Administrivia

- Project 3 due Thursday
  - As usual, due at 4:15pm
  - Extension to midnight if you come to class
  - For longer extensions, please email cs140-staff
- Project 4 goes out at end of week
- This Friday will have section on project 4

Lab 1 grades

- Mean: 149

Lab 2 grades

- Mean: 162

Networks

- What is a network?
  - A system of lines/channels that interconnect
  - E.g., railroad, highway, plumbing, communication, telephone, computer
- What is a computer network?
  - A form of communication network—moves information
  - Nodes are general-purpose computers
- Computer networks are particularly interesting
  - You can program the nodes
  - Very easy to innovate and develop new uses of network
  - Contrast: Telephone network—can’t program most phones, need FCC approval for new devices, etc.

Physical connectivity

- Computers send bits over physical links
  - E.g., Coax, twisted pair, fiber, radio, …
  - Bits may be encoded as multiple lower-level “chips”
- Two categories of physical links
  - Point-to-point networks (e.g., fiber, twisted pair):
  - Shared transmission medium networks (e.g., coax, radio):
    - Any message can be seen by all nodes
    - Allows broadcast/multicast, but introduces contention
- One implication: speed of light matters!
  - $\sim 300,000$ km/sec in a vacum, slower in fiber
  - $\text{SF} \geq 15\text{ msec} \rightarrow \text{NYC}$ Moore’s law does not apply!

Bandwidth-delay

- Network delay over WAN will never improve much
- But throughput (bits/sec) is constantly improving
- Can view network as a pipe

  - For full utilization want $\#\text{bytes in flight} \geq \text{bandwidth} \times \text{delay}$ (But don’t want to overload the network, either)
- What if protocol doesn’t involve bulk transfer?
  - E.g., ping-pong protocol will have poor throughput
- Another implication: Concurrency is critical for good network utilization
Indirect connectivity

- Rarely have direct physical connection to destination
- Instead, communications usually “hop” through multiple devices
  - Allows links and devices to be shared for multiple purposes
  - Must determine which bits are part of which messages intended for which destinations

- Circuit switched networks provide virtual links
  - E.g., old telephone network when numbers dialed

- Packet switched networks
  - Pack a bunch of bytes together intended for same destination
  - Slap a header on packet describing where it should go
  - Most networks today are packet switched

Why Ethernet is insufficient

- Ethernet Limits
  - 2,500m diameter
  - 100 nodes
- Can bridge multiple Ethereats
  - First time you see destination address, send packet to all segments
  - Then learn where devices are, and avoid forwarding useless packets
- A switch is like a bridge with \( n > 2 \) ports
  - Widely used within organizations
  - But could never scale to the size of the Internet
- Moreover, need to communicate across networks
  - E.g., laptop w. DSL or wireless contacting server w. Ethernet

Internet Protocol (IP)

- IP used to connect multiple networks
  - Runs over a variety of physical networks
  - Most computers today speak IP
- Every host has a unique 4-byte IP address
  - (Or at least thinks it has, when address shortages)
  - E.g., \( \text{www.ietf.org} \rightarrow 132.151.6.21 \)
- Packets are routed based on destination IP address
  - Address space is structured to make routing practical at global scale
  - E.g., \( 171.66.*.* \) goes to Stanford
  - So packets need IP addresses in addition to MAC addresses

Layering

- Stick packets inside packets
  - E.g., an Ethernet packet may encapsulate an IP packet
  - An IP router forwards a packet from one Ethernet to another, creating a new Ethernet packet containing the same IP packet
  - In principle, an inner layer should not depend on outer layers
  - E.g., IP packets should be independent of Ethernet
  [Annoyingly, TCP (next) has a checksum that violates this]

ARP

- When forwarding an IP packet from Ethernet \( A \) to Ethernet \( B \), what MAC address should you use?
  - If destination host physically connected, use its MAC address
  - Otherwise, use MAC address of next router (given IP address)
- Must map IP addresses into physical addresses
- ARP = address resolution protocol
  - Broadcast request for MAC address of IP address
  - Everybody learns the requesting node’s MAC address
  (since broadcast request has requester’s MAC and IP addresses)
  - Target machine responds with its MAC address
- OS keeps ARP cache w. IP \( \rightarrow \) MAC addr. mappings
  - Periodically discard entries that have not been refreshed
  - E.g., run \( \text{arp} -a \) on Unix to see contents of ARP cache
ARP Ethernet packet format

