CSCI-1680
Layering and Encapsulation

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Administrivia

• Sign and hand in Collaboration Policy
• Signup for Snowcast milestone
  – Thursday from 8pm to 11pm
  – See Piazza for links
• Github
Today

• Review
  – Switching, Multiplexing

• Layering and Encapsulation
• Intro to IP, TCP, UDP
• Performance Metrics
A Taxonomy of networks

Communication Network

Switched Communication Network

Circuit-Switched Communication Network

Packet-Switched Communication Network

Point-to-point network

Broadcast Communication Network

Datagram Network

Virtual Circuit Network

A hybrid of circuits and packets; headers include a “circuit identifier” established during a setup phase
Circuit Switching

• Guaranteed allocation
  – Time division / Frequency division multiplexing
• Low space overhead
• Easy to reason about

• Failures: must re-establish connection
  – For any failures along path
• Overload: all or nothing
  – No graceful degradation
• Waste: allocate for peak, waste for less than peak
• Set up time
Packet Switching

• Break information in small chunks: *packets*
• Each packet forwarded independently
  – Must add metadata to each packet
• **Allows statistical multiplexing**
  – High utilization
  – Very flexible
  – Fairness not automatic
  – Highly variable queueing delays
  – Different paths for each packet
Traceroute map of the Internet, ~5 million edges, circa 2003. opte.org
Managing Complexity

• *Very* large number of computers
• Incredible variety of technologies
  – Each with very different constraints
• No single administrative entity
• Evolving demands, protocols, applications
  – Each with very different requirements!

• How do we make sense of all this?
Layering

- Separation of concerns
  - Break problem into separate parts
  - Solve each one independently
  - Tie together through common interfaces: abstraction
  - Encapsulate data from the layer above inside data from the layer below
  - Allow independent evolution

<table>
<thead>
<tr>
<th>Layer</th>
<th>Application</th>
<th>TCP</th>
<th>UDP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Protocol</td>
<td></td>
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</tr>
<tr>
<td>TCP</td>
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<tr>
<td>IP</td>
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<tr>
<td>Link Layer</td>
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</tbody>
</table>
Layers

- **Application** – what the users sees, *e.g.*, HTTP
- **Presentation** – crypto, conversion between representations
- **Session** – can tie together multiple streams (*e.g.*, audio & video)
- **Transport** – demultiplexes, provides reliability, flow and congestion control
- **Network** – sends packets, using routing
- **Data Link** – sends frames, handles media access
- **Physical** – sends individual bits
OSI Reference Model

End host

Application
Presentation
Session
Transport
Network
Data link
Physical

Application
Presentation
Session
Transport
Network
Data link
Physical

One or more nodes within the network
Layers, Services, Protocols

Layer N+1

Service: abstraction provided to layer above
API: concrete way of using the service

Layer N

Protocol: rules for communication within same layer

Layer N uses the services provided by N-1 to implement its protocol and provide its own services

Layer N-1
<table>
<thead>
<tr>
<th>Layer</th>
<th>Service Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Application</td>
<td>Service: user-facing application. Application-defined messages</td>
</tr>
<tr>
<td>Transport</td>
<td>Service: multiplexing applications Reliable byte stream to other node (TCP), Unreliable datagram (UDP)</td>
</tr>
<tr>
<td>Network</td>
<td>Service: move packets to any other node in the network IP: Unreliable, best-effort service model</td>
</tr>
<tr>
<td>Link</td>
<td>Service: move frames to other node across link. May add reliability, medium access control</td>
</tr>
<tr>
<td>Physical</td>
<td>Service: move bits to other node across link</td>
</tr>
</tbody>
</table>
Protocols

• **What do you need to communicate?**
  – Definition of message formats
  – Definition of the semantics of messages
  – Definition of valid sequences of messages
    • Including valid timings

• **Also, who do you talk to? …**
Addressing

• Each node typically has a unique* name
  – When that name also tells you how to get to the node, it is called an address
• Each layer can have its own naming/addressing
• Routing is the process of finding a path to the destination
  – In packet switched networks, each packet must have a destination address
  – For circuit switched, use address to set up circuit
• Special addresses can exist for broadcast/multicast/anycast

* within the relevant scope
Challenge

- Decide on how to factor the problem
  - What services at which layer?
  - What to leave out?
  - More on this later (End-to-end principle)

- For example:
  - IP offers pretty crappy service, even on top of reliable links… why?
  - TCP: offers reliable, in-order, no-duplicates service. Why would you want UDP?
IP as the Narrow Waist

• Many applications protocols on top of UDP & TCP
• IP works over many types of networks
• This is the “Hourglass” architecture of the Internet.
  – If every network supports IP, applications run over many different networks (e.g., cellular network)
Network Layer: Internet Protocol (IP)

• Used by most computer networks today
  – Runs over a variety of physical networks, can connect Ethernet, wireless, modem lines, etc.

