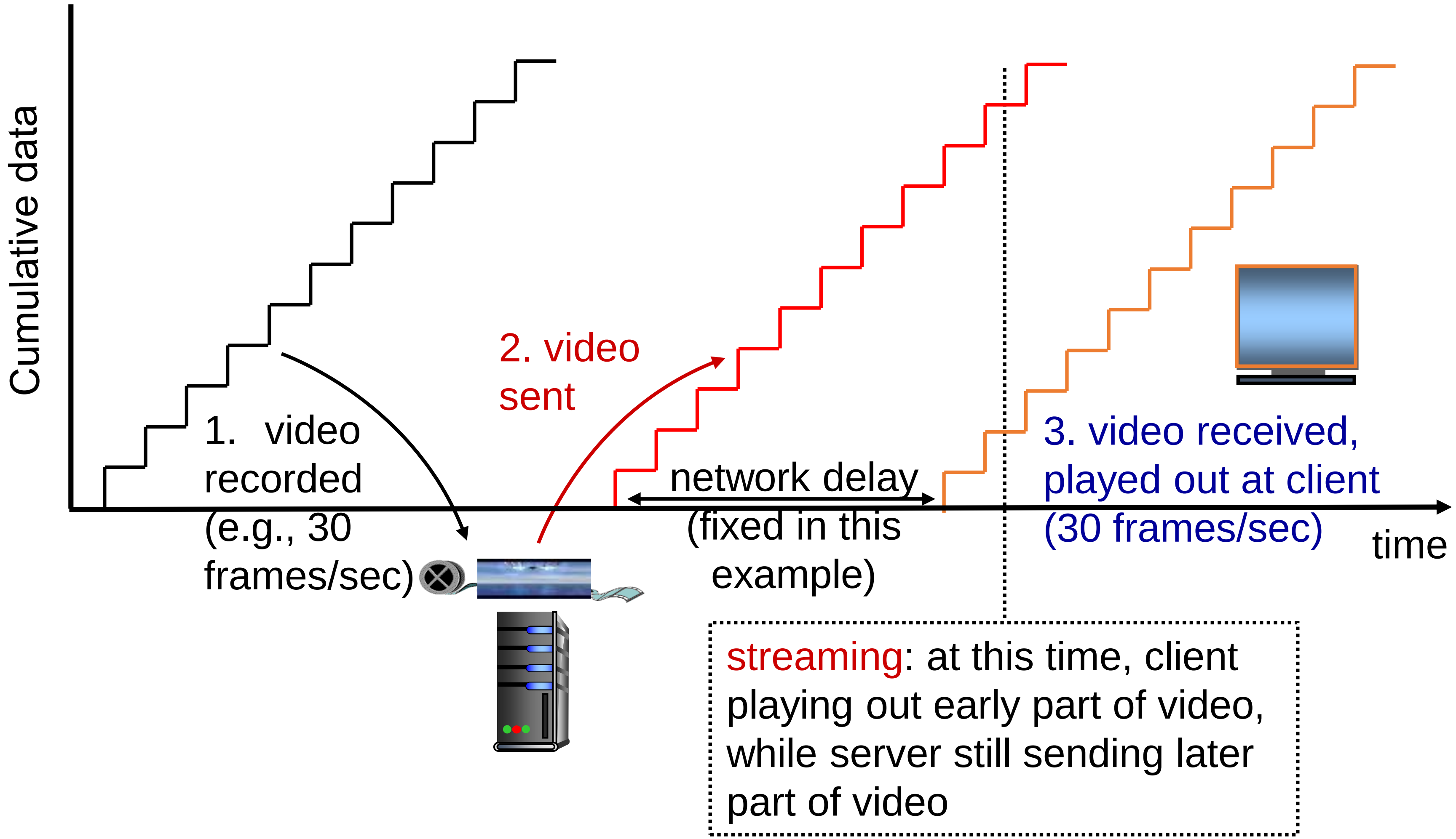
- stateless packet filters
- stateful packet filters
- application gateways

Multimedia: audio

- analog audio signal sampled at constant rate
  - telephone: 8,000 samples/sec
  - CD music: 44,100 samples/sec
- each sample quantized, i.e., rounded
  - e.g., 2^8=256 possible quantized values
  - each quantized value represented by bits, e.g., 8 bits for 256 values



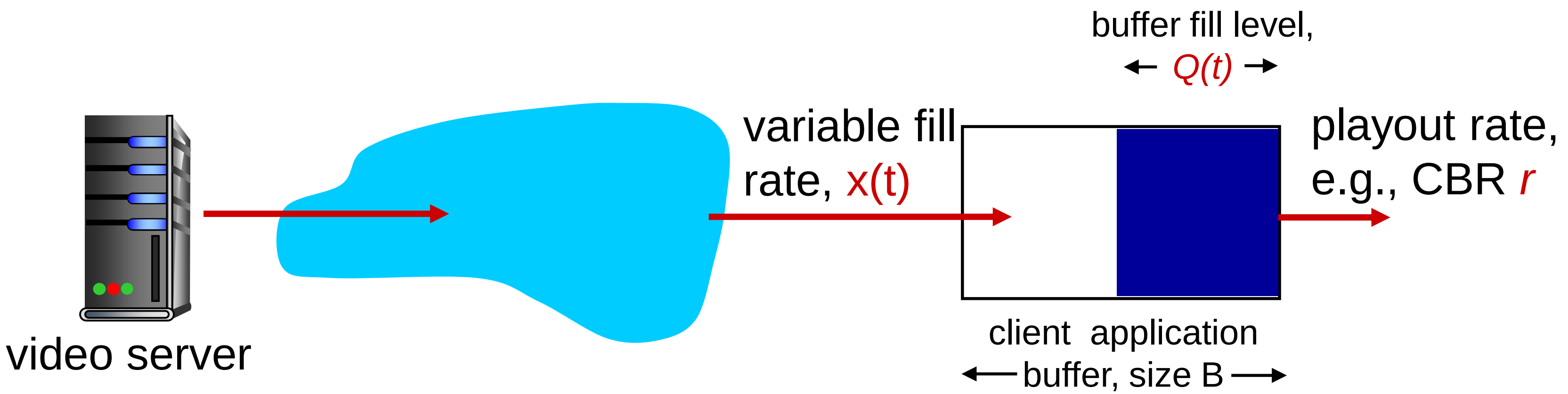
Streaming stored video:



Streaming stored video: challenges

- continuous playout constraint: once client playout begins, playback must match original timing
  - … but network delays are variable (jitter), so will need client-side buffer to match playout requirements
- other challenges:
  - client interactivity: pause, fast-forward, rewind, jump through video
  - video packets may be lost, retransmitted

Client-side buffering, playout



playout buffering: average fill rate ($\bar{x}$), playout rate (r):

- $\bar{x} < r$: buffer eventually empties (causing freezing of video playout until buffer again fills)
- $\bar{x} > r$: buffer will not empty, provided initial playout delay is large enough to absorb variability in x(t)
  - *initial playout delay tradeoff:* buffer starvation less likely with larger delay, but larger delay until user begins watching

Voice-over-IP (VoIP)

- *VoIP end-end-delay requirement*: needed to maintain "conversational" aspect
  - higher delays noticeable, impair interactivity
  - < 150 msec: good
  - > 400 msec bad
  - includes application-level (packetization, playout), network delays
- *session initialization:* how does callee advertise IP address, port number, encoding algorithms?
- *value-added services:* call forwarding, screening, recording
- *emergency services:* 911

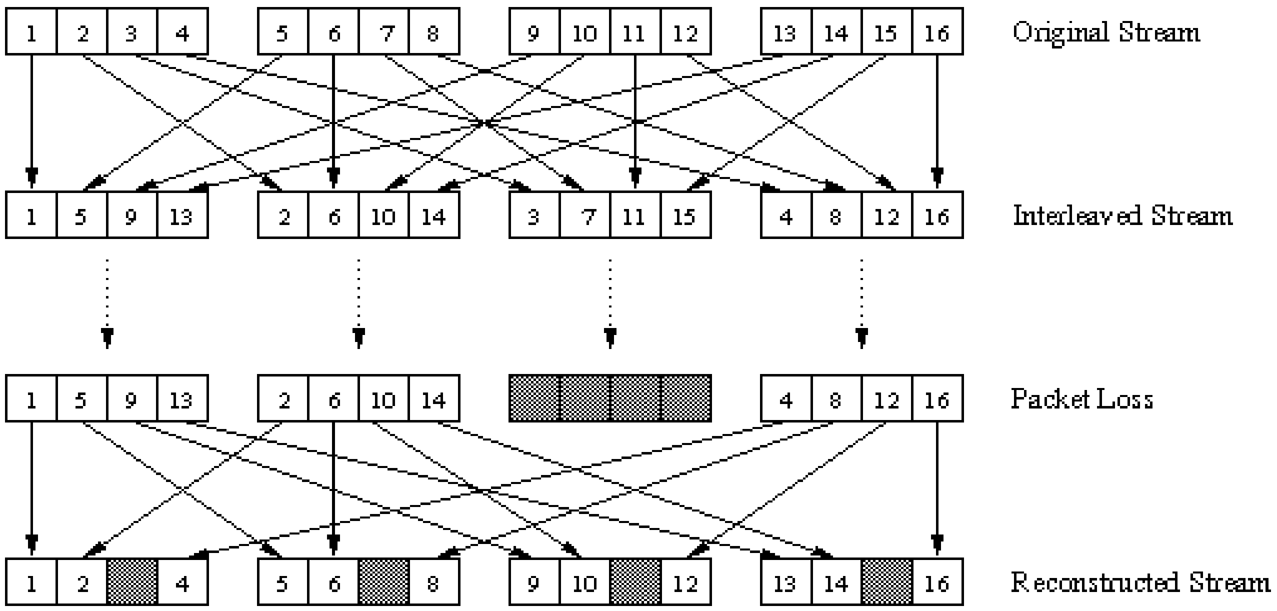
VoIP characteristics

- speaker's audio: alternating talk spurts, silent periods.
  - 64 kbps during talk spurt
  - pkts generated only during talk spurts
  - 20 msec chunks at 8 Kbytes/sec: 160 bytes of data
- application-layer header added to each chunk
- chunk+header encapsulated into UDP or TCP segment
- application sends segment into socket every 20 msec during talkspurt

VoIP: packet loss, delay

- *network loss:* IP datagram lost due to network congestion (router buffer overflow)
- *delay loss:* IP datagram arrives too late for playout at receiver
  - delays: processing, queueing in network; end-system (sender, receiver) delays
  - typical maximum tolerable delay: 400 ms
- *loss tolerance:* depending on voice encoding, loss concealment, packet loss rates between 1% and 10% can be tolerated

VoiP: recovery from packet loss (3)



interleaving to conceal loss:

- audio chunks divided into smaller units, e.g. four 5 msec units per 20 msec audio chunk
- packet contains small units from different chunks

- if packet lost, still have *most* of every original chunk
- no redundancy overhead, but increases playout delay