<table>
<thead>
<tr>
<th></th>
<th>0</th>
<th>8</th>
<th>16</th>
<th>31</th>
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<tbody>
<tr>
<td>Hardware type</td>
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<td></td>
<td></td>
</tr>
<tr>
<td>ProtocolType</td>
<td>0x0800</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>HLen</td>
<td>48</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>PLen</td>
<td>32</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>SourceHardwareAddr</td>
<td></td>
<td>0-3</td>
<td>0-1</td>
<td></td>
</tr>
<tr>
<td>SourceProtocolAddr</td>
<td></td>
<td>2-3</td>
<td>0-1</td>
<td>0-3</td>
</tr>
<tr>
<td>TargetHardwareAddr</td>
<td>0-1</td>
<td></td>
<td>0-1</td>
<td></td>
</tr>
<tr>
<td>TargetProtocolAddr</td>
<td></td>
<td>2-5</td>
<td></td>
<td>0-3</td>
</tr>
</tbody>
</table>

[Hardware address = MAC address]

Inter-process communication

- Want abstraction of inter-process (not just inter-node) communication
- Solution: Encapsulate other protocols within IP

Failure

- Many more failure modes on net than w. local IPC
- Several types of error can affect packet delivery
  - Bit errors (e.g., electrical interference, cosmic rays)
  - Packet loss (packets dropped when queues fill on overload)
  - Link and node failure
- In addition, properly delivered frames can be delayed, reordered, even duplicated
- How much should OS expose to application
  - Some failures cannot be masked (e.g., server dead)
  - Others can be (e.g., retransmit lost packet)
  - But masking errors may be wrong for some applications (e.g., old audio packet no longer interesting if too late to play)

UDP and TCP

- UDP and TCP most popular protocols on IP
  - Both use 16-bit port number as well as 32-bit IP address
  - Applications bind a port & receive traffic to that port
- UDP – unreliable datagram protocol
  - Exposes packet-switched nature of Internet
  - Sent packets may be dropped, reordered, even duplicated (but generally not corrupted)
- TCP – transmission control protocol
  - Provides illusion of a reliable "pipe" between two processes on two different machines
  - Masks lost & reordered packets so apps don’t have to worry
  - Handles congestion & flow control

Uses of TCP

- Most applications use TCP
  - Easier interface to program to (reliability)
  - Automatically avoids congestion (don’t need to worry about taking down network)
- Servers typically listen on well-known ports
  - SSH: 22
  - Email: 25
  - Finger: 79
  - Web / HTTP: 80
- Example: Interacting with www.stanford.edu
  - Browser resolves IP address of www.stanford.edu
    (171.67.216.15)
  - Browser connects to TCP port 80 on 171.67.216.15
  - Over TCP connection, browser requests and gets home page

Unreliability of IP

- Network does not deliver packets reliably
  - May drop packets, reorder packets, delay packets
  - May even corrupt packets, or duplicate them
- How to implement reliable TCP on top of IP network?
  - Note: This is entirely the job of the OS at the end nodes
- Straw man: Wait for ack for each packet
  - Send a packet, wait for acknowledgment, send next packet
  - If no ack, timeout and try again
- Problems:
  - Low performance over high-delay network
    (bandwidth is one packet per round-trip time)
  - Possible congestive collapse of network
    (if everyone keeps retransmitting when network overloaded)
A little bit about TCP

- Want to save network from congestive collapse
  - Packet loss usually means congestion, so back off exponentially
- Want multiple outstanding packets at a time
  - Get transmit rate up to \( n \)-packet window per round-trip
- Must figure out appropriate value of \( n \) for network
  - Slowly increase transmission by one packet per acked window
  - When a packet is lost, cut window size in half
- Connection set up and tear down complicated
  - Sender never knows when last packet might be lost
  - Must keep state around for a while after close
- Lots more hacks required for good performance
  - Initially ramp \( n \) up faster (but too fast caused collapse in 1986 [Jacobson], so TCP had to be changed)
  - Fast retransmit when single packet lost

Lots of OS issues for TCP

- Have to track unacknowledged data
  - Keep a copy around until recipient acknowledges it
  - Keep timer around to retransmit if no ack
- Receiver must keep out of order segments & reassemble
- When to wake process receiving data?
  - E.g., sender calls write (fd, message, 8000);
  - First TCP segment arrives, but is only 512 bytes
  - Could wake recipient, but useless w/o full message
  - TCP sets “PUSH” bit at end of 8000 byte write data
- When to send short segment, vs. wait for more data
  - Usually send only one unacked short segment
  - But bad for some apps, so provide NODELAY option
- Must ack received segments very quickly
  - Otherwise, effectively increases RTT, decreasing bandwidth

