



# CSCI 6760 - Computer Networks - Fall 2024

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# Application layer: overview

## Our goals:

- conceptual *and* implementation aspects of application-layer protocols
  - transport-layer service models
  - client-server paradigm
  - peer-to-peer paradigm
- learn about protocols by examining popular application-layer protocols
  - HTTP
  - SMTP, IMAP
  - DNS
- programming network applications
  - socket API

# Some network apps

- social networking
- Web
- text messaging
- e-mail
- multi-user network games
- streaming stored video  
(YouTube, Hulu, Netflix)
- P2P file sharing
- voice over IP (e.g., Skype)
- real-time video conferencing
- Internet search
- remote login
- ...

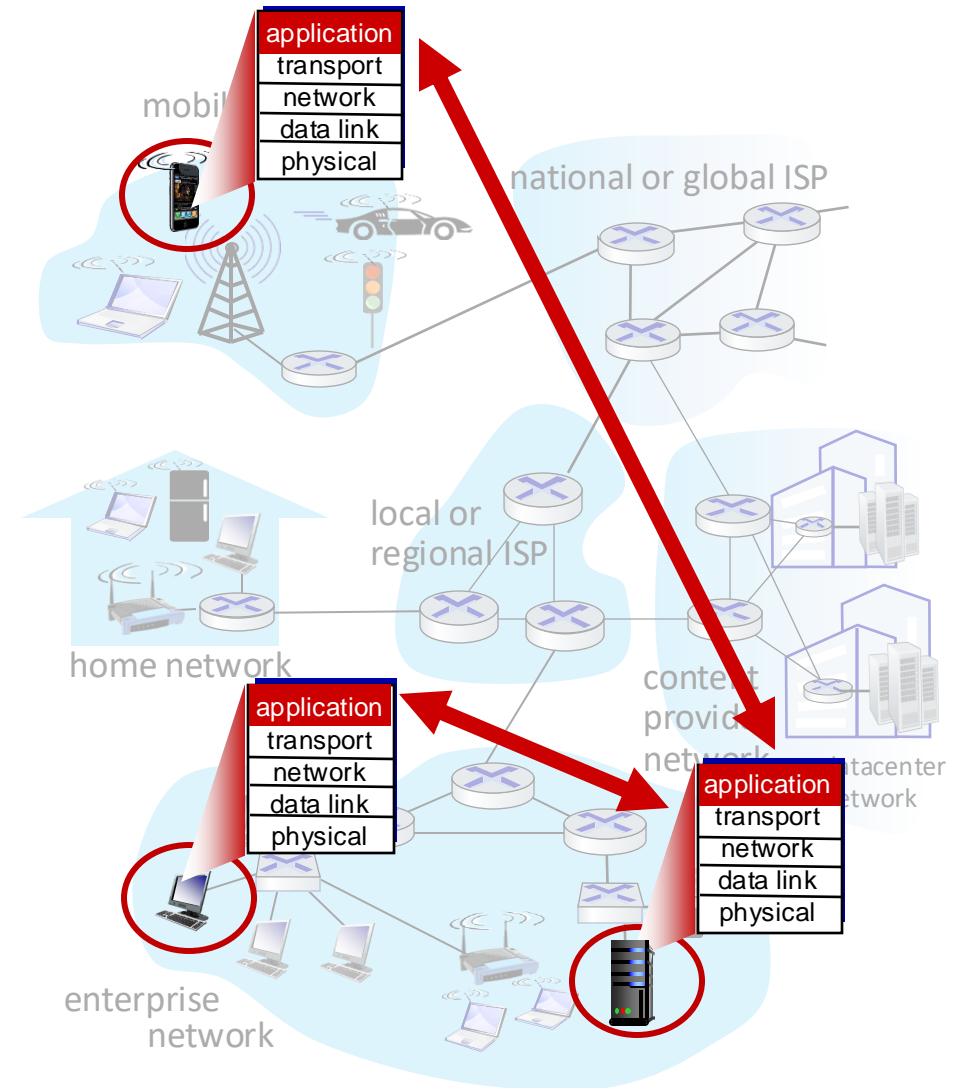
# Creating a network app

write programs that:

- run on (different) end systems
- communicate over network
- e.g., web server software communicates with browser software

no need to write software for network-core devices

- network-core devices do not run user applications
- applications on end systems allows for rapid app development, propagation



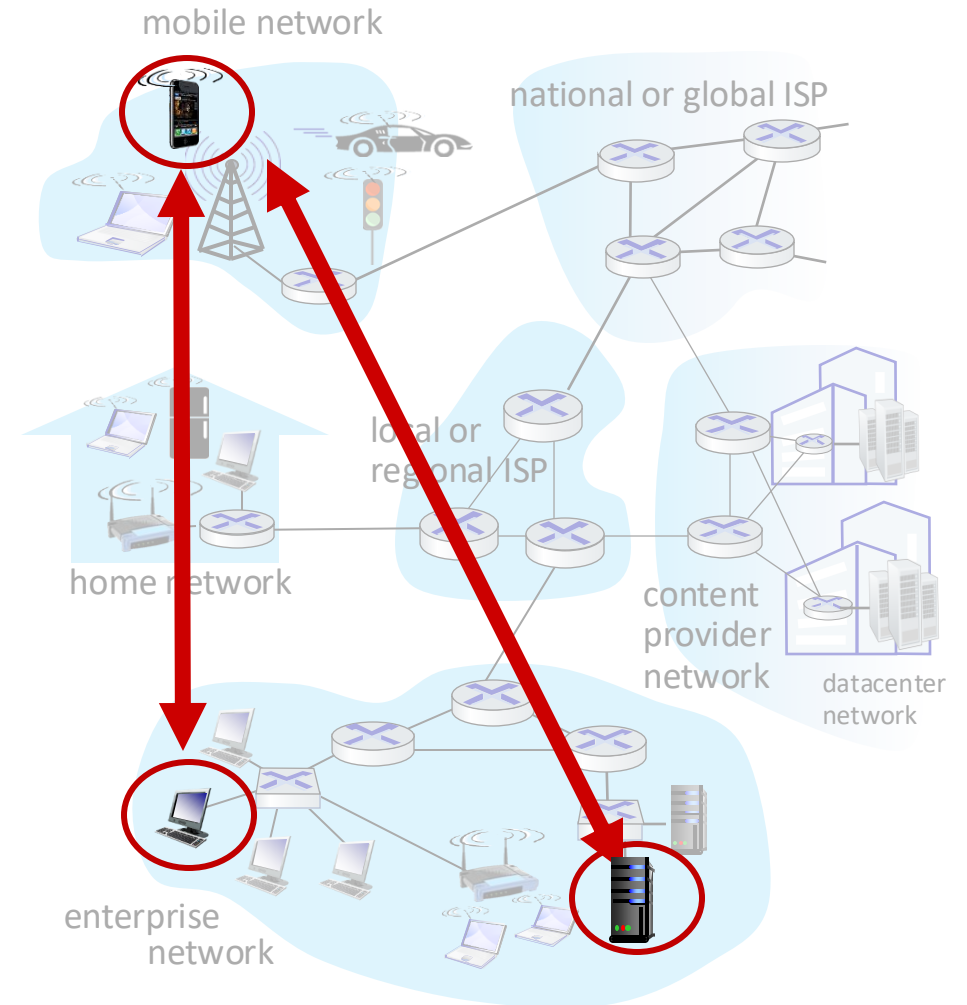
# Client-server paradigm

## server:

- always-on host
- permanent IP address
- often in data centers, for scaling

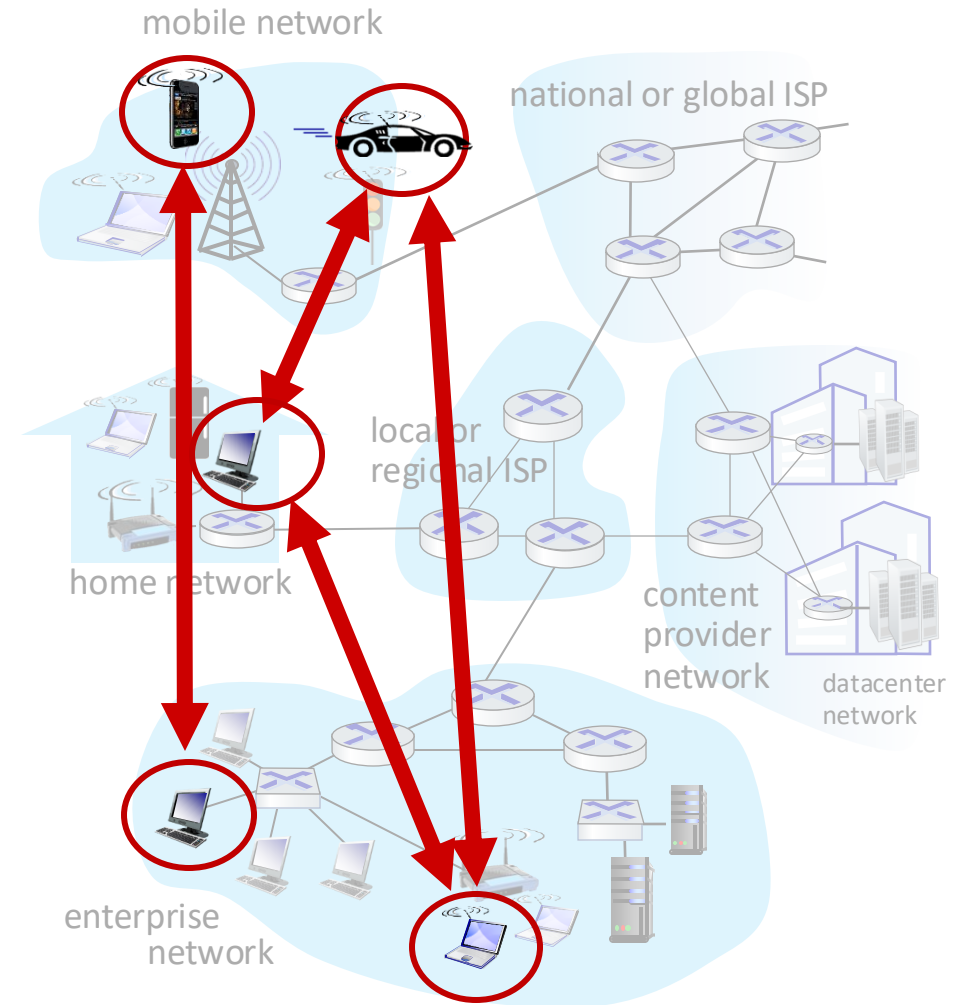
## clients:

- contact, communicate with server
- may be intermittently connected
- may have dynamic IP addresses
- do *not* communicate directly with each other
- examples: HTTP, IMAP, FTP



# Peer-peer architecture

- *no* always-on server
- arbitrary end systems directly communicate
- peers request service from other peers, provide service in return to other peers
  - *self scalability* – new peers bring new service capacity, as well as new service demands
- peers are intermittently connected and change IP addresses
  - complex management
- example: P2P file sharing



# Processes communicating

*process*: program running within a host

- within same host, two processes communicate using **inter-process communication** (defined by OS)
- processes in different hosts communicate by exchanging **messages**

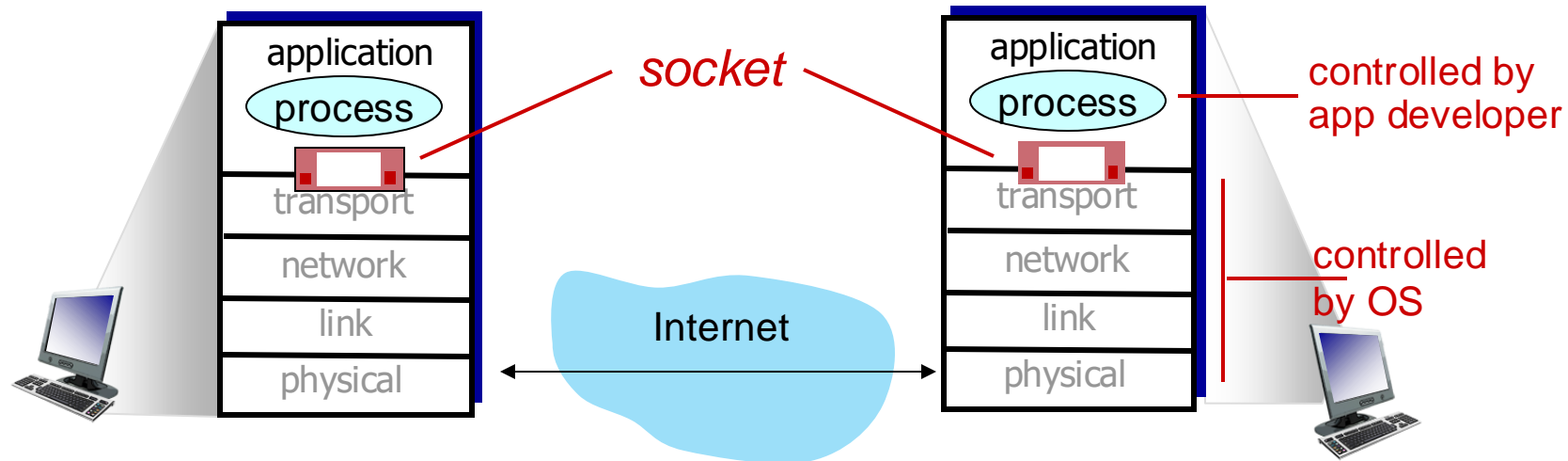
clients, servers

*client process*: process that initiates communication

*server process*: process that waits to be contacted

# 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
  - two sockets involved: one on each side





# Addressing processes

- to receive messages, process must have *identifier*
- host device has unique 32-bit IPv4 address (128-bit for IPv6)
- 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

# An application-layer protocol defines:

- **types of messages exchanged**,
  - e.g., request, response
- **message syntax**:
  - what fields in messages & how fields are delineated
- **message semantics**
  - meaning of information in fields
- **rules** for when and how processes send & respond to messages

## **open protocols:**

- defined in RFCs, everyone has access to protocol definition
- allows for interoperability
- e.g., HTTP, SMTP

## **proprietary protocols:**

- e.g., Skype

# What transport service does an app need?

## data integrity

- some apps (e.g., file transfer, web transactions) require 100% reliable data transfer
- other apps (e.g., audio) can tolerate some loss

## timing

- some apps (e.g., Internet telephony, interactive games) require low delay to be “effective”

## throughput

- some apps (e.g., multimedia) require minimum amount of throughput to be “effective”
- other apps (“elastic apps”) make use of whatever throughput they get

## security

- encryption, data integrity, ...

