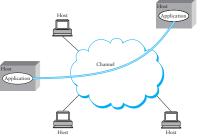
Outline

- Networking overview
- 2 Systems issues
- OS networking facilities
- Implementing networking in the kernel

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Inter-process communication



- Want abstraction of inter-process (not just inter-node) communication
- · Goal: two different applications, running on different computers, can exchange data as if they had a pipe between them.

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Physical Layer

- 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

 - Shared transmission medium networks (e.g., coax, radio):

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- 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

SF $\longrightarrow \sim 15 \text{ msec}$ NYC Moore's law does not apply!

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.

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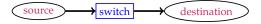
The 7-Layer and 4-Layer Models

	OSI	TCP/IP
7	Application	Applications (FTP, SMTP, HTTP, etc.)
6	Presentation	
5	Session	
4	Transport	TCP (host-to-host)
3	Network	IP
2	Data link	Network access (usually Ethernet)
1	Physical	

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Link Layer, Indirect Connectivity

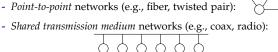
- Rarely have direct physical connection to destination
- Instead, communications usually "hop" through multiple devices



- Allows links and devices to be shared for multiple purposes
- Must determine which bits are part of which messages intended for which destinations

Packet switched networks

- Pack a bunch of bytes together intended for same destination
- Slap a header on packet describing where it should go





Link Layer: Ethernet

- Originally designed for shared medium (coax), now generally not shared medium (switched)
- Vendors give each device a unique 48-bit MAC address - Specifies which card should receive a packet
- Ethernet switches can scale to switch local area networks (thousands of hosts), but not much larger



- Preamble helps device recognize start of packet
- CRC allows card to ignore corrupted packets
- Body up to 1,500 bytes for same destination
- All other fields must be set by sender's OS (NIC cards tell the OS what the card's MAC address is, Special addresses used for broadcast/multicast)

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Network Layer: 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 there is address shortage)
- E.g., 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



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

Why Ethernet is insufficient

- Ethernet Limits
 - 2,500m diameter
 - 100 nodes
- Can bridge multiple Ethernets
 - 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

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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

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Principle: Packet Switching

- A packet is a self contained unit of data which contains information necessary for it to reach its destination
- Packet switching: independently for each arriving packet, compute its outgoing link. If the link is free, send it. Otherwise, queue it for later (or drop).
 - Makes forwarding very simple
 - Allows simple sharing of links

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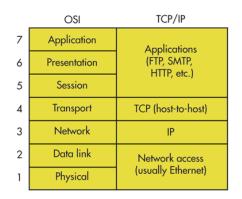
Bridge Port 2 - First time you see destination address, send packet to all segments

← Port 1

Principle: Layering

- Break system functionality into a set of components
- Each component ("layer") provides a well-defined service
- Each layer uses only the service of the layer below it
- Layers communicate sequentially with the layers above or below

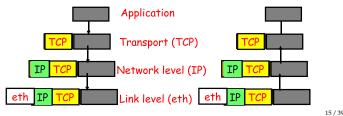
The 7-Layer and 4-Layer Models



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Principle: Encapsulation

- Stick packets inside packets
- How you realize packet switching and layering in a system
 - 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 (not always true)



Unreliability of IP

- Network does not deliver packets reliably
 - May drop packets, reorder packets, delay packetsMay 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?

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 - 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)

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Performance: 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 # bytes in flight ≥ bandwidth×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 & response time critical for good network utilization

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A little bit about TCP

- Want to save network from congestion 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

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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)

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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

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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/recv
- 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

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System calls for using TCP

Client Server socket – make socket bind – assign address listen – listen for clients

bind* - assign address
connect - connect to listening socket

accept – accept connection

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

Client interface

```
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 = hton1 (IP_ADDRESS);
connect (s, (sockaddr *) &sin, sizeof (sin));
```

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Server interface

```
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);
```

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Using UDP

• Call socket with SOCK_DGRAM, bind as before

New system calls for sending individual packets

- 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/recv syscalls, like sendto/recvfrom w/o last 2 args

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I Networking overview

- m_next m_nextpkt m_len m_data m_type m_type m_flags pkt.len pkt.rcvif ext.buf 108 ext.free bytes m dat ext.size optional
- mbuf details
 - Pkts made up of multiple mbufs

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- Chained together by m_next
- Such linked mbufs called chains
- Chains linked w. m_nextpkt
 - Linked chains known as queues
 - E.g., device output queue
- Most mbufs have ≈230 data bytes (depends on pointers)
 - 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 ³⁹

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);
 - Trim |len| bytes from head or (if negative) tail of chain
- mbuf *m_pullup(struct mbuf *n, int len);
 - Put first len bytes of chain contiguously into first mbuf
- Example: Ethernet packet containing IP datagram
 - Trim Ethernet header w. $m_{-}adj$
 - Call m_pullup (n, sizeof (ip_hdr));
 - Access IP header as regular C data structure

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)
- BSD Mbufs working example for this lecture
 - Linux uses sk_buffs, which are similar idea

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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

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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 w. 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

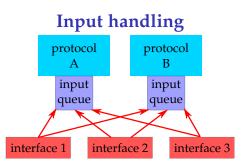
- *type, domain, protocol* to match socket syscall args, so know which protosw to select
- flags to specify important properties of protocol

Some protocol flags:

- ATOMIC exchange atomic messages only (like UDP, not TCP)
- ADDR address given w. messages (like unconnected UDP)
- CONNREQUIRED requires connection (like TCP)
- WANTRCVD notify socket of consumed data (e.g., so TCP can wake up a sending process blocked by flow control)

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- NIC driver determines packet protocol
- Enqueues packet for appropriate protocol handler
 - If queue full, drop packet (can create livelock [Mogul])
- **Posts "soft interrupt" for protocol-layer processing** - Runs at lower priority than hardware (NIC) interrupt
 - \ldots but higher priority than process-context kernel code

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 OSes have features to redirect based on source IP)

• OS maintains routing table

- Maps IP address & prefix-length \rightarrow 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

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