OS interface to TCP/IP

- What interface should OS provide to TCP/IP?
- Inspired by pipes (int pipe (int fds[2]);)
  - Allow Inter-process communication on one machine
  - Writes to \(fds[1]\) will be read on \(fds[0]\)
  - Can give each file descriptor to a different process (w. fork)
- Idea: Provide similar abstraction across machines
  - Write data on one machine, read it on the other
  - Allows processes to communicate over the network
- Complications across machines
  - How do you set up the file descriptors between processes?
  - How do you deal with failure?
  - How do you get good performance?

Sockets

- Abstraction for communication between machines
- Datagram sockets: Unreliable message delivery
  - With IP, gives you UDP
  - Send atomic messages, which may be reordered or lost
  - Special system calls to read/write: send/recev
- Stream sockets: Bi-directional pipes
  - With IP, gives you TCP
  - Bytes written on one end read on the other
  - Reads may not return full amount requested—must re-read

Socket naming

- TCP & UDP name communication endpoints by
  - 32-bit IP address specifies machine
  - 16-bit TCP/UDP port number demultiplexes within host
- A connection is thus named by 5 components
  - Protocol (TCP), local IP, local port, remote IP, remote port
  - TCP requires connected sockets, but not UDP
- OS keeps connection state in protocol control block (PCB) structure
  - Keep all PCB’s in a hash table
  - When packet arrives (if destination IP address belongs to host), use 5-tuple to find PCB and determine what to do with packet

System calls for using TCP

<table>
<thead>
<tr>
<th>Client</th>
<th>Server</th>
</tr>
</thead>
<tbody>
<tr>
<td>socket – make socket</td>
<td></td>
</tr>
<tr>
<td>bind – assign address</td>
<td></td>
</tr>
<tr>
<td>listen – listen for clients</td>
<td></td>
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<td>connect – connect to listening socket</td>
<td></td>
</tr>
<tr>
<td>accept – accept connection</td>
<td></td>
</tr>
</tbody>
</table>

*This call to bind is optional; connect can choose address & port.
Client interface

```c
struct sockaddr_in {
    short sin_family; /* = AF_INET */
    u_short sin_port; /* = htons (PORT) */
    struct in_addr sin_addr;
    char sin_zero[8];
} sin;

int s = socket (AF_INET, SOCK_STREAM, 0);
bzero (&sin, sizeof (sin));
sin.sin_family = AF_INET;
sin.sin_port = htons (13); /* daytime port */
sin.sin_addr.s_addr = htonl (IP_ADDRESS);
connect (s, (sockaddr *) &sin, sizeof (sin));
```

Using UDP

- **Call** socket with SOCK_DGRAM, bind as before
- **New system calls** for sending individual packets
  - int sendto(int s, const void *msg, int len, int flags, const struct sockaddr *to, socklen_t tolen);
  - int recvfrom(int s, void *buf, int len, int flags, struct sockaddr *from, socklen_t *fromlen);
  - Must send/get peer address with each packet
- **Can use UDP in connected mode**
  - connect assigns remote address
  - send/receiv syscalls, like sendto/recvfrom w/o last 2 args

Server interface

```c
struct sockaddr_in sin;
int s = socket (AF_INET, SOCK_STREAM, 0);
bzero (&sin, sizeof (sin));
sin.sin_family = AF_INET;
sin.sin_port = htons (9999);
sin.sin_addr.s_addr = htonl (INADDR_ANY);
bind (s, (struct sockaddr *) &sin, sizeof (sin));
listen (s, 5);
for (;;) {
    socklen_t len = sizeof (sin);
    int cfd = accept (s, (struct sockaddr *) &sin, &len);
    /* cfd is new connection; you never read/write s */
    do_something_with (cfd);
    close (cfd);
}
```

