University of Glasgow

Using the Transport Layer

Networked Systems 3
Lecture 12

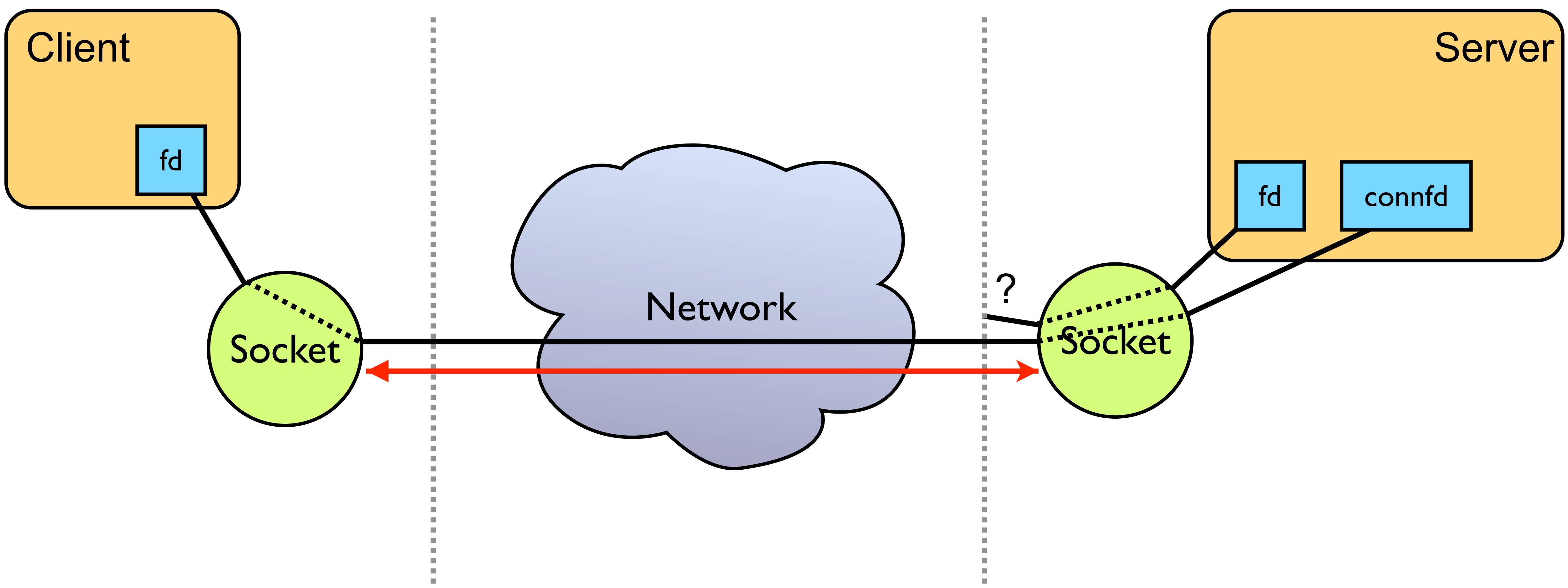
Lecture Outline

- Using TCP connections
- Using UDP datagrams
- NAT traversal concepts

Using TCP Connections

- A TCP connection provides a reliable byte stream abstraction, identifying applications by a 16 bit port number
  - Role of the port number
  - Implications of the reliable byte stream abstraction
    - Record boundaries
    - Head of line blocking
  - Implications of NAT on TCP connections

Using TCP Connections




```
int fd = socket(...)

connect(fd, ..., ...)

write(fd, data, datalen)

read(fd, buffer, buflen)

close(fd)
```

source and destination ports specified

```
int fd = socket(...)
bind(fd, ..., ...)
listen(fd, ...)
connfd = accept(fd, ...)

read(connfd, buffer, buflen)

write(connfd, data, datalen)

close(connfd)
```