• Every host has a unique 4-byte IP address (IPv4)
  – E.g., www.cs.brown.edu → 128.148.32.110
  – The network knows how to route a packet to any address

• Need more to build something like the Web
  – Need naming (DNS)
  – Interface for browser and server software (next lecture)
  – Need demultiplexing within a host: which packets are for web browser, Skype, or the mail program?
Inter-process Communication

- Talking from host to host is great, but we want abstraction of inter-process communication
- Solution: *encapsulate* another protocol within IP
Transport: UDP and TCP

- **UDP and TCP most popular protocols on IP**
  - Both use 16-bit *port* number & 32-bit IP address
  - Applications *bind* a port & receive traffic on that port
- **UDP – User (unreliable) Datagram Protocol**
  - Exposes packet-switched nature of Internet
  - Sent packets may be dropped, reordered, even duplicated (but there is corruption protection)
- **TCP – Transmission Control Protocol**
  - Provides illusion of reliable ‘pipe’ or ‘stream’ between two processes anywhere on the network
  - Handles congestion and flow control
Uses of TCP

• Most applications use TCP
  – Easier to program (reliability is convenient)
  – Automatically avoids congestion (don’t need to worry about taking down the network)

• Servers typically listen on well-know ports:
  – SSH: 22
  – SMTP (email): 25
  – Finger: 79
  – HTTP (web): 80
Transport: UDP and TCP

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Internet Layering

- **Strict layering not required**
  - TCP/UDP “cheat” to detect certain errors in IP-level information like address
  - Overall, allows evolution, experimentation
• We didn’t cover these in class, but these concepts about the socket API are useful for, and exercised by, the Snowcast assignment!
Using TCP/IP

• How can applications use the network?
• Sockets API.
  – Originally from BSD, widely implemented (*BSD, Linux, Mac OS X, Windows, …)
  – Important do know and do once
  – Higher-level APIs build on them
• After basic setup, much like files
Sockets: Communication Between Machines

• Network sockets are file descriptors too

• 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/recv

• Stream sockets: bi-directional pipes
  – With IP, gives you TCP
  – Bytes written on one end read on another
  – Reads may not return full amount requested, must re-read
# System calls for using TCP

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

- This call to bind is optional, connect can choose address & port.
Socket Naming

• Recall how TCP & UDP name communication endpoints
  – IP address specifies host (128.148.32.110)
  – 16-bit port number demultiplexes within host
  – Well-known services listen on standard ports (*e.g.* ssh – 22, http – 80, mail – 25, see `/etc/services` for list)
  – Clients connect from arbitrary ports to well known ports

• A connection is named by 5 components
  – Protocol, local IP, local port, remote IP, remote port
  – TCP requires connected sockets, but not UDP
Dealing with Address Types

• All values in network byte order (Big Endian)
  – htonl(), htons(): host to network, 32 and 16 bits
  – ntohl(), ntohs(): network to host, 32 and 16 bits
  – Remember to always convert!

• All address types begin with family
  – sa_family in sockaddr tells you actual type

• Not all addresses are the same size
  – e.g., struct sockaddr_in6 is typically 28 bytes, yet
    generic struct sockaddr is only 16 bytes
  – So most calls require passing around socket length
  – New sockaddr_storage is big enough
Client Skeleton (IPv4)

```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));
while ((n = read (s, buf, sizeof (buf))) > 0)
    write (1, buf, n);
```
Server Skeleton (IPv4)

```c
int s = socket (AF_INET, SOCK_STREAM, 0);
struct sockaddr_in sin;
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);
}
```
Using UDP

• Call socket with SOCK_DGRAM, bind as before
• New calls for sending/receiving individual packets
  – sendto(int s, const void *msg, int len, int flags, const struct sockaddr *to, socklen_t tolen);
  – recvfrom(int s, void *buf, int len, int flags, struct sockaddr *from, socklen_t *fromlen);
  – Must send/get peer address with each packet

• Example: udpecho.c

• Can use UDP in connected mode (Why?)
  – connect assigns remote address
  – send/recv syscalls, like sendto/recvfrom w/o last two arguments
Uses of UDP Connected Sockets

• Kernel demultiplexes packets based on port
  – Can have different processes getting UDP packets from different peers

• Feedback based on ICMP messages (future lecture)
  – Say no process has bound UDP port you sent packet to
  – Server sends port unreachable message, but you will only receive it when using connected socket
Serving Multiple Clients

• A server may block when talking to a client
  – Read or write of a socket connected to a slow client can block
  – Server may be busy with CPU
  – Server might be blocked waiting for disk I/O

• Concurrency through multiple processes
  – Accept, fork, close in parent; child services request

• Advantages of one process per client
  – Don’t block on slow clients
  – May use multiple cores
  – Can keep disk queues full for disk-heavy workloads
Threads

• One process per client has disadvantages:
  – High overhead – fork + exit ~100μsec
  – Hard to share state across clients
  – Maximum number of processes limited

• Can use threads for concurrency
  – Data races and deadlocks make programming tricky
  – Must allocate one stack per request
  – Many thread implementations block on some I/O or have heavy thread-switch overhead

Rough equivalents to fork(), waitpid(), exit(), kill(), plus locking primitives.
Non-blocking I/O

• `fcntl` sets O_NONBLOCK flag on descriptor
  ```c
  int n;
  if ((n = fcntl(s, F_GETFL)) >= 0)
      fcntl(s, F_SETFL, n | O_NONBLOCK);
  ```

• Non-blocking semantics of system calls:
  – read immediately returns -1 with errno EAGAIN if no data
  – write may not write all data, or may return EAGAIN
  – connect may fail with EINPROGRESS (or may succeed, or may fail with a real error like ECONNREFUSED)
  – accept may fail with EAGAIN or EWOULDBLOCK if no connections present to be accepted
How do you know when to read/write?

struct timeval {
    long   tv_sec;    /* seconds */
    long   tv_usec;   /* and microseconds */
};

int select (int nfds, fd_set *readfds, fd_set *writefds,
            fd_set *exceptfds, struct timeval *timeout);

FD_SET(fd, &fdset);
FD_CLR(fd, &fdset);
FD_ISSET(fd, &fdset);
FD_ZERO(&fdset);

• Entire program runs in an event loop
Event-driven servers

• Quite different from processes/threads
  – Race conditions, deadlocks rare
  – Often more efficient

• But…
  – Unusual programming model
  – Sometimes difficult to avoid blocking
  – Scaling to more CPUs is more complex
Coming Up

• Next class: Physical Layer
• Thu 13th: Snowcast milestones