High-