Role of the TCP Port Number

Port Range		Name	Intended use
0	1023	Well-known (system) ports	Trusted operating system services
1024	49151	Registered (user) ports	User applications and services
49152	65535	Dynamic (ephemeral) ports	Private use, peer-to-peer applications, source ports for TCP client connections

RFC 6335

- Servers must listen on a known port; IANA maintains a registry
- Distinction between system and user ports ill-advised – security problems resulted
- Insufficient port space available (>75% of ports are registered)

- TCP clients traditionally connect from a randomly chosen port in the ephemeral range
  - The port must be chosen randomly, to prevent spoofing attacks
  - Many systems use the entire port range for source ports, to increase the amount of randomness available

http://www.iana.org/assignments/port-numbers

TCP Congestion Control

- A TCP connection reliably delivers a byte stream
  - The transmission speed depends on network conditions (→ lecture 14)
  - The `write()` call will block if there is insufficient network or buffer capacity
    - Can use the `select()` function to determine if a call will block, but cannot determine how long a particular `write()` will take

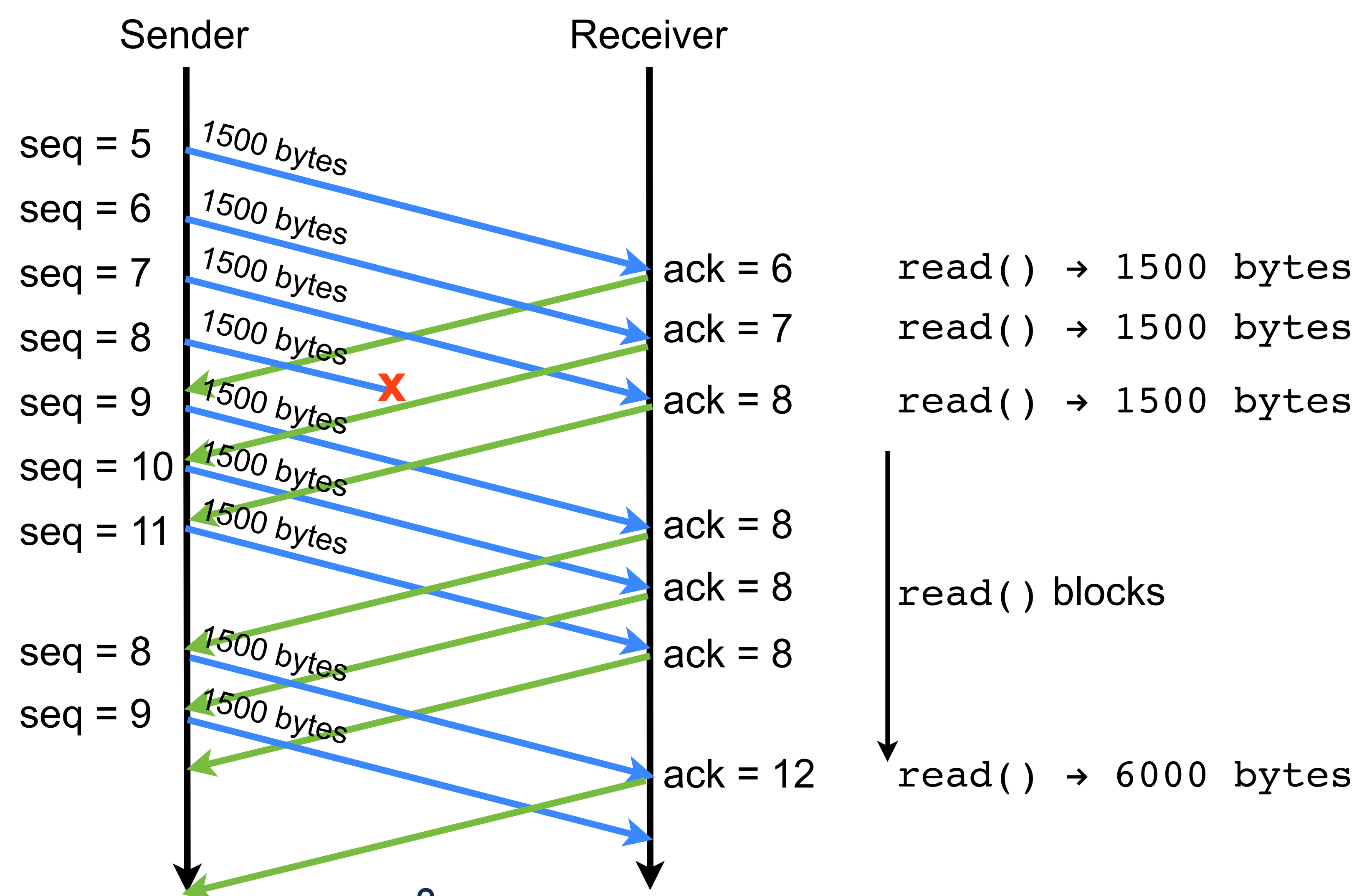
```
int     fd;
fd_set wfds;
...
FD_ZERO(&wfds);
FD_SET(fd, & wfds);
if (select(fd+1, NULL, &wfds, NULL, NULL) > 0) {
    // Space is available to write()
}
```