# Transport service requirements: common apps

application	data loss	throughput	time sensitive?
file transfer/download	no loss	elastic	no
e-mail	no loss	elastic	no
Web documents	no loss	elastic	no
real-time audio/video	loss-tolerant	audio: 5Kbps-1Mbps video:10Kbps-5Mbps	yes, 10's msec
streaming audio/video	loss-tolerant	same as above	yes, few secs
interactive games	loss-tolerant	Kbps+	yes, 10's msec
text messaging	no loss	elastic	yes and no

# 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 transport protocols services

<b>application</b>	<b>application layer protocol</b>	<b>transport protocol</b>
file transfer/download	FTP [RFC 959]	TCP
e-mail	SMTP [RFC 5321]	TCP
Web documents	HTTP 1.1 [RFC 7320], HTTP2	TCP
Internet telephony	SIP [RFC 3261], RTP [RFC 3550], or proprietary	TCP or UDP
streaming audio/video	HTTP [RFC 7320], DASH	TCP
interactive games	WOW, FPS (proprietary)	UDP or TCP

# Securing TCP

## Vanilla TCP & UDP sockets:

- no encryption
- cleartext passwords sent into socket traverse Internet in cleartext (!)

## Transport Layer Security (TLS)

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

## TSL implemented in application layer

- apps use TSL libraries, that use TCP in turn

## TLS socket API

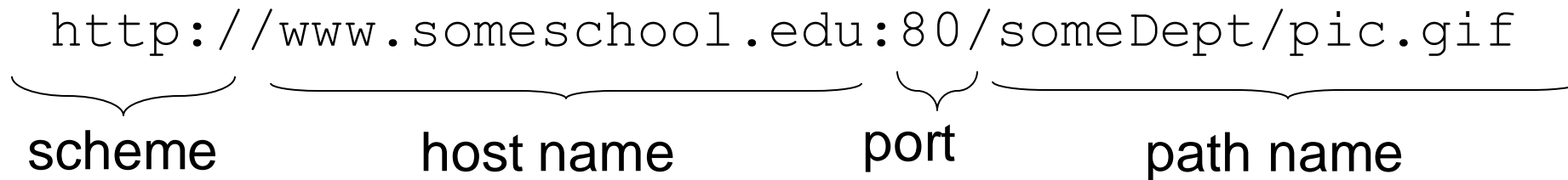
- cleartext sent into socket traverse Internet *encrypted*
- see Chapter 8

# Web and HTTP

*First, a quick review...*

- web page consists of *objects*, each of which can be stored on different Web servers
- object can be HTML file, JPEG image, JS scripts, videos....
- web page consists of *base HTML-file* which includes *several referenced objects, each* addressable by a *URL*, e.g.,

`http://www.someschool.edu:80/someDept/pic.gif`



scheme                      host name                      port                      path name

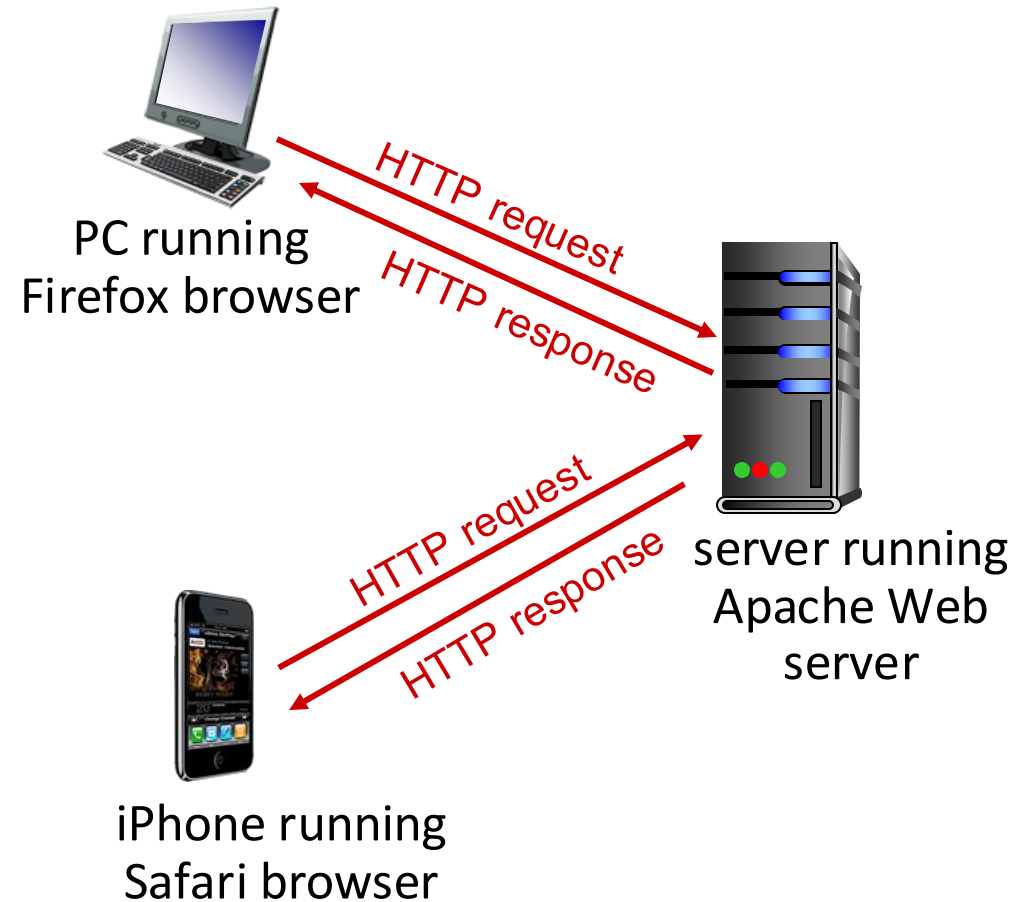
[https://www.amazon.com:443/store/sports/shoes/nike?\\_encoding=UTF8&\\_ref=dlx\\_gate\\_sd\\_dcl\\_tlt\\_1a650f5f\\_dt&pd\\_rd\\_w=vzrih&content-id=amzn1.sym.26a365d6](https://www.amazon.com:443/store/sports/shoes/nike?_encoding=UTF8&_ref=dlx_gate_sd_dcl_tlt_1a650f5f_dt&pd_rd_w=vzrih&content-id=amzn1.sym.26a365d6)



# HTTP overview

## HTTP: hypertext transfer protocol

- Web's application layer protocol
- client/server model:
  - *client*: browser that requests, receives, (using HTTP protocol) and “displays” Web objects
  - *server*: Web server sends (using HTTP protocol) objects in response to requests



# HTTP overview (continued)

## *HTTP uses TCP:*

- client initiates TCP connection (creates socket) to server, port 80
- server accepts TCP connection from client
- HTTP messages (application-layer protocol messages) exchanged between browser (HTTP client) and Web server (HTTP server)
- TCP connection closed

## *HTTP is “stateless”*

- server maintains *no* information about past client requests

*aside*  
protocols that maintain “state” are complex!

- past history (state) must be maintained
- if server/client crashes, their views of “state” may be inconsistent, must be reconciled

# HTTP connections: two types

## *Non-persistent HTTP*

1. TCP connection opened
2. at most one object sent over TCP connection
3. TCP connection closed

downloading multiple objects required multiple connections

## *Persistent HTTP*

- TCP connection opened to a server
- multiple objects can be sent over *single* TCP connection between client, and that server
- TCP connection closed

# Non-persistent HTTP: example

User enters URL: `www.someSchool.edu/someDepartment/home.index`  
(containing text, references to 10 jpeg images)



**1a.** HTTP client initiates TCP connection to HTTP server (process) at `www.someSchool.edu` on port 80



**1b.** HTTP server at host `www.someSchool.edu` waiting for TCP connection at port 80 “accepts” connection, notifying client

**2.** HTTP client sends HTTP *request message* (containing URL) into TCP connection socket. Message indicates that client wants object `someDepartment/home.index`

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

time



# Non-persistent HTTP: example (cont.)

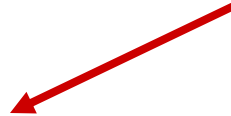
User enters URL: `www.someSchool.edu/someDepartment/home.index`  
(containing text, references to 10 jpeg images)



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

4. HTTP server closes TCP connection.

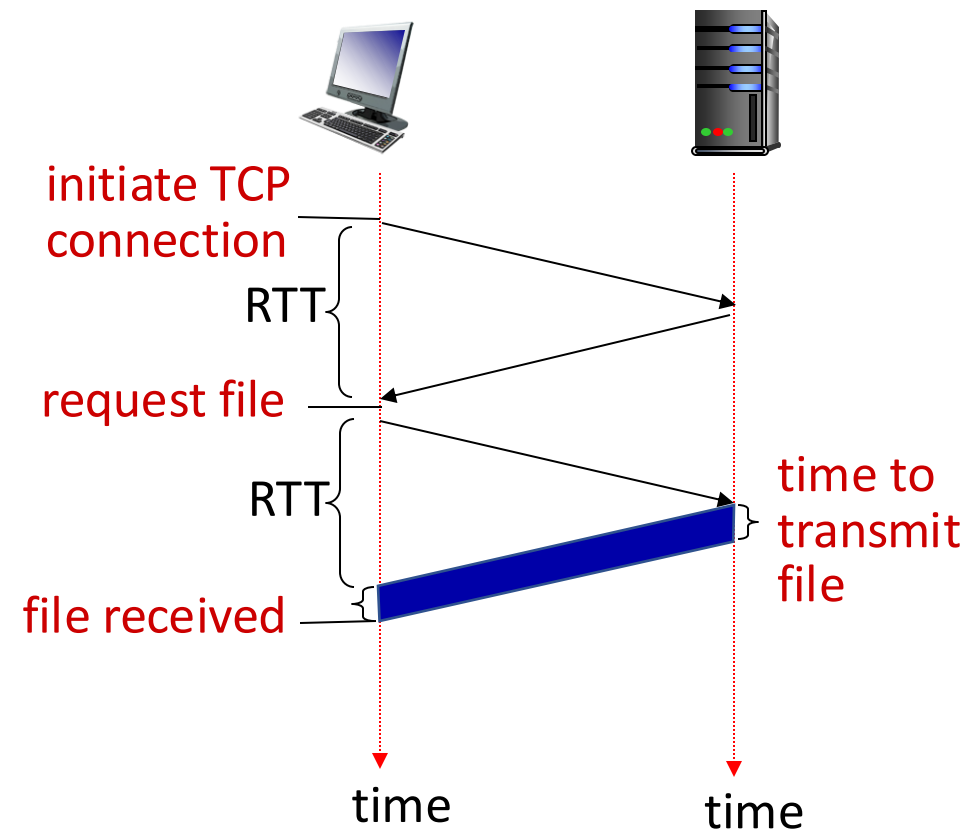


# Non-persistent HTTP: response time

**RTT (definition):** time for a small packet to travel from client to server and back

**HTTP response time (per object):**

- one RTT to initiate TCP connection
- one RTT for HTTP request and first few bytes of HTTP response to return
- object/file transmission time



*Non-persistent HTTP response time =  $2RTT + \text{file transmission time}$*

# Persistent HTTP (HTTP 1.1)

## *Non-persistent HTTP issues:*

- requires 2 RTTs per object
- OS overhead for *each* TCP connection
- browsers often open multiple parallel TCP connections to fetch referenced objects in parallel

## *Persistent HTTP (HTTP1.1):*

- server leaves connection open after sending response
- subsequent HTTP messages between same client/server sent over open connection
- client sends requests as soon as it encounters a referenced object
- as little as one RTT for all the referenced objects (cutting response time in half)