Socket implementation

- **Need to implement layering efficiently**
  - Add UDP header to data, Add IP header to UDP packet, …
  - De-encapsulate Ethernet packet so IP code doesn’t get confused by Ethernet header
- **Don’t store packets in contiguous memory**
  - Moving data to make room for new header would be slow
- **BSD solution: mbufs** [Leffler]
  (Note [Leffler] calls m_nextpkt by old name m_act)
  - Small, fixed-size (256 byte) structures
  - Makes allocation/deallocation easy (no fragmentation)
- **Mbufs working example for this lecture**
  - Linux uses sk_buffs, which are similar idea

mbuf details

- **Pkts made up of multiple mbufs**
  - Chained together by m_next
- **Chains linked w. m_nextpkt**
  - Linked chains known as queues
    - E.g., device output queue
- **Most mbufs have 108 data bytes**
  - First in chain has pkt header
- **Cluster mbufs have more data**
  - ext header points to data
    - Up to 2 KB not collocated w. mbuf
    - m_dat not used
  - m_flags or of various bits
    - E.g., if cluster, or if pkt header used

Adding/deleting data w. mbufs

- **m_data always points to start of data**
  - Can be m_dat, or ext_buf for cluster mbuf
  - Or can point into middle of that area
- **To strip off a packet header (e.g., TCP/IP)**
  - Increment m_data, decrement m_len
- **To strip off end of packet**
  - Decrement m_len
- **Can add data to mbuf if buffer not full**
- **Otherwise, add data to chain**
  - Chain new mbuf at head/tail of existing chain
mbuf utility functions

- `mbuf *m_copym(mbuf *m, int off, int len, int wait);`
  - Creates a copy of a subset of an mbuf chain
  - Doesn't copy clusters, just increments reference count
  - `wait` says what to do if no memory (wait or return NULL)
- `void m_adj(struct mbuf *mp, int len);`
  - Trims |len| bytes from head or (if negative) tail of chain
- `mbuf *m_pullup(struct mbuf *n, int len);`
  - Puts first |len| bytes of chain contiguously into first mbuf
  - Example: Ethernet packet containing IP datagram
    - Trim Ethernet header with `m_adj`
    - Call `m_pullup(n, sizeof(ip_hdr))`
    - Access IP header as regular C data structure

Socket implementation

- Each socket fd has associated socket structure with:
  - Send and receive buffers
  - Queues of incoming connections (on listen socket)
  - A protocol control block (PCB)
  - A protocol handle (struct protosw *)
- PCB contains protocol-specific info. E.g., for TCP:
  - Pointer to IP TCB with source/destination IP address and port
  - Information about received packets & position in stream
  - Information about unacknowledged sent packets
  - Information about timeouts
  - Information about connection state (setup/teardown)

protosw structure

- **Goal**: abstract away differences between protocols
  - In C++, might use virtual functions on a generic socket struct
  - Here just put function pointers in protosw structure
- **Also includes a few data fields**
  - `type, domain, protocol` – to match socket syscall args, so know which protosw to select
  - `flags` – to specify important properties of protocol
- **Some protocol flags**:
  - `ATOMC` – exchange atomic messages only (like UDP, not TCP)
  - `ADDR` – address given w. messages (like unconnected UDP)
  - `CONNREQUIRED` – requires connection (like TCP)
  - `WANTRCV` – notify socket of consumed data (e.g., so TCP can wake up a sending process blocked by flow control)

protosw functions

- `pr_slowtimo` – called every 1/2 sec for timeout processing
- `pr_drain` – called when system low on space
- `pr_input` – takes mbuf chain of data to be read from socket
- `pr_output` – takes mbuf chain of data written to socket
- `pr_usrreq` – multi-purpose user-request hook
  - Used for bind/listen/accept/connect/disconnect operations
  - Used for out-of-band data
  - Various other control operations

Network interface cards

- Each NIC driver provides an ifnet data structure
  - Like protosw, tries to abstract away the details
- **Data fields**:
  - Interface name (e.g., “eth0”)
  - Address list (e.g., Ethernet address, broadcast address, …)
  - Maximum packet size
  - Send queue
- **Function pointers**
  - `if_output` – prepend header, enqueue packet
  - `if_start` – start transmitting queued packets
  - Also ioctl, timeout, initialize, reset

Routing

- An OS must route all transmitted packets
  - Machine may have multiple NICs plus “loopback” interface
  - Which interface should a packet be sent to, and what MAC address should packet have?
- **Routing is based purely on the destination address**
  - Even if host has multiple NICs w. different IP addresses
    - (Though some packet filters can redirect based on source IP)
- OS maintains routing table
  - Maps IP address & prefix-length → next hop
- **Use radix tree for efficient lookup**
  - Branch at each node in tree based on single bit of target
  - When you reach leaf, that is your next hop
- **Most OSes provide packet forwarding**
  - Received packets for non-local address routed out another if