Record Boundaries in TCP Connections

- If the data in a `write()` is bigger than the data link layer MTU, TCP will send the data as fragments
- Similarly, multiple small `write()` requests may be aggregated into a single TCP packet
- Implication: the data returned by a `read()` doesn't necessarily match that sent in a single `write()`
  - There often appears to be a correspondence, but this *is not* guaranteed (it may work in the lab, but not when you use it over a different link)

Head of Line Blocking in TCP

- What if data is lost due to network congestion?
  - TCP will retransmit the missing data, transparently to the application (→ lecture 13)
  - A `read()` for the missing data will block until it arrives; TCP delivers all data in-order



Application Level Framing

Data may arrive in arbitrary sized chunks; must parse and understand the data, no matter where it is split by the network – it's a byte stream (colours indicate one possible split of the data into chunks)

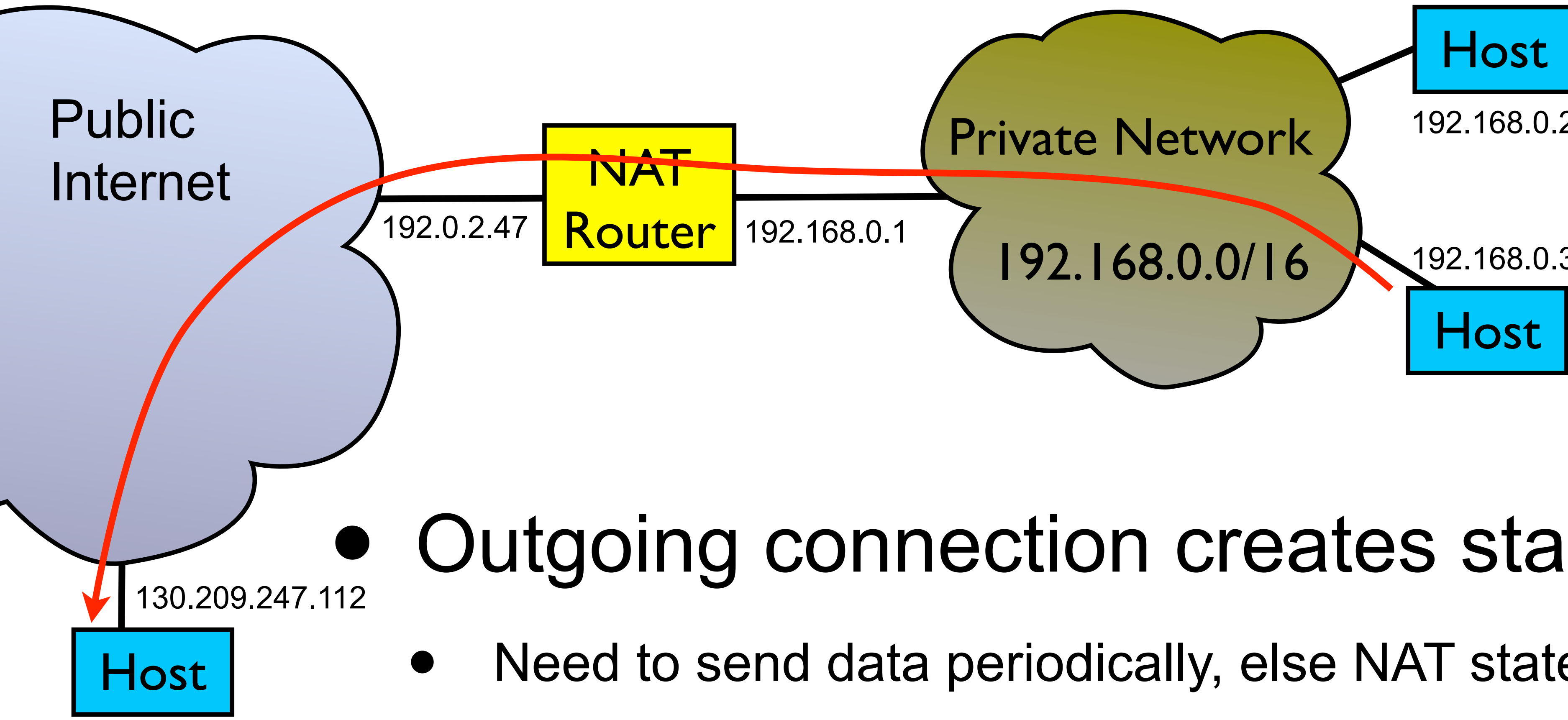
```
HTTP/1.1 200 OK
Date: Mon, 19 Jan 2009 22:25:40 GMT
Server: Apache/2.0.46 (Scientific Linux)
Last-Modified: Mon, 17 Nov 2003 08:06:50 GMT
ETag: "57c0cd-e3e-17901a80"
Accept-Ranges: bytes
Content-Length: 3646
Connection: close
Content-Type: text/html; charset=UTF-8

<HTML>
<HEAD>
<TITLE>Computing Science, University of Glasgow </TITLE>
...
</BODY>
</HTML>
```

Example: HTTP response

Known marker (blank line) signals end of headers

Size of payload indicated in the headers

Implications of NAT for TCP Connections



- Outgoing connection creates state in NAT
  - Need to send data periodically, else NAT state times out
  - Recommended time out interval is 2 hours, many NATs use shorter RFC5382
- Server behind NAT requires configured mapping
- Peer-to-peer connections difficult
  - Simultaneous open with external mapping service