# HTTP request message

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

request line (GET, POST, HEAD commands) →

header lines

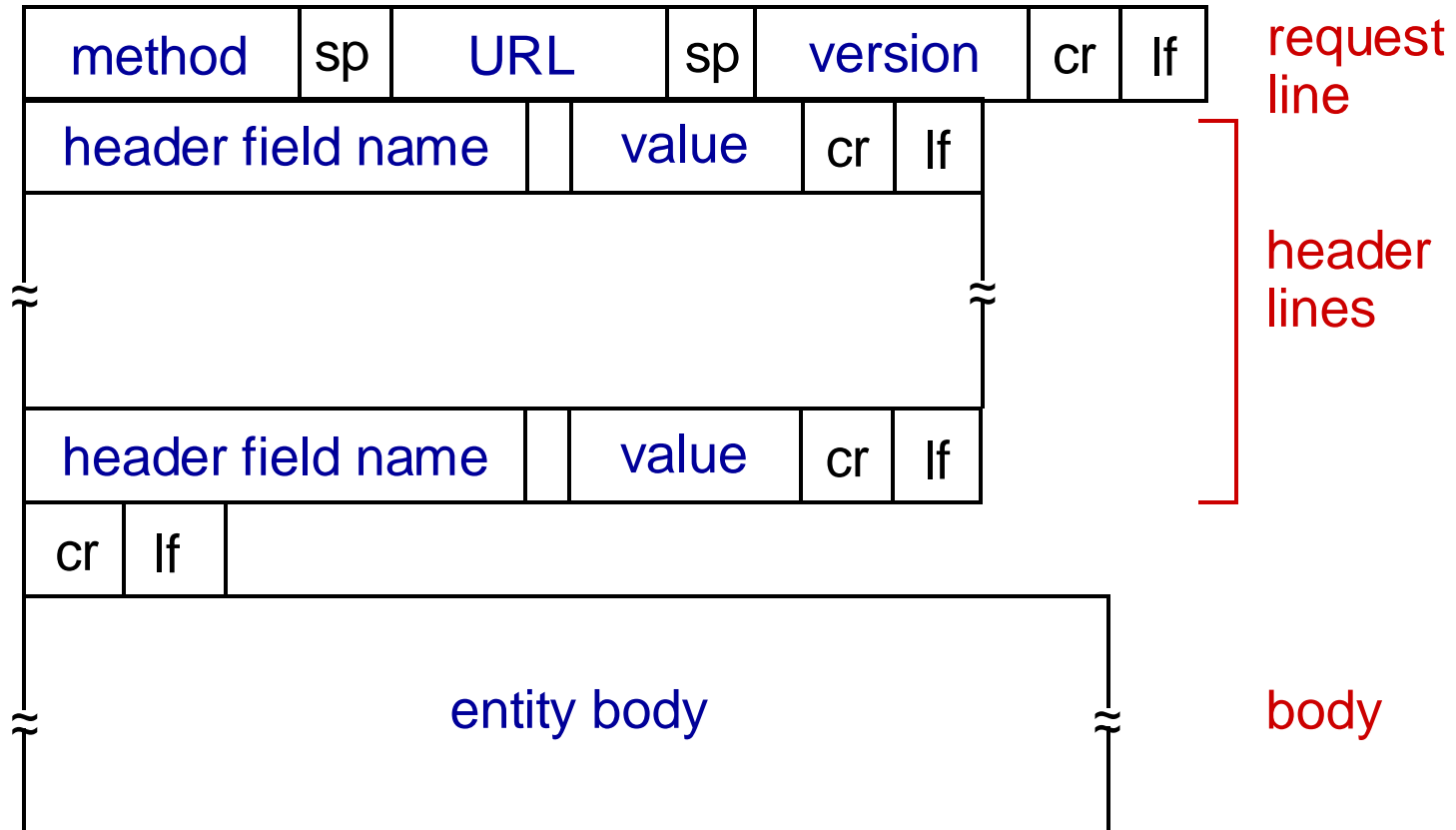
```
GET /index.html HTTP/1.1\r\n
Host: www-net.cs.umass.edu\r\n
User-Agent: Firefox/3.6.10\r\n
Accept: text/html,application/xhtml+xml\r\n
Accept-Language: en-us,en;q=0.5\r\n
Accept-Encoding: gzip,deflate\r\n
Accept-Charset: ISO-8859-1,utf-8;q=0.7\r\n
Keep-Alive: 115\r\n
Connection: keep-alive\r\n
\r\n
```

carriage return character  
line-feed character

carriage return, line feed →  
at start of line indicates  
end of header lines



# HTTP request message: general format



# Other HTTP request messages

## POST method:

- web page often includes form input
- user input sent from client to server in entity body of HTTP POST request message

## GET method (for sending data to server):

- include user data in URL field of HTTP GET request message (following a '?'):

`www.somesite.com/animalsearch?monkeys&banana`

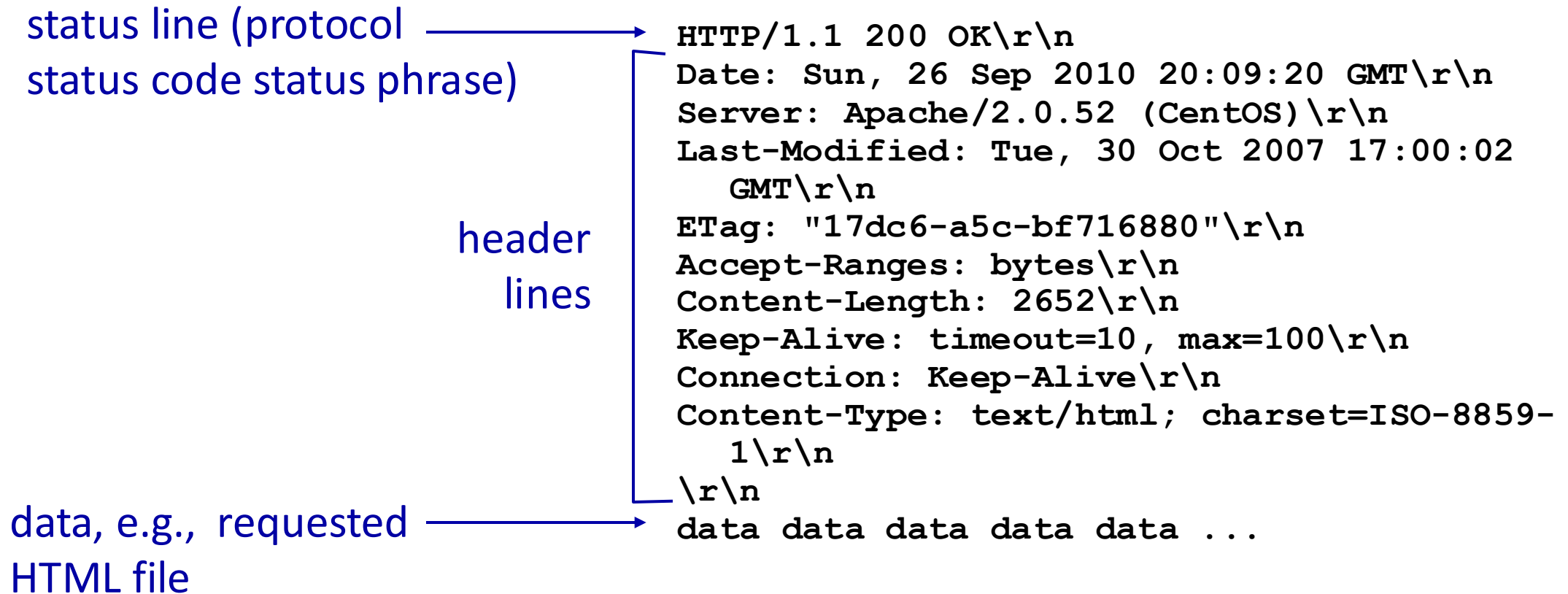
## HEAD method:

- requests headers (only) that would be returned *if* specified URL were requested with an HTTP GET method.

## PUT method:

- uploads new file (object) to server
- completely replaces file that exists at specified URL with content in entity body of POST HTTP request message

# HTTP response message



\* Check out the online interactive exercises for more examples: [http://gaia.cs.umass.edu/kurose\\_ross/interactive/](http://gaia.cs.umass.edu/kurose_ross/interactive/)

# HTTP response status codes

- status code appears in 1st line in server-to-client response message.
- some sample codes:

## 200 OK

- request succeeded, requested object later in this message

## 301 Moved Permanently

- requested object moved, new location specified later in this message (in Location: field)

## 400 Bad Request

- request msg not understood by server

## 404 Not Found

- requested document not found on this server

## 505 HTTP Version Not Supported

# Trying out HTTP (client side) for yourself

1. Telnet to your favorite Web server:

```
telnet gaia.cs.umass.edu 80
```

- opens TCP connection to port 80 (default HTTP server port) at gaia.cs.umass.edu.
- anything typed in will be sent to port 80 at gaia.cs.umass.edu

2. type in a GET HTTP request:

```
GET /kurose_ross/interactive/index.php HTTP/1.1  
Host: gaia.cs.umass.edu
```

- by typing this in (hit carriage return twice), you send this minimal (but complete) GET request to HTTP server

3. look at response message sent by HTTP server!

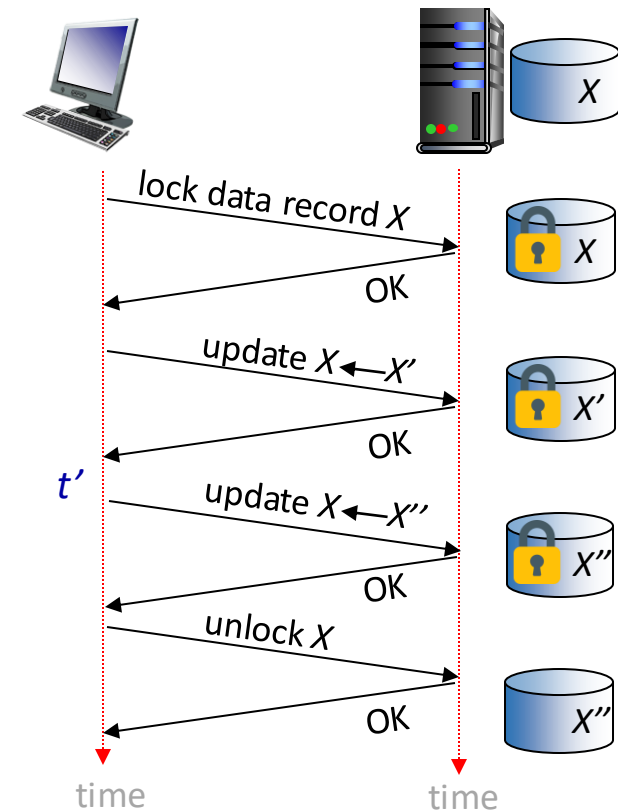
(or use Wireshark to look at captured HTTP request/response)

# Maintaining user/server state: cookies

Recall: HTTP GET/response interaction is *stateless*

- no notion of multi-step exchanges of HTTP messages to complete a Web “transaction”
  - no need for client/server to track “state” of multi-step exchange
  - all HTTP requests are independent of each other
  - no need for client/server to “recover” from a partially-completed-but-never-completely-completed transaction

a *stateful protocol*: client makes two changes to  $X$ , or none at all



**Q:** what happens if network connection or client crashes at  $t'$ ?

# Maintaining user/server state: cookies [[RFC6265](#)]

Web sites and client browser use *cookies* to maintain some state between transactions

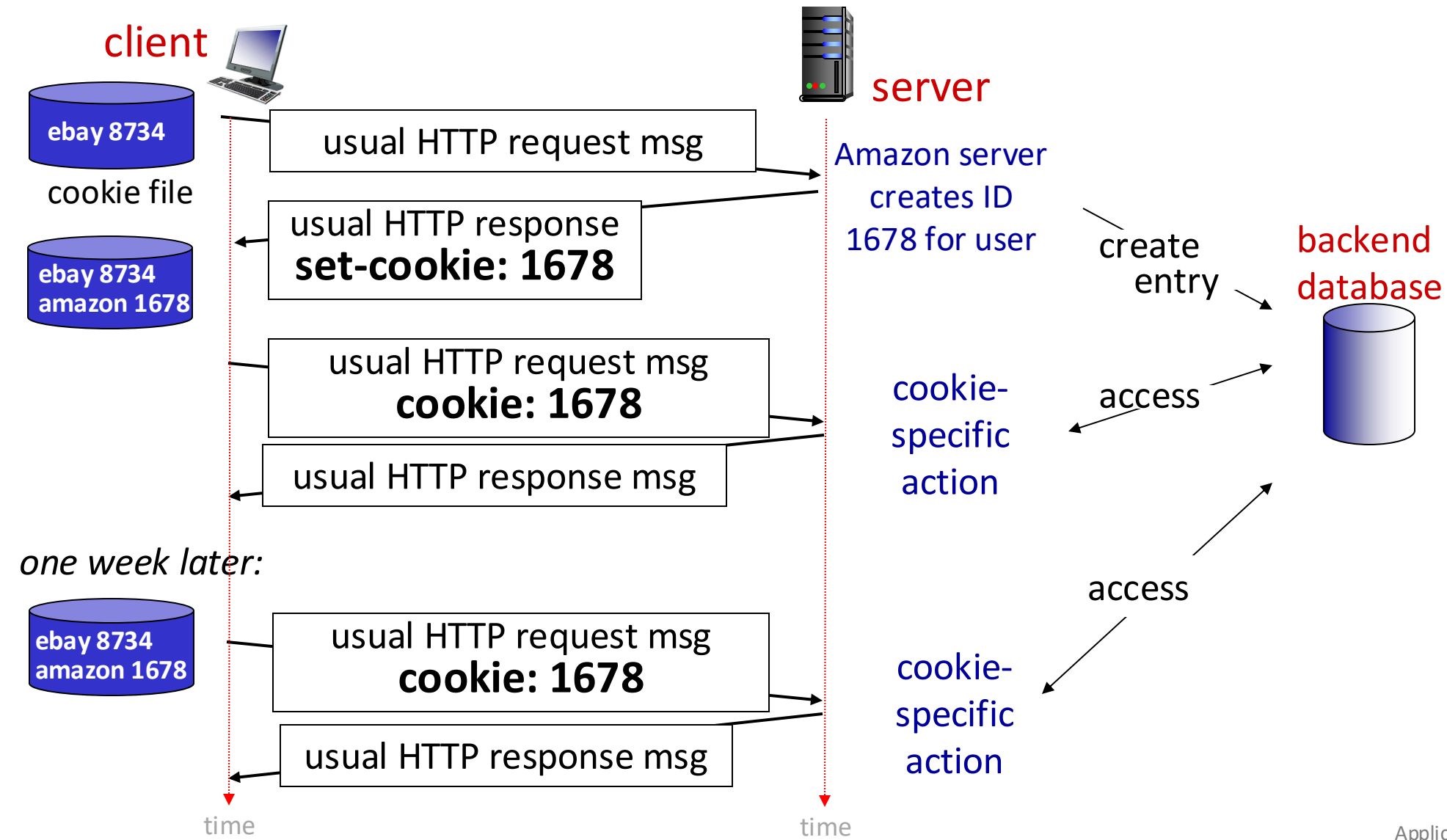
## *four components:*

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

## Example:

- Susan uses browser on laptop, visits specific e-commerce site for first time
- when initial HTTP requests arrives at site, site creates:
  - unique ID (aka “cookie”)
  - entry in backend database for ID
- subsequent HTTP requests from Susan to this site will contain cookie ID value, allowing site to “identify” Susan

# Maintaining user/server state: cookies





# HTTP cookies: attributes

- Expires
  - the date and time at which the cookie expires
- Max-age
  - number of seconds until the cookie expires
- Domain
  - specifies hosts to which the cookie will be sent
- Path
  - limits the scope of a cookie to a given (set of) path(s)
- Secure
  - Cookies marked as "Secure" are only sent over HTTPS
- HttpOnly
  - HttpOnly cookies cannot be accessed directly by JS code

# HTTP cookies

- Different types of cookies
  - Session vs. persistent
  - First-party vs. third-party
- Third-party cookies (see <http://tools.ietf.org/html/rfc2965>)
  - You visit [www.example.com](http://www.example.com), which contains a banner from ads.clicks-for-me.net
    - in simple terms ads.clicks-for-me.net is third-party because it does not match the domain showed on the URL bar
    - third-party sites should be denied setting or reading cookies
  - The browser allows ads.clicks-for-me.net to drop a third-party cookie
  - Then you visit [www.another-example.com](http://www.another-example.com) , which also loads ads from ads.clicks-for-me.net
  - ads.clicks-for-me.net can track the fact that you visited both [www.example.com](http://www.example.com) and [www.another-example.com](http://www.another-example.com) !!!