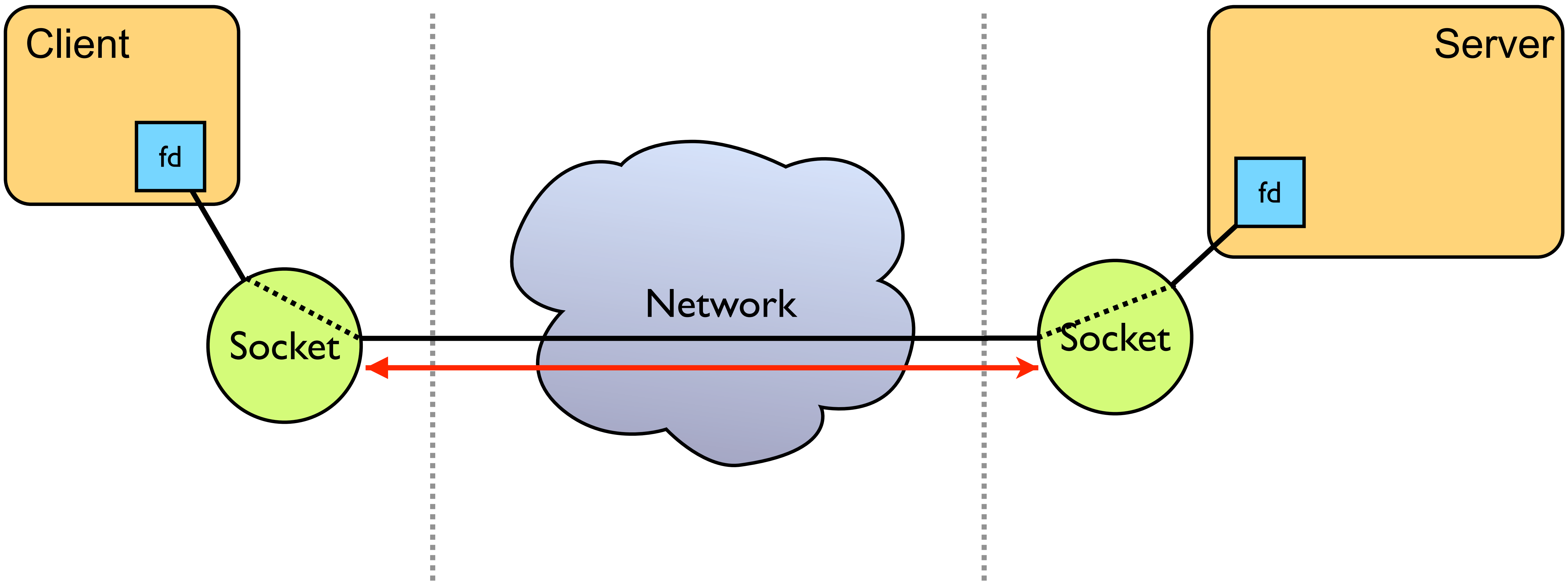
Using UDP Datagrams

- UDP provides an unreliable datagram service, identifying applications via a 16 bit port number
  - UDP ports are separate from TCP ports
  - Often used peer-to-peer (e.g. for VoIP), so both peers must `bind()` to a known port
  - Create via `socket()` as usual, but specify `SOCK_DGRAM` as the socket type:

```
int     fd;
...
fd = socket(AF_INET, SOCK_DGRAM, 0);
```

  - No need to `connect()` or `accept()`, since no connections in UDP

Using UDP Datagrams




```
int fd = socket(...)

bind(fd, ..., ...)

sendto(fd, data, datalen, addr, addrlen)

recvfrom(fd, buffer, buflen, flags, addr, addrlen)

close(fd)
```

Sending UDP Datagrams

The `sendto()` call sends a single datagram. Each call to `sendto()` can send to a different address, even though they use the same socket.

```
int                   fd;
char                  buffer[...];
int                   buflen = sizeof(buffer);
struct sockaddr_in    addr;
...
if (sendto(fd, buffer, buflen, (struct sockaddr *) addr, sizeof(addr)) < 0) {
    // Error...
}
```

Alternatively, `connect()` to an address, then use `write()` to send the data. There is no connection made at the UDP layer, the `connect()` call only sets the destination address for future packets.

Receiving UDP Datagrams

The `read()` call may be used to read a single datagram, but doesn't provide the source address of the datagram. Most code uses `recvfrom()` instead – this fills in the source address of the received datagram:

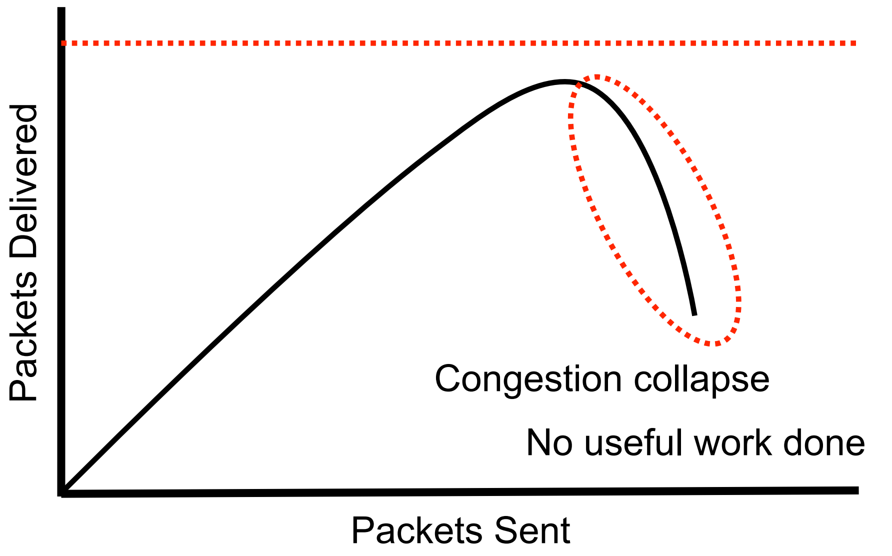
```
int                  fd;
char                 buffer[...];
int                  buflen = sizeof(buffer);
struct sockaddr      addr;
socklen_t            addr_len = sizeof(addr);
int                  rlen;
...
rlen = recvfrom(fd, buffer, buflen, 0, &addr, &addrlen);
if (rlen < 0) {
    // Error...
}
```

UDP Framing and Reliability

- Unlike TCP, each UDP datagram is sent as exactly one IP packet (which may be fragmented in IPv4)
  - Each `read()` corresponds to a single `write()`
- But, transmission is unreliable: packets may be lost, delayed, reordered, or duplicated in transit
  - The application is responsible for correcting the order, detecting duplicates, and repairing loss – if necessary
  - Generally requires the sender to include some form of sequence number in each packet sent