# HTTP cookies and security

- Authentication Cookies can be stolen
  - Without TLS, an attacker may be able to “sniff” your authentication cookies
  - The attacker will be able to login as you on a website (e.g., Facebook, Twitter, etc...)
- See FireSheep for a concrete example!
  - <http://codebutler.com/firesheep>

# Session IDs

- Cookies are not the only way you can keep state
  - Session IDs are commonly used by web applications
    - [http://example.com/index.php?user\\_id=0F4C26A1&topic=networking](http://example.com/index.php?user_id=0F4C26A1&topic=networking)
- What are the main difference between cookies and Session IDs?
  - Session IDs are typically passed in the URL (added to web app links)
  - Cookies are passed through HTTP req/resp headers
  - Cookies are stored in the browser's cache and have an expiration date
  - Session IDs are volatile: never stored, only used until end of session

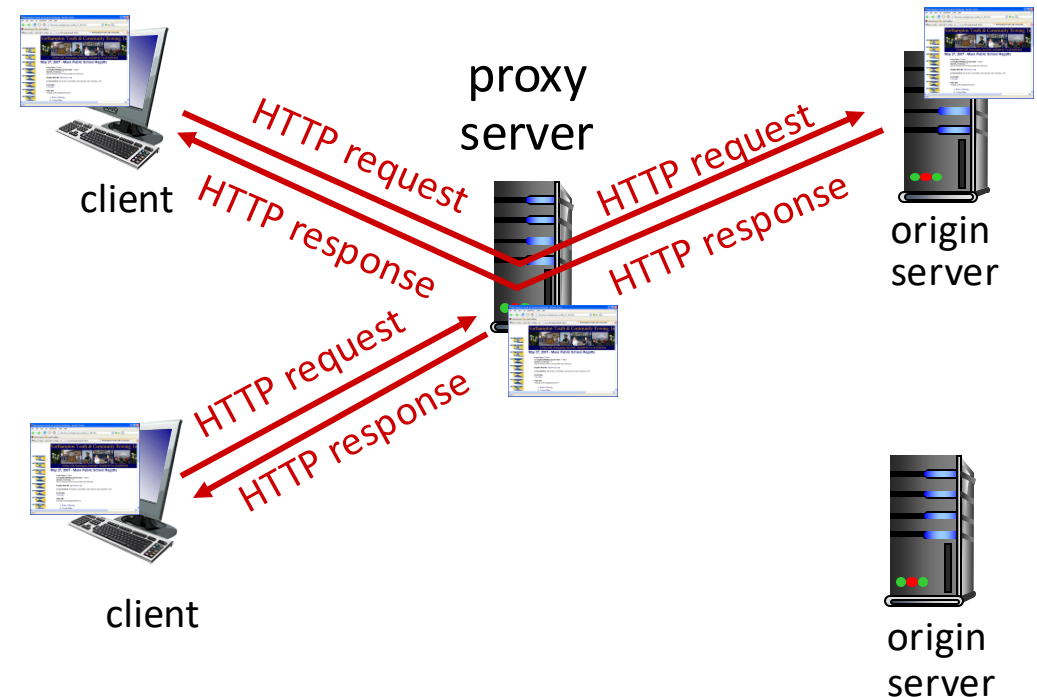
# Analyzing in-browser HTTP requests

- Demo time...
- Use DevTools to investigate how web objects needed to render a web page are retrieve in Chrome
- Understanding the difference between Sites and Origins:
  - <https://web.dev/same-site-same-origin/>

# Web caches (proxy servers)

*Goal:* satisfy client request without involving origin server

- user configures browser to point to a *Web cache*
- browser sends all HTTP requests to cache
  - *if* object in cache: cache returns object to client
  - *else* cache requests object from origin server, caches received object, then returns object to client



[RFC7234] - HTTP/1.1 Caching: <https://tools.ietf.org/html/rfc7234>

# Web caches (proxy servers)

- Web cache acts as both client and server
  - server for original requesting client
  - client to origin server
- typically cache is installed by ISP (university, company, residential ISP)

## *Why* Web caching?

- reduce response time for client request
  - cache is closer to client
- reduce traffic on an institution's access link
- Internet is dense with caches
  - enables “poor” content providers to more effectively deliver content

# Caching example

## Scenario:

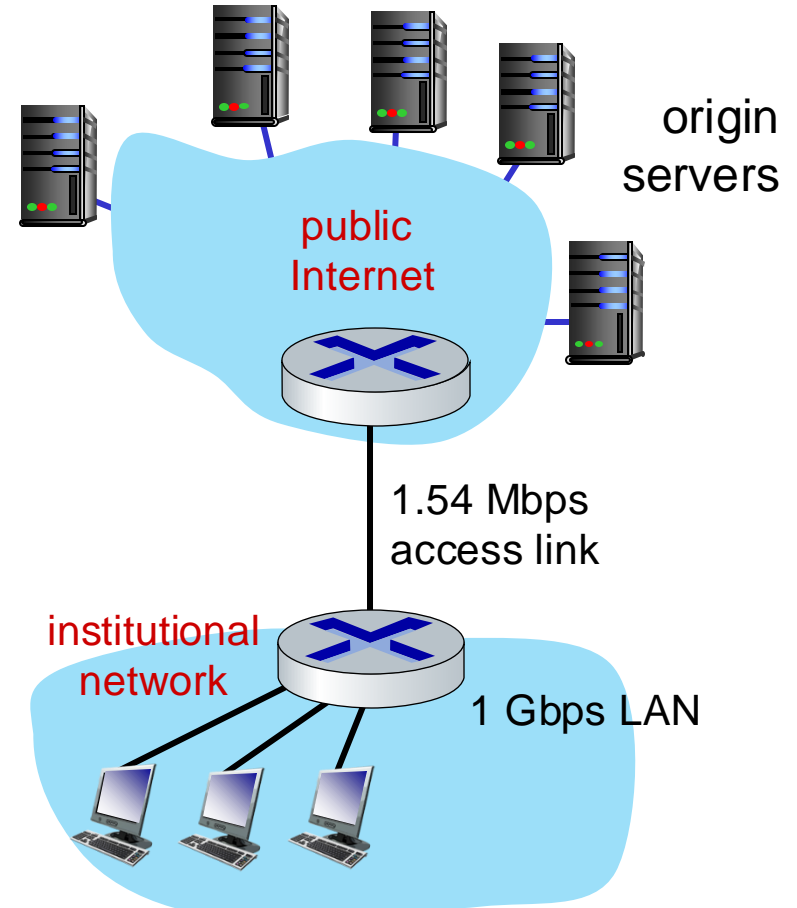
- access link rate: 1.54 Mbps
- RTT from institutional router to server: 2 sec
- Web object size: 100K bits
- Average request rate from browsers to origin servers: 15/sec
  - average data rate to browsers: 1.50 Mbps

## Performance:

- LAN utilization: .0015
- access link utilization = .97
- end-end delay = Internet delay + access link delay + LAN delay  
= 2 sec + minutes + usecs

Due to traffic intensity  $\approx 1$   
on the access link

problem: large  
delays at high  
utilization!





# Caching example: buy a faster access link

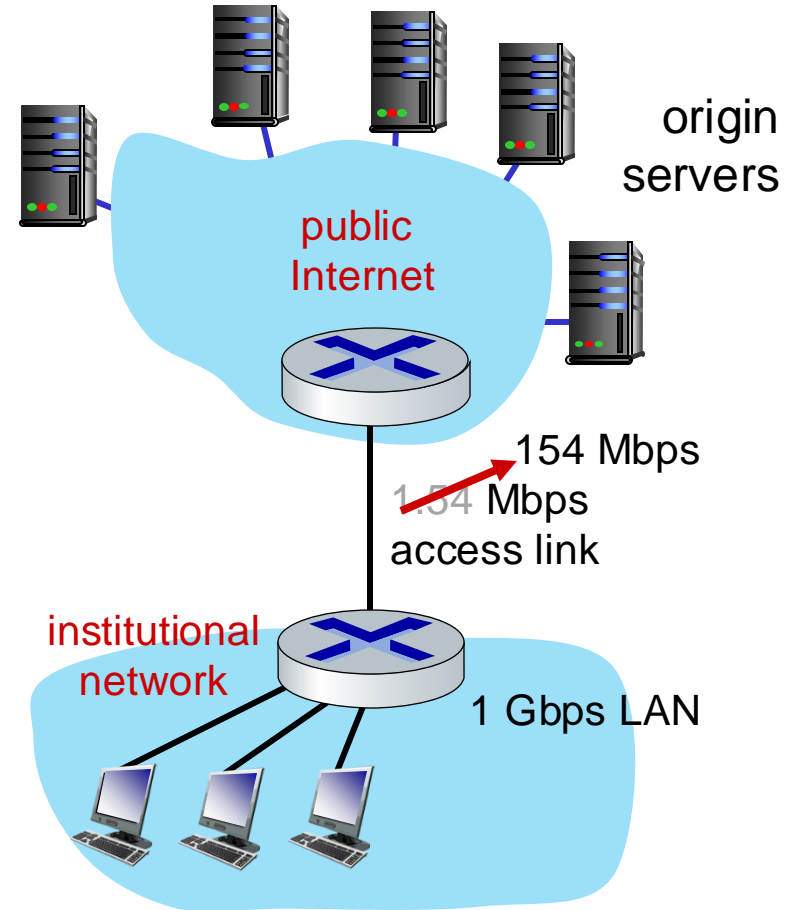
## Scenario:

- access link rate: ~~1.54 Mbps~~ <sup>154 Mbps</sup>
- RTT from institutional router to server: 2 sec
- Web object size: 100K bits
- Avg request rate from browsers to origin servers: 15/sec
  - avg data rate to browsers: 1.50 Mbps

## Performance:

- LAN utilization: .0015
- access link utilization = ~~.97~~ <sup>.0097</sup>
- end-end delay = Internet delay +  
access link delay + LAN delay  
= 2 sec + ~~minutes~~ + usecs

**Cost:** faster access link (expensive!) <sup>msecs</sup>



# Caching example: install a web cache

## Scenario:

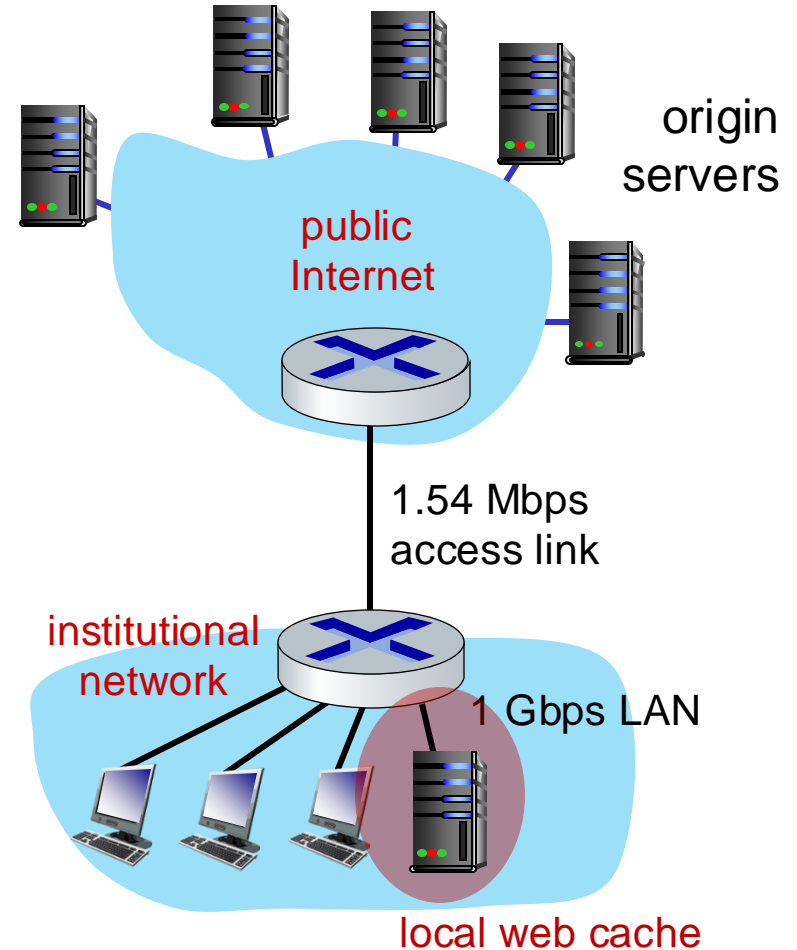
- access link rate: 1.54 Mbps
- RTT from institutional router to server: 2 sec
- Web object size: 100K bits
- Avg request rate from browsers to origin servers: 15/sec
  - avg data rate to browsers: 1.50 Mbps

## Performance:

- LAN utilization: .?
- access link utilization = ?
- average end-end delay = ?

*How to compute link utilization, delay?*

*Cost:* web cache (cheap!)