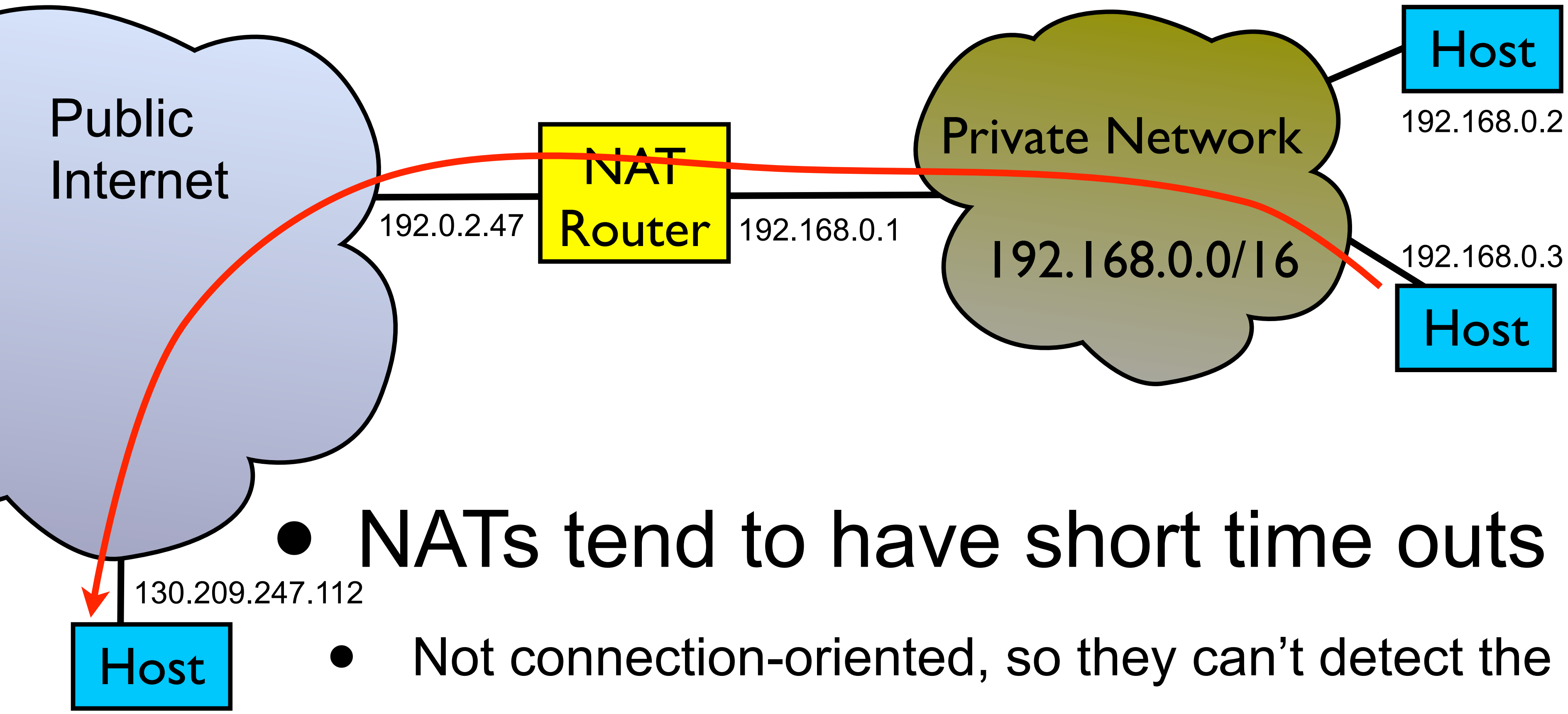
UDP Guidelines

- Need to implement congestion control in applications
  - To avoid congestion collapse of the network
  - Should be approximately fair to TCP
  - RFC 3448 provides one algorithm for doing this