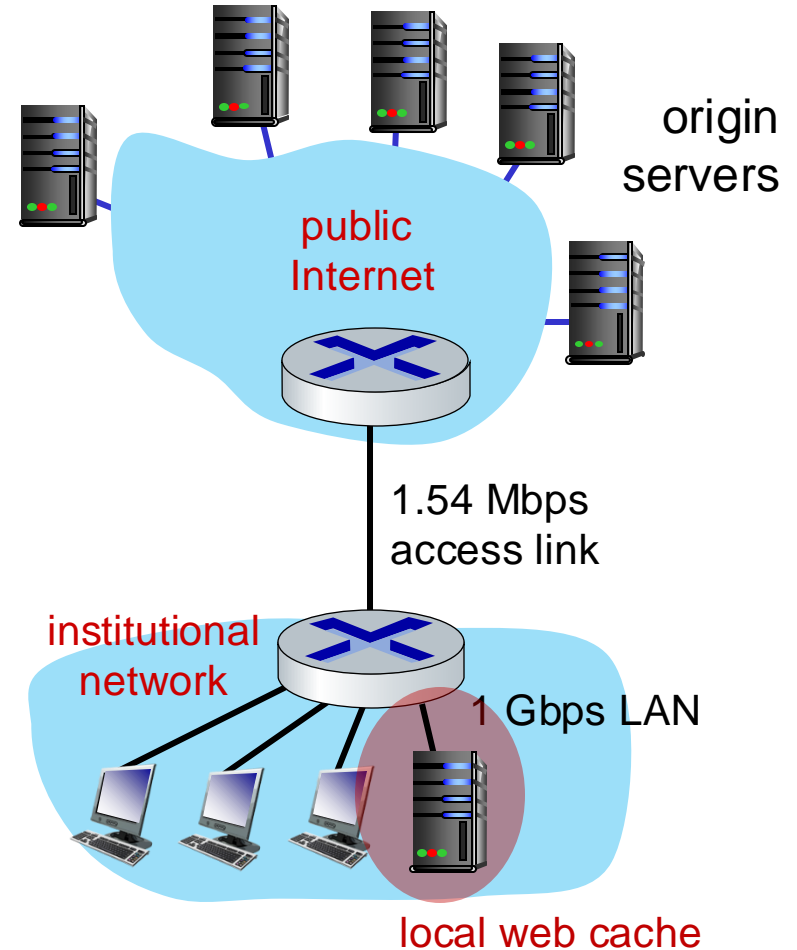


# Caching example: install a web cache

## Calculating access link utilization, end-end delay with cache:

- suppose cache hit rate is 0.4: 40% requests satisfied at cache, 60% requests satisfied at origin
- access link: 60% of requests use access link
- data rate to browsers over access link  
 $= 0.6 * 1.50 \text{ Mbps} = .9 \text{ Mbps}$
- utilization  $= 0.9 / 1.54 = .58$
- average end-end delay  
 $= 0.6 * (\text{delay from origin servers})$   
 $+ 0.4 * (\text{delay when satisfied at cache})$   
 $= 0.6 (\sim 2 \text{ sec}) + 0.4 (\sim \text{msecs}) = \sim 1.2 \text{ secs}$

*lower average end-end delay than with 154 Mbps link (and cheaper too!)*



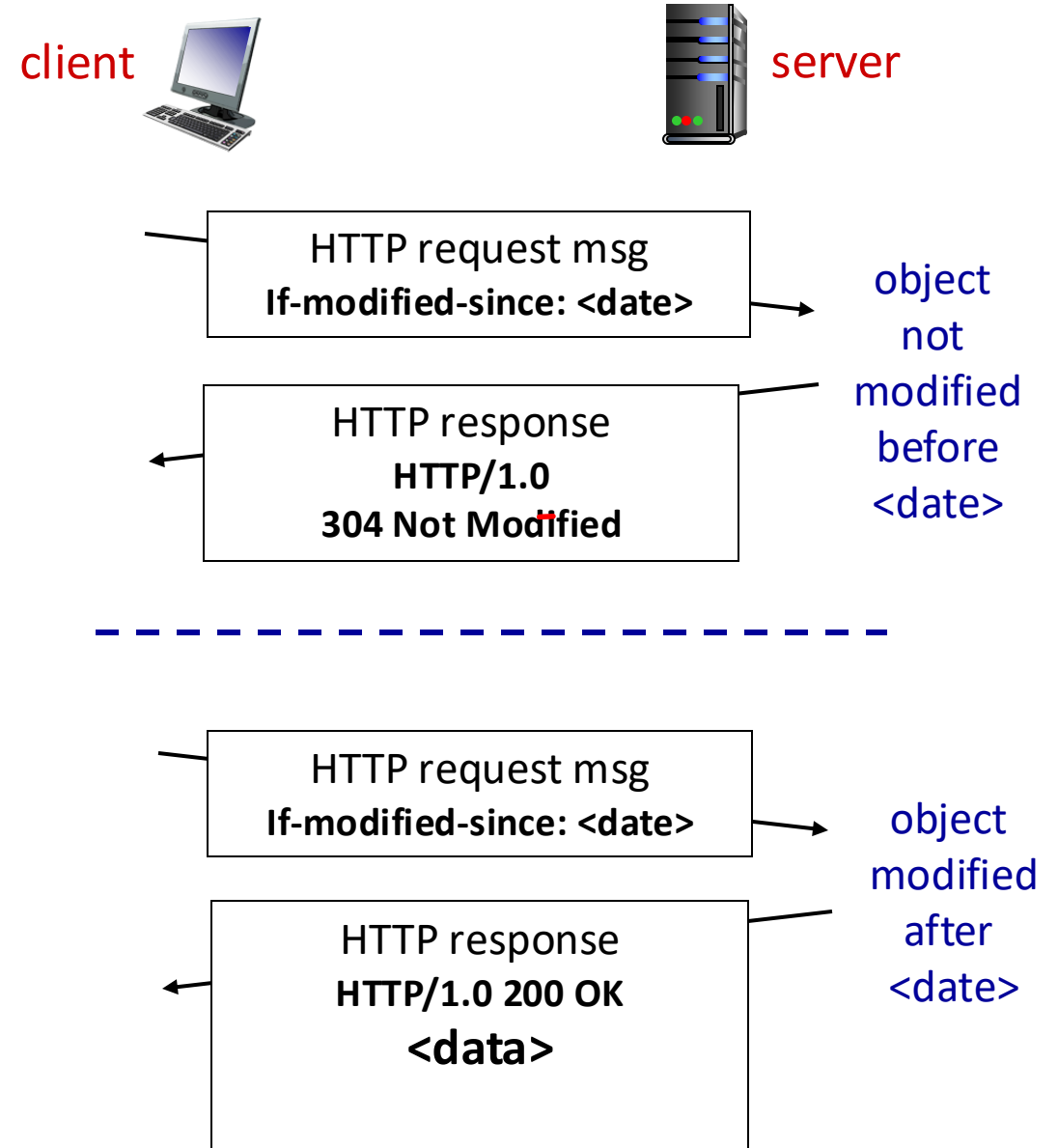
# Conditional GET

**Goal:** don't send object if cache has up-to-date cached version

- no object transmission delay
- lower link utilization
- **cache:** specify date of cached copy in HTTP request  
**If-modified-since: <date>**
- **server:** response contains no object if cached copy is up-to-date:  
**HTTP/1.0 304 Not Modified**

[RFC7232] - HTTP/1.1 Conditional Requests

[RFC7234] - HTTP/1.1 Caching



# HTTP Pipelining and Range

## ■ Pipelining

- The client sends multiple HTTP request without waiting for server response
- The server sends the response one after the other

## ■ Range

- HTTP allows downloading pieces of objects
- Example:
  - 10MB image to be downloaded
  - We can open 10 different TCP connection and send 10 HTTP requests in parallel
  - Download 1MB of data from each connection and stitch them back together

# HTTP/2

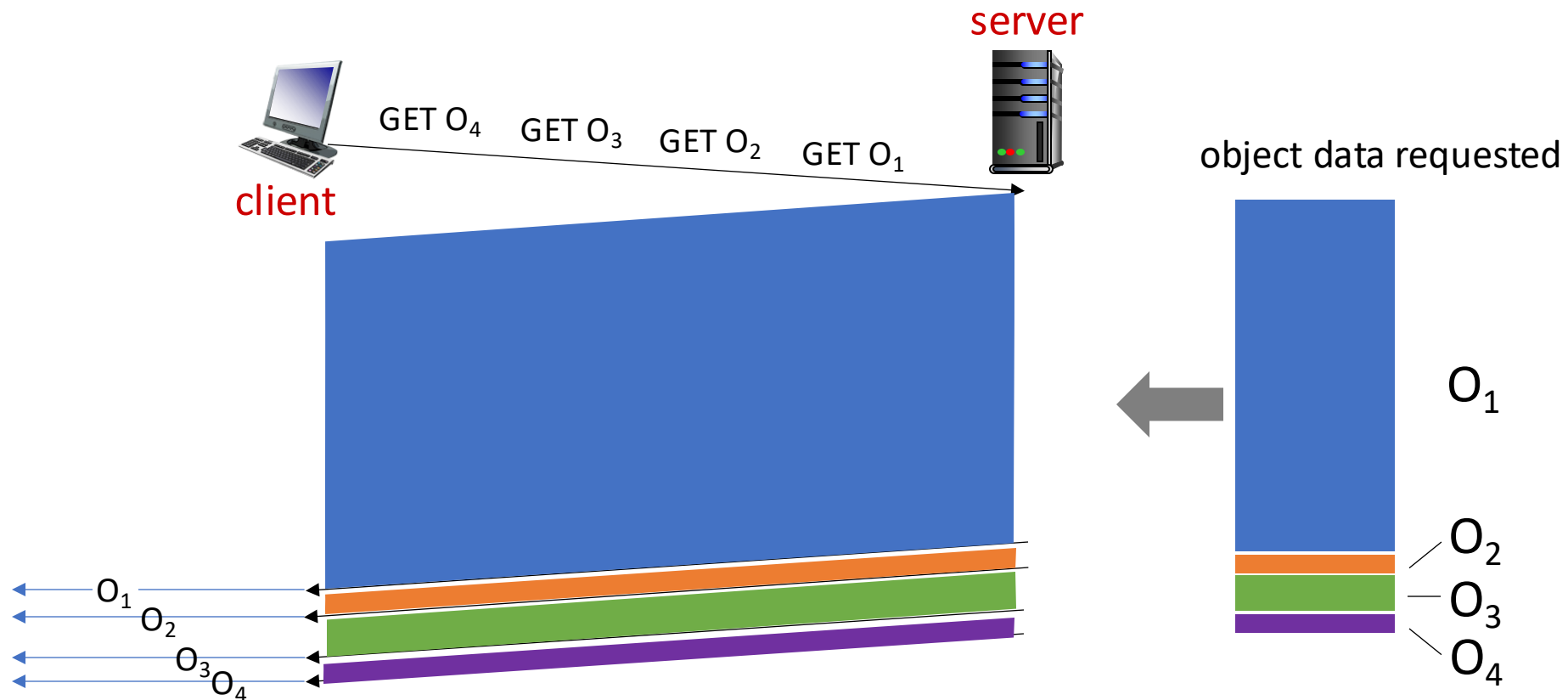
*Key goal:* decreased delay in multi-object HTTP requests

HTTP1.1: introduced **multiple, pipelined GETs** over single TCP connection

- server responds *in-order* (FCFS: first-come-first-served scheduling) to GET requests
- with FCFS, small object may have to wait for transmission (**head-of-line (HOL) blocking**) behind large object(s)
- loss recovery (retransmitting lost TCP segments) stalls object transmission

# HTTP 1.1 pipelining: HOL blocking

Client requests 1 large object (e.g., large image), and 3 smaller objects



*objects delivered in order requested: `O2`, `O3`, `O4` wait behind `O1`*

# HTTP/2

*Key goal:* decreased delay in multi-object HTTP requests

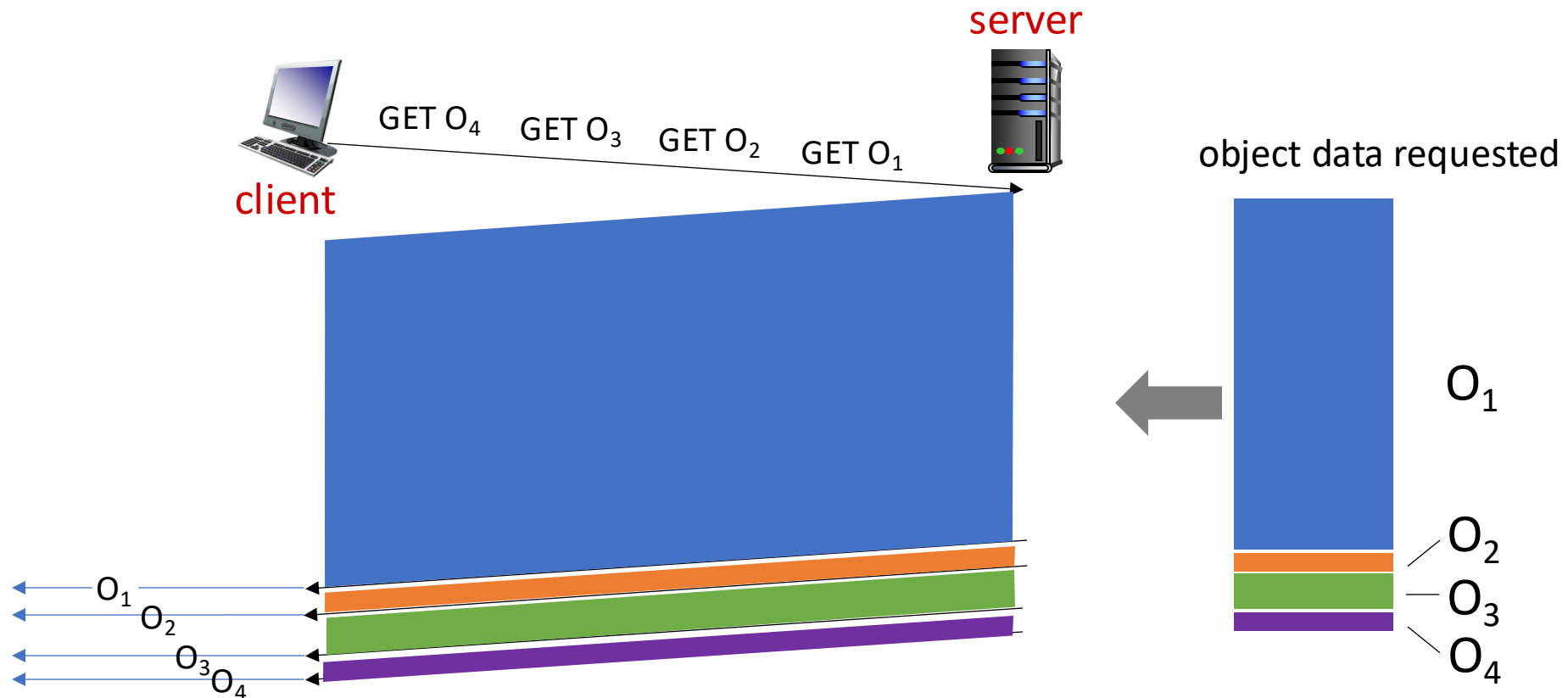
HTTP/2: [RFC 7540, 2015] increased flexibility at *server* in sending objects to client:

- methods, status codes, most header fields unchanged from HTTP 1.1
- transmission order of requested objects based on client-specified object priority (not necessarily FCFS)
- *push* unrequested objects to client
- divide objects into frames, schedule frames to mitigate HOL blocking



# HTTP/2: mitigating HOL blocking

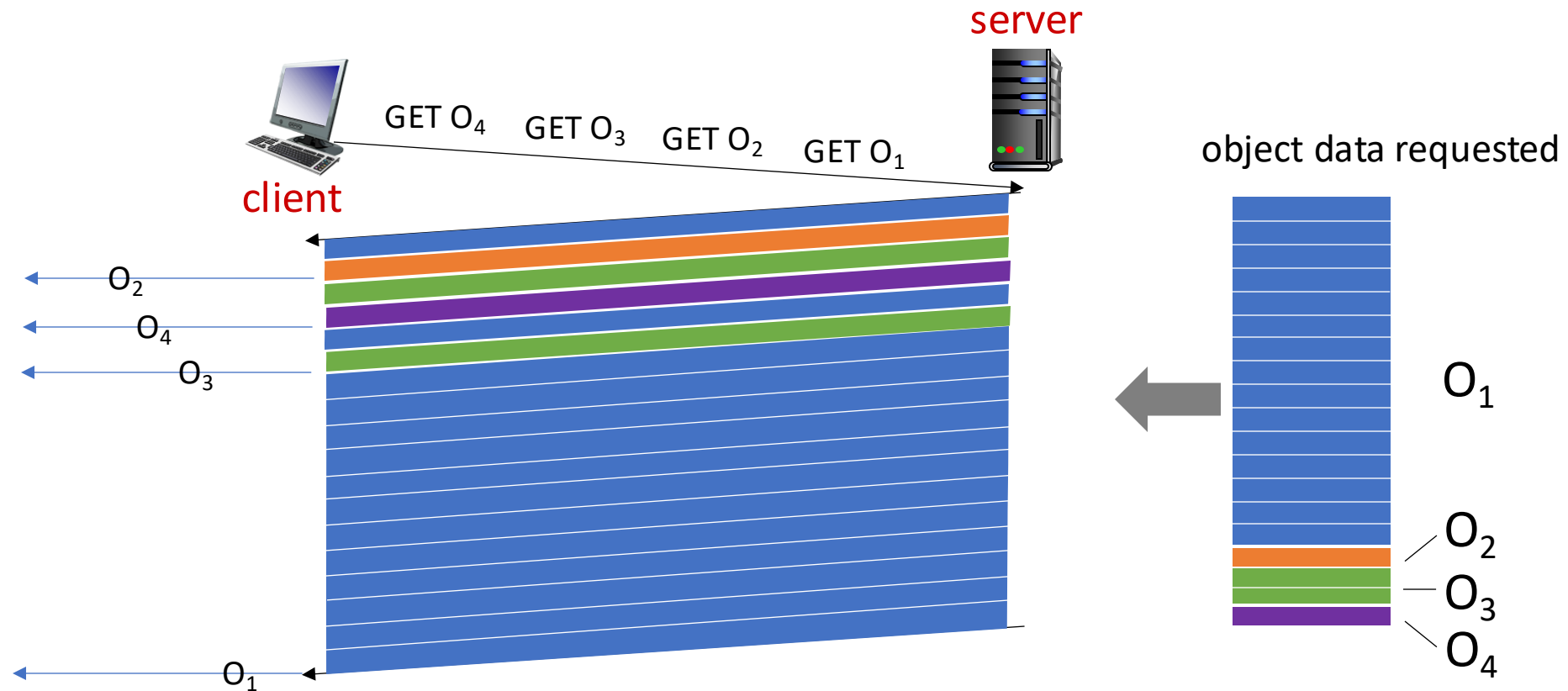
HTTP 1.1: requests for 1 large object and 3 smaller objects



*objects delivered in order requested:  $O_2$ ,  $O_3$ ,  $O_4$  wait behind  $O_1$*

# HTTP/2: mitigating HOL blocking

HTTP/2: objects divided into frames, frame transmission interleaved



*$O_2$ ,  $O_3$ ,  $O_4$  delivered quickly,  $O_1$  slightly delayed*

# Upgrading from HTTP/1.1 to HTTP/2

## Client

GET / HTTP/1.1

Host: server.example.com

Connection: Upgrade, HTTP2-Settings

Upgrade: h2c

HTTP2-Settings: <base64url encoding of HTTP/2 SETTINGS payload>

## Server

A server that does not support HTTP/2 can respond to the request as though the Upgrade header field were absent:

HTTP/1.1 200 OK

Content-Length: 243

Content-Type: text/html

...

## Server

A server that supports HTTP/2 accepts the upgrade with a 101 (Switching Protocols) response.

For example:

HTTP/1.1 101 Switching Protocols

Connection: Upgrade

Upgrade: h2c

[ HTTP/2 connection ...

RFC 7540: <https://tools.ietf.org/html/rfc7540>

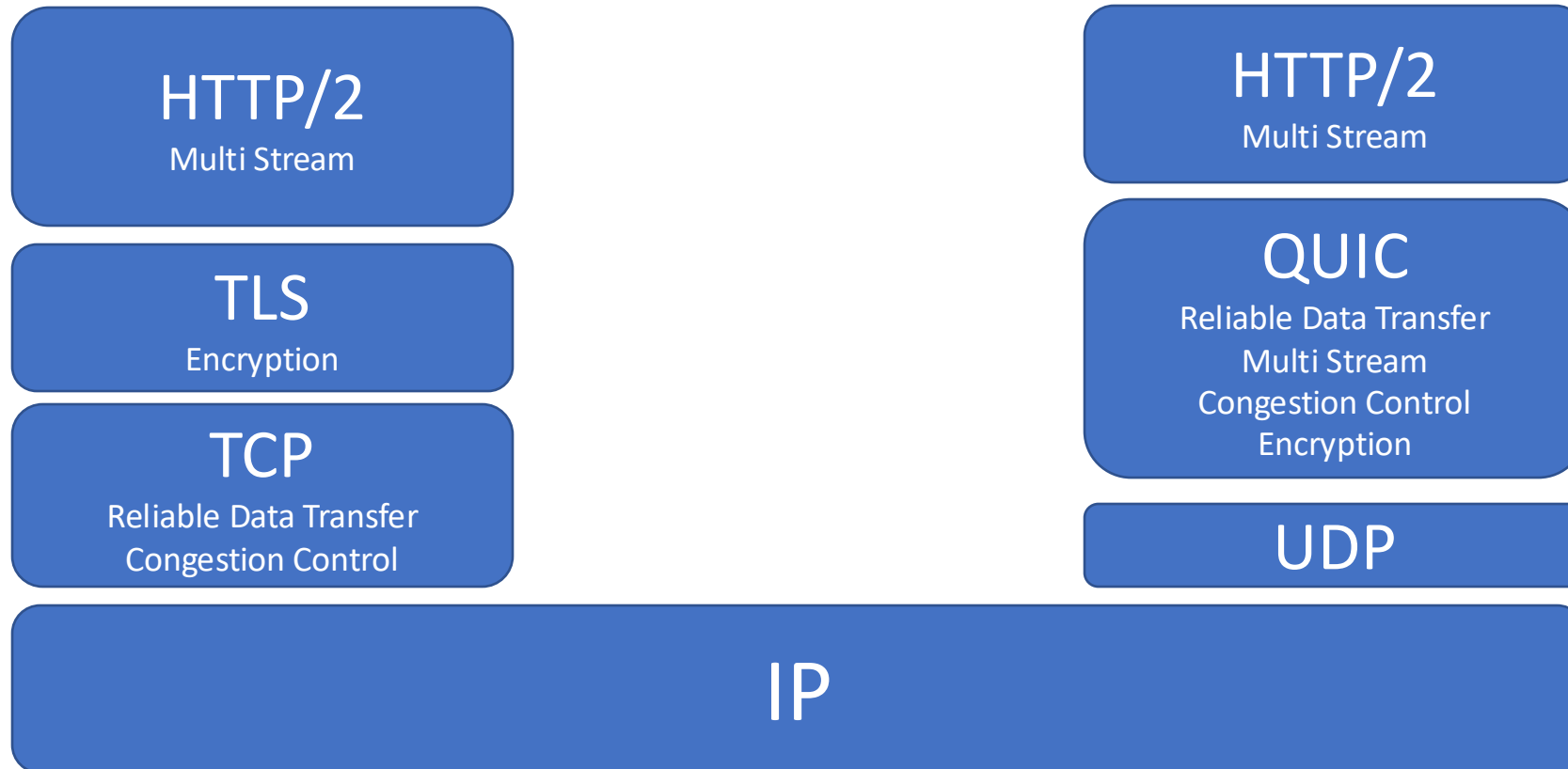
# HTTP/2 to HTTP/3

*Key goal:* decreased delay in multi-object HTTP requests

HTTP/2 over single TCP connection means:

- recovery from packet loss still stalls all object transmissions
  - as in HTTP 1.1, browsers have incentive to open multiple parallel TCP connections to reduce stalling, increase overall throughput
- no security over vanilla TCP connection
- **HTTP/3 (uses QUIC transport)** adds security, per object error- and congestion-control
  - (more pipelining) over UDP
  - more on HTTP/3 in transport layer

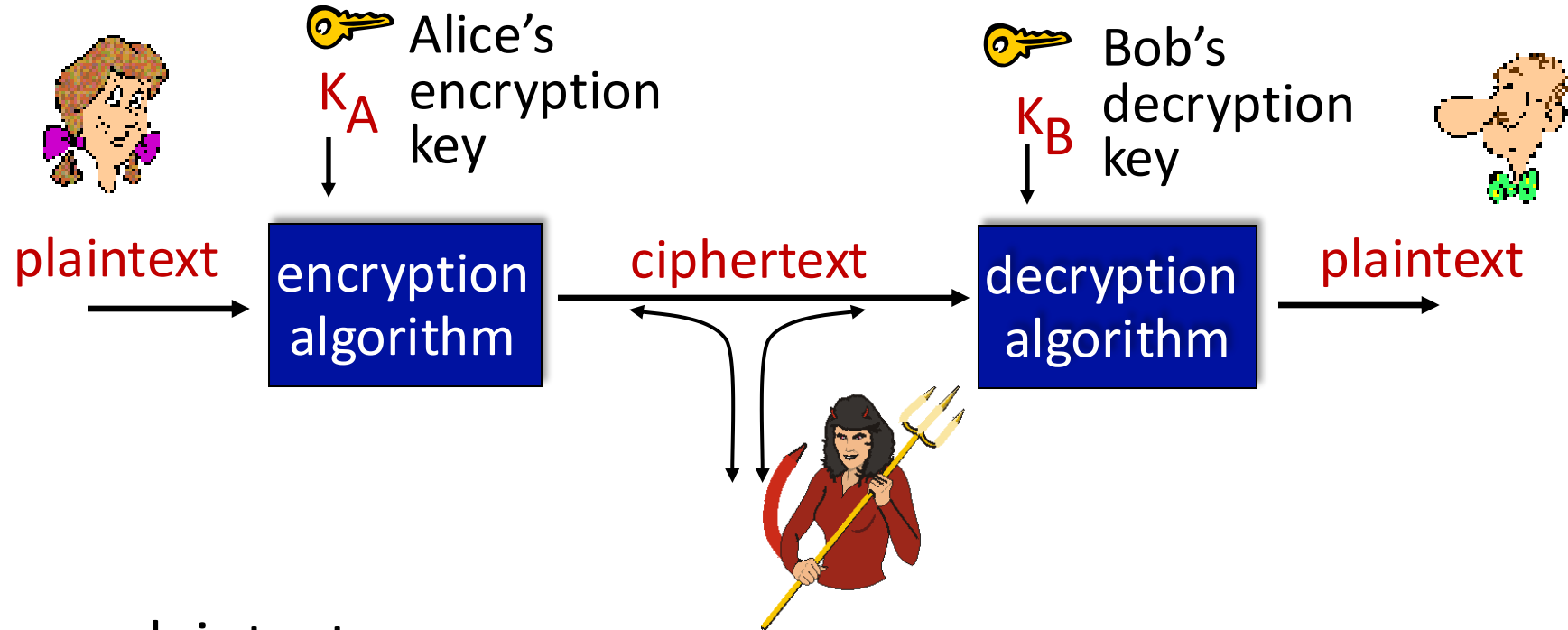
# HTTP/3 $\approx$ HTTP/2 over QUIC



# HTTPS: HTTP over TLS

- Client opens TCP connection towards Web Server (default port = 443)
- TLS session is established (highly simplified overview)
  - Use established TCP connection
  - Client sends *hello* message asking to use TLS
  - Server sends digital certificate to verify its identity
  - Client Server exchange information needed to compute cryptographic keys
  - All data exchanged through TLS session is encrypted
- HTTP request/responses are passed through the “TLS-wrapped” TCP connection

# Cryptography Concepts

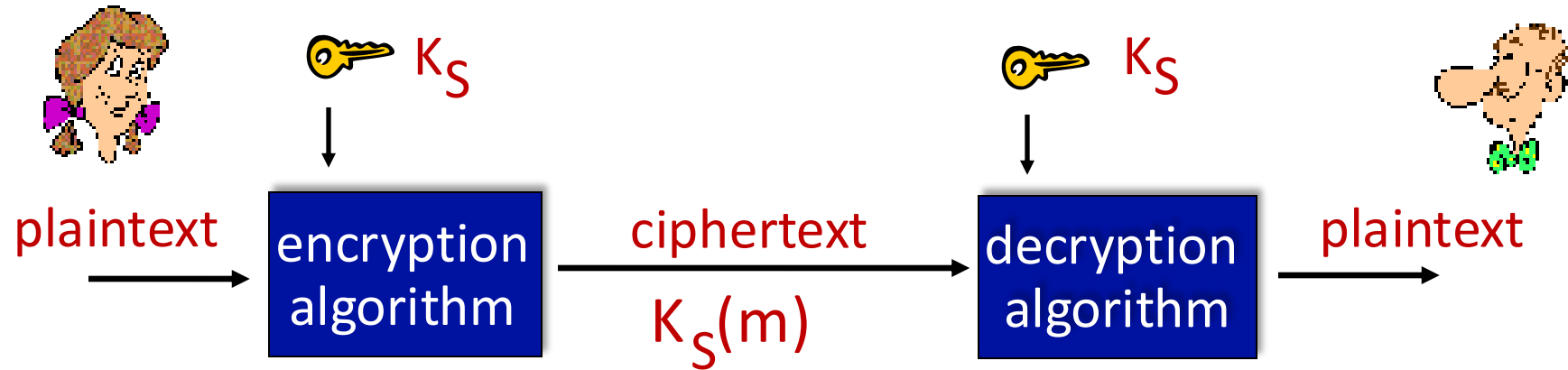


$m$ : plaintext message

$K_A(m)$ : ciphertext, encrypted with key  $K_A$

$m = K_B(K_A(m))$

# Symmetric key cryptography



**symmetric key crypto:** Bob and Alice share same (symmetric) key:  $K$

- e.g., key is knowing substitution pattern in mono alphabetic substitution cipher

Q: how do Bob and Alice agree on key value?