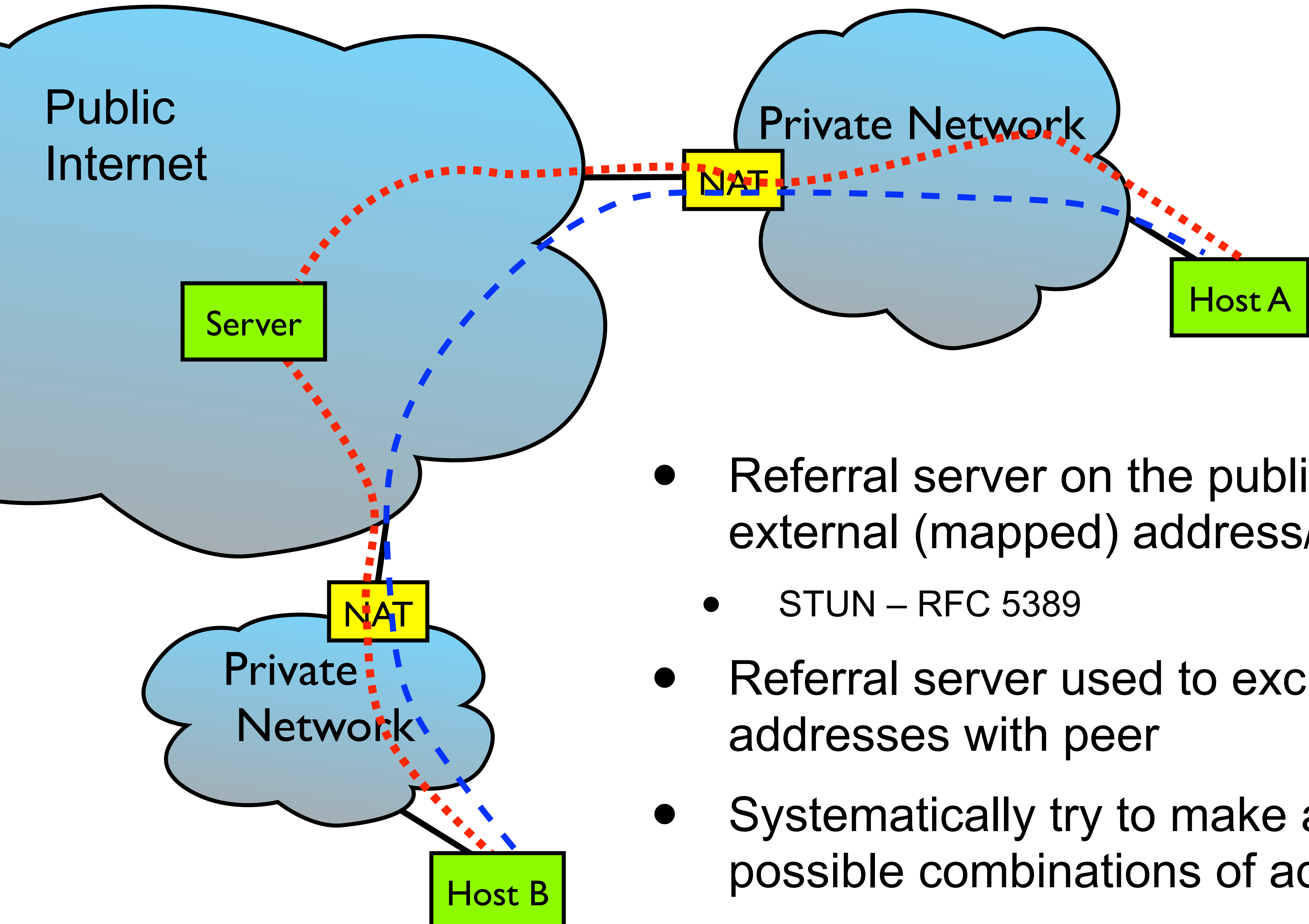
- Need to provide sequencing, reliability, and timing in applications
  - Sequence numbers and acknowledgements
  - Retransmission and/or forward error correction
  - Timing recovery
- UDP programming guidelines: RFC 5405