# Simple encryption scheme

*substitution cipher*: substituting one thing for another

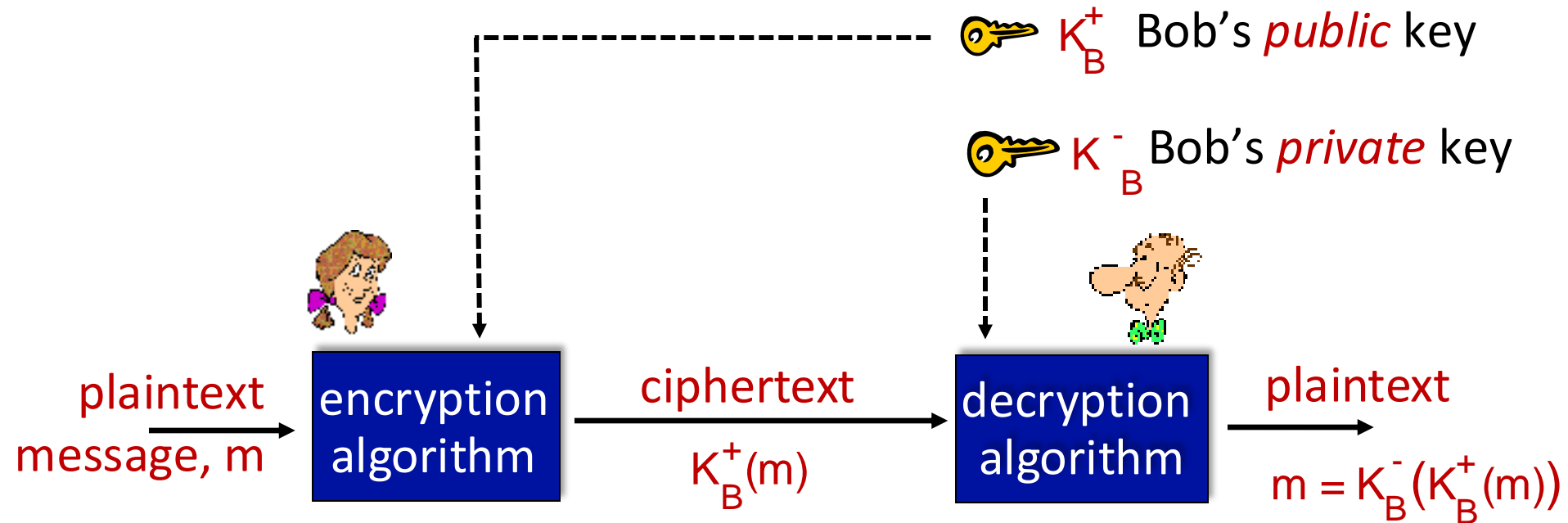
- monoalphabetic cipher: substitute one letter for another

plaintext:	a	b	c	d	e	f	g	h	i	j	k	l	m	n	o	p	q	r	s	t	u	v	w	x	y	z
		↓																								↓
ciphertext:	m	n	b	v	c	x	z	a	s	d	f	g	h	j	k	l	p	o	i	u	y	t	r	e	w	q

e.g.: Plaintext: bob. i love you. alice  
ciphertext: nkn. s gktc wky. mgsbc

🔑 *Encryption key*: mapping from set of 26 letters  
to set of 26 letters

# Public Key Cryptography



Public key cryptography revolutionized 2000-year-old (previously only symmetric key) cryptography!

- similar ideas emerged at roughly same time, independently in US and UK (classified)

# Public key encryption algorithms

requirements:

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

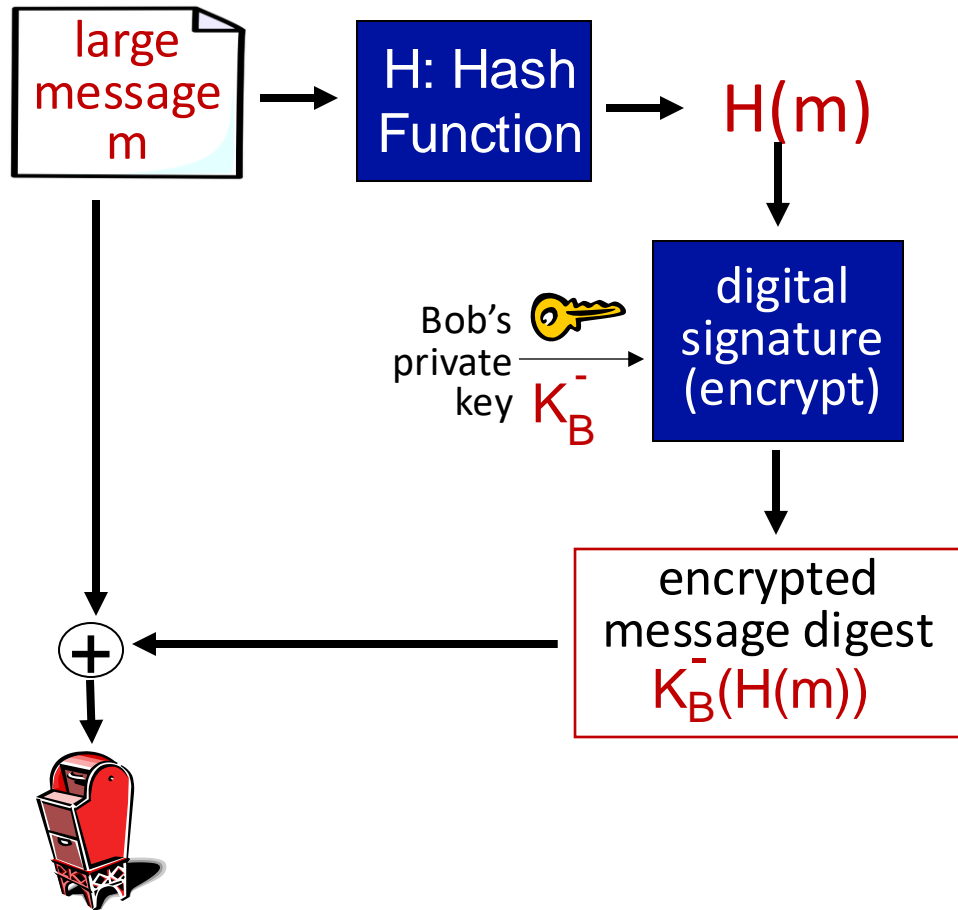
$$K_B^-(K_B^+(m)) = m$$

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

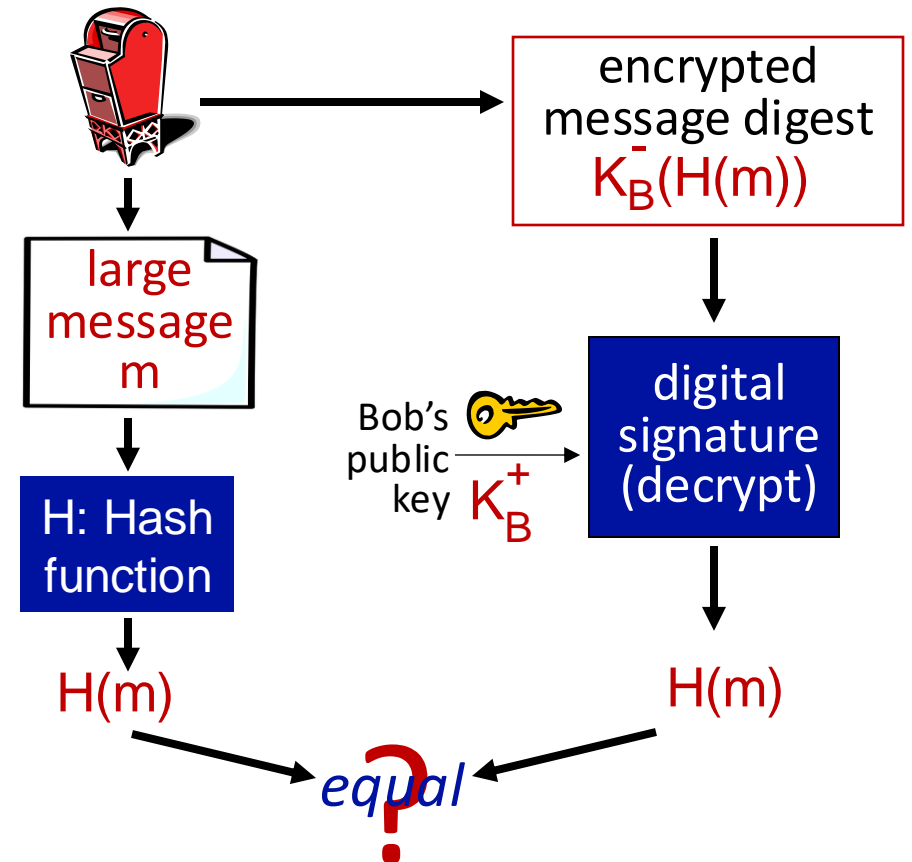
**RSA:** Rivest, Shamir, Adelson algorithm

# Digital signature = signed message digest

Bob sends digitally signed message:

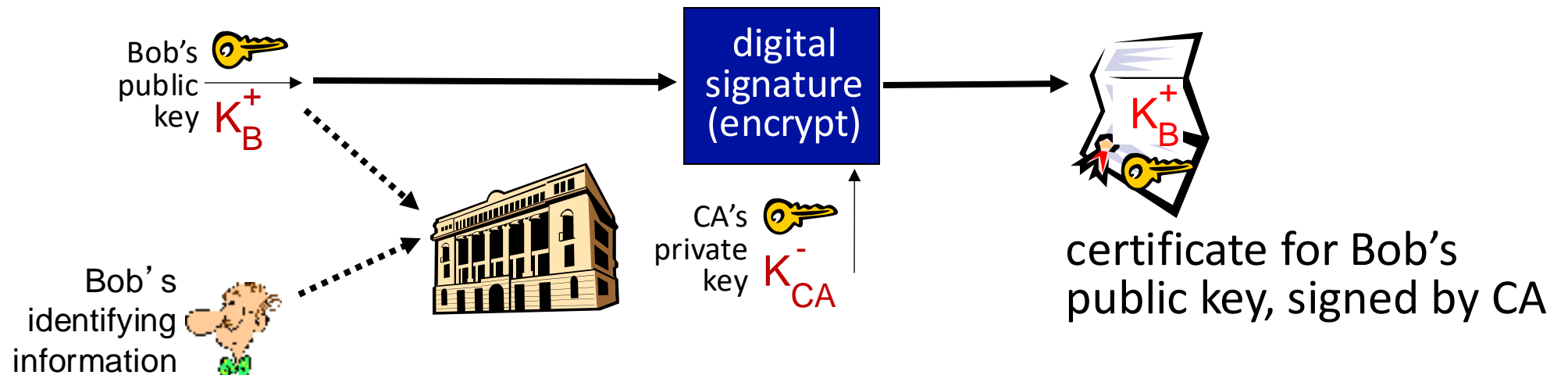


Alice verifies signature, integrity of digitally signed message:



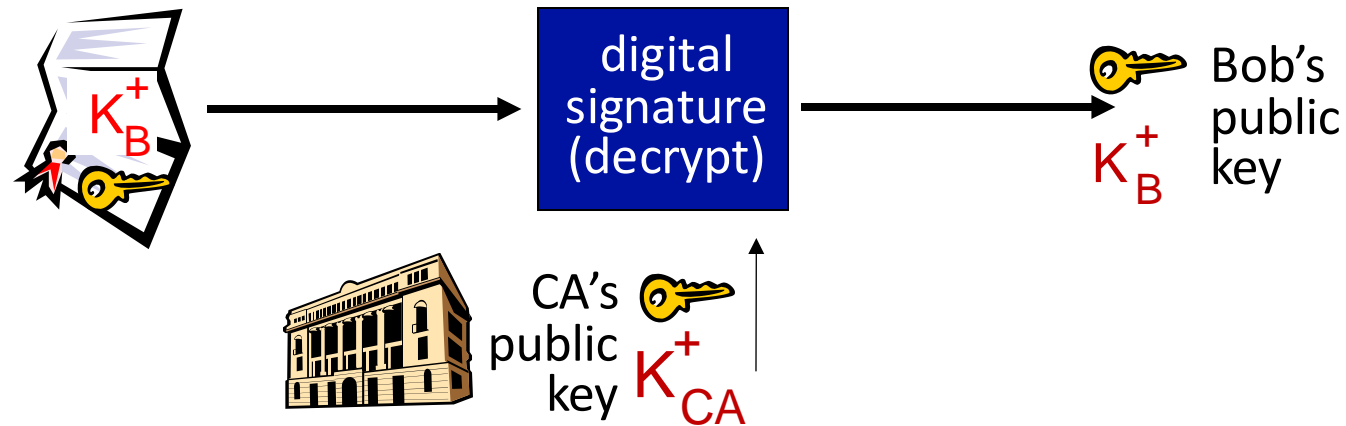
# Public key Certification Authorities (CA)

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



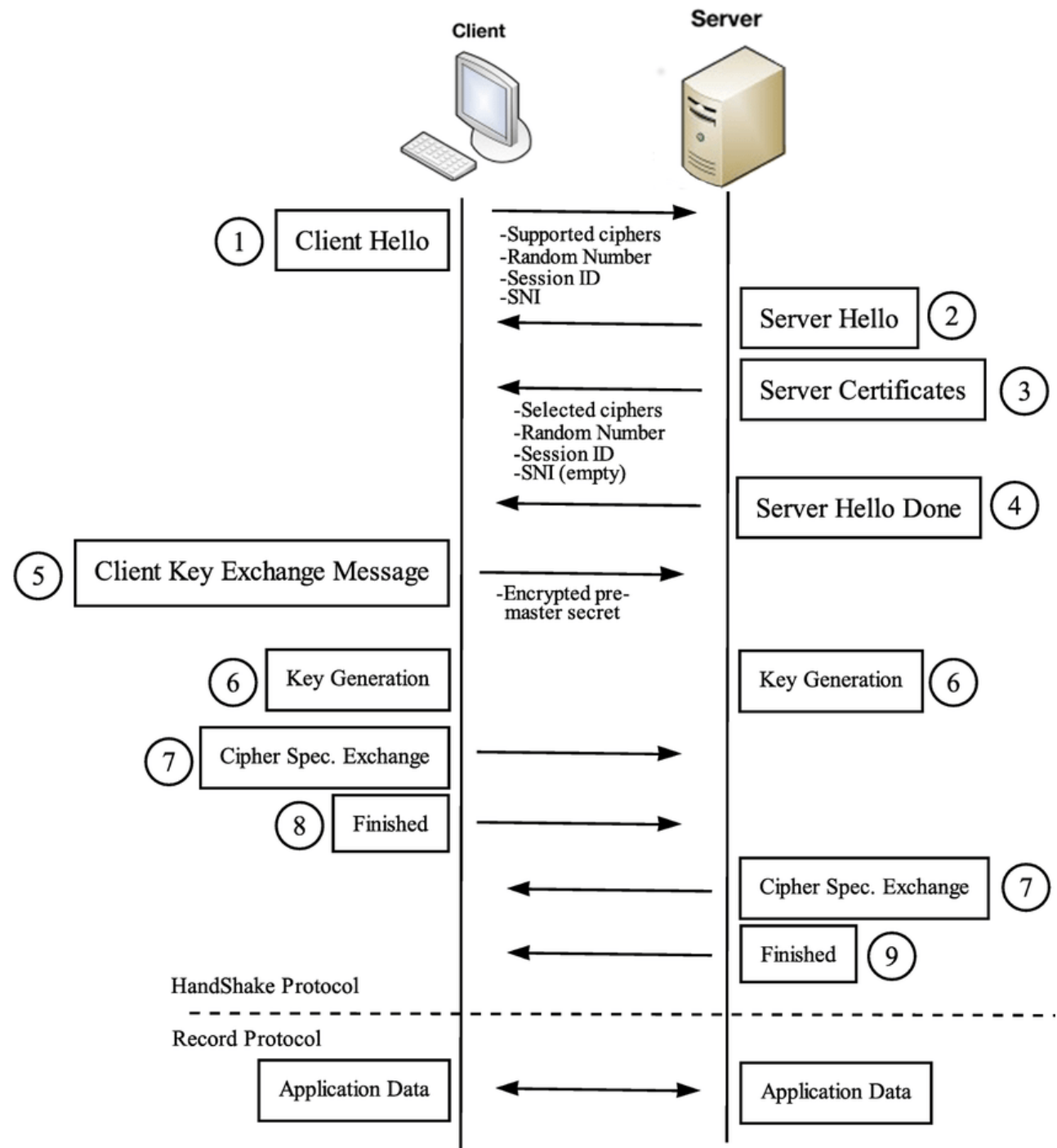
# Public key Certification Authorities (CA)

- when Alice wants Bob's public key:
  - gets Bob's certificate (Bob or elsewhere)
  - apply CA's public key to Bob's certificate, get Bob's public key



# TLS handshake

**Server Name Indication (SNI)** is an extension to the Transport Layer Security (TLS) computer networking protocol by which a client indicates which hostname it is attempting to connect to at the start of the handshaking process. This allows a server to present one of multiple possible certificates on the same IP address and TCP port number and hence allows multiple secure (HTTPS) websites (or any other service over TLS) to be served by the same IP address without requiring all those sites to use the same certificate. (source: wikipedia.org)



# Socket programming

Two socket types for two transport services:

- *UDP*: unreliable datagram
- *TCP*: reliable, byte stream-oriented

## Application Example:

1. client reads a line of characters (data) from its keyboard and sends data to server
2. server receives the data and converts characters to uppercase
3. server sends modified data to client
4. client receives modified data and displays line on its screen



# Socket programming with UDP

**UDP:** no “connection” between client & server

- no handshaking before sending data
- sender explicitly attaches IP destination address and port # to each packet
- receiver extracts sender IP address and port# from received packet

**UDP:** transmitted data may be lost or received out-of-order

**Application viewpoint:**

- UDP provides *unreliable* transfer of groups of bytes (“datagrams”) between client and server

# Client/server socket interaction: UDP



**server** (running on serverIP)

create socket, port= x:  
`serverSocket =  
socket(AF_INET,SOCK_DGRAM)`

read datagram from  
`serverSocket`

write reply to  
`serverSocket`  
specifying  
client address,  
port number

**client**



create socket:  
`clientSocket =  
socket(AF_INET,SOCK_DGRAM)`

Create datagram with server IP and  
port=x; send datagram via  
`clientSocket`

read datagram from  
`clientSocket`

close  
`clientSocket`

# Example app: UDP client

## *Python UDPClient*

include Python's socket library → `from socket import *`  
`serverName = 'hostname'`  
`serverPort = 12000`  
create UDP socket for server → `clientSocket = socket(AF_INET,  
SOCK_DGRAM)`  
get user keyboard input → `message = raw_input('Input lowercase sentence:')`  
attach server name, port to message; send into socket → `clientSocket.sendto(message.encode(),  
(serverName, serverPort))`  
read reply characters from socket into string → `modifiedMessage, serverAddress =  
clientSocket.recvfrom(2048)`  
print out received string and close socket → `print modifiedMessage.decode()  
clientSocket.close()`

# Example app: UDP server

## *Python UDPServer*

```
from socket import *
serverPort = 12000

create UDP socket → serverSocket = socket(AF_INET, SOCK_DGRAM)
bind socket to local port number 12000 → serverSocket.bind("", serverPort)
print ("The server is ready to receive")

loop forever → while True:
    Read from UDP socket into message, getting → message, clientAddress = serverSocket.recvfrom(2048)
    client's address (client IP and port)      modifiedMessage = message.decode().upper()
    send upper case string back to this client → serverSocket.sendto(modifiedMessage.encode(),
                                                    clientAddress)
```

# Socket programming with TCP

## Client must contact server

- server process must first be running
- server must have created socket (door) that welcomes client's contact

## Client contacts server by:

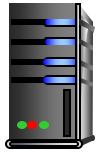
- Creating TCP socket, specifying IP address, port number of server process
- *when client creates socket*: client TCP establishes connection to server TCP

- when contacted by client, *server TCP creates new socket* for server process to communicate with that particular client
  - allows server to talk with multiple clients
  - source port numbers used to distinguish clients (more in Chap 3)

## Application viewpoint

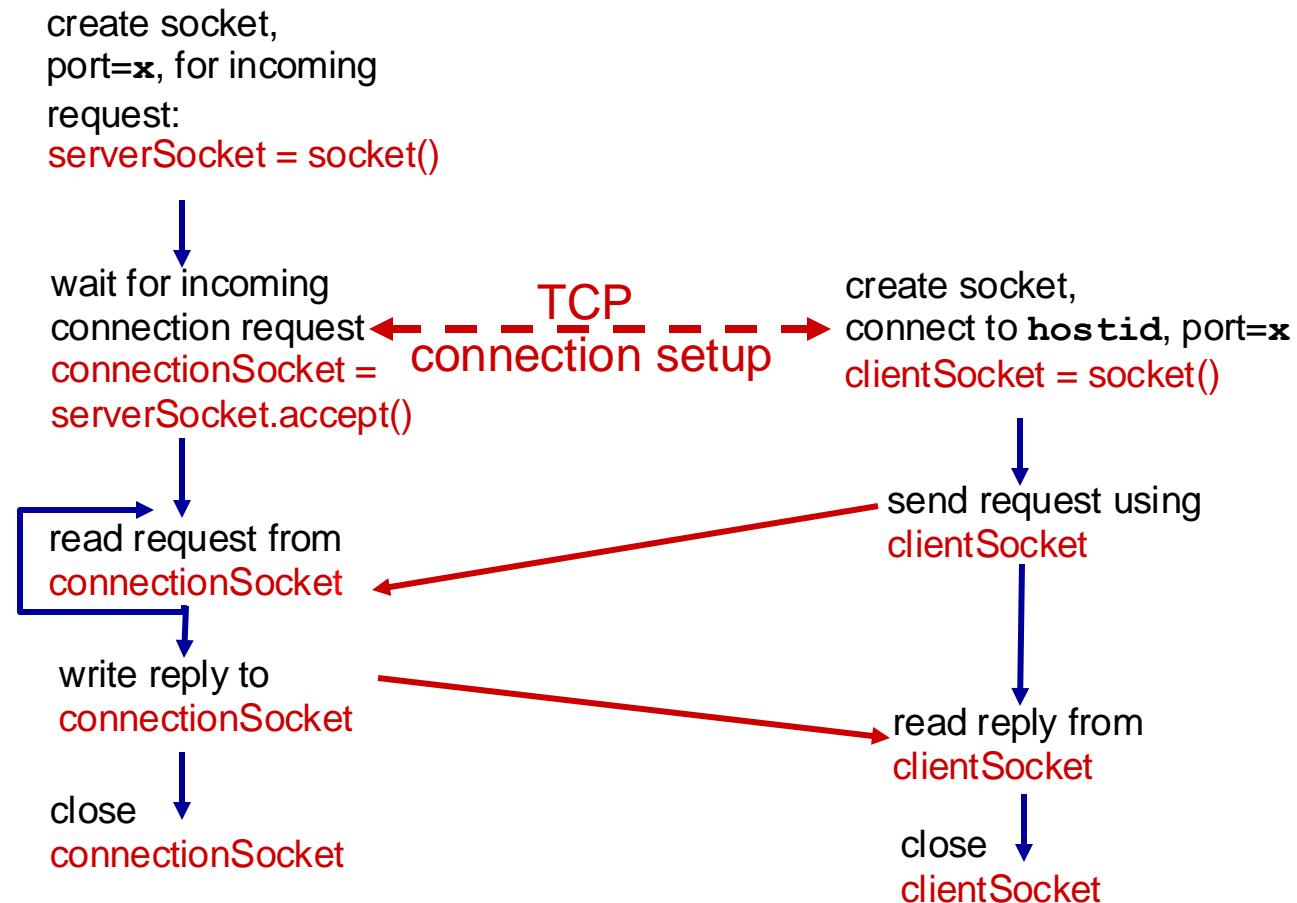
TCP provides reliable, in-order byte-stream transfer ("pipe") between client and server

# Client/server socket interaction: TCP



server (running on `hostid`)

client



# Example app: TCP client

## *Python TCPClient*

create TCP socket for server,  
remote port 12000

```
from socket import *
serverName = 'servername'
serverPort = 12000
clientSocket = socket(AF_INET, SOCK_STREAM)
clientSocket.connect((serverName,serverPort))
sentence = raw_input('Input lowercase sentence:')
clientSocket.send(sentence.encode())
modifiedSentence = clientSocket.recv(1024)
print ('From Server:', modifiedSentence.decode())
clientSocket.close()
```

No need to attach server name, port

# Example app: TCP server

## *Python TCPServer*

		<pre>from socket import *</pre>
		<pre>serverPort = 12000</pre>
create TCP welcoming socket	→	<pre>serverSocket = <b>socket</b>(AF_INET,SOCK_STREAM)</pre>
		<pre>serverSocket.<b>bind</b>(('',serverPort))</pre>
server begins listening for incoming TCP requests	→	<pre>serverSocket.<b>listen</b>(1)</pre>
		<pre>print 'The server is ready to receive'</pre>
loop forever	→	<pre>while True:</pre>
server waits on accept() for incoming requests, new socket created on return	→	<pre>    connectionSocket, addr = serverSocket.<b>accept</b>()</pre>
		<pre>    sentence = connectionSocket.<b>recv</b>(1024).decode()</pre>
read bytes from socket (but not address as in UDP)	→	<pre>    capitalizedSentence = sentence.upper()</pre>
		<pre>    connectionSocket.<b>send</b>(capitalizedSentence.                            encode())</pre>
close connection to this client (but <i>not</i> welcoming socket)	→	<pre>    connectionSocket.<b>close</b>()</pre>



# TCP/IP socket programming in C

- See example from book:
  - "[TCP/IP Sockets in C: Practical Guide for Programmers, Second Edition](#)"  
by Michael J. Donahoo and Kenneth L. Calvert