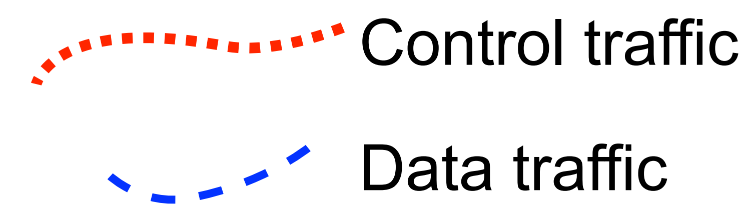
Implications of NAT for UDP Flows



- NATs tend to have short time outs for UDP
  - Not connection-oriented, so they can't detect the end of flows
  - Recommended time out interval is not less than two minutes, but many NATs use shorter intervals – the VoIP NAT traversal standards suggest sending a keep alive message every 15 seconds RFC4787
- Peer-to-peer connections easier than TCP
  - UDP NATs are often more permissive about allowing incoming packets than TCP NATs; many allow replies from anywhere to an open port

NAT Traversal Concepts





- Referral server on the public network used to discover external (mapped) address/port on the NAT
  - STUN – RFC 5389
- Referral server used to exchange possible connection addresses with peer
- Systematically try to make a connection using all possible combinations of addresses
  - Every possible network interface and protocol, mapped and local
  - ICE – RFC 5245

Happy Eyeballs for Dual-Stack Hosts

- Hosts may have both IPv4 and IPv6 addresses – which to use?
  - Prefer IPv6 if available – but deployments are new, and can be unreliable
  - Need to fallback to IPv4 in case of problems
- Basic approach: try all possible addresses in turn
  - Problematic, since connections may take 10s of seconds to fail
- Happy eyeballs approach: try addresses in parallel

```
1. Call getaddinfo(): returns a list of IP addresses sorted by
   the host's address preference policy
2. Try to connection to first address in that list (e.g., IPv6)
3. If that connection does not complete within a short period of
   time (e.g., 200-300ms), try parallel connection to the first
   address belonging to the other address family (e.g., IPv4)
4. First connection established is used; the other is discarded
5. If neither succeeds, repeat with next pair of addresses until
   all possibilities tried
```

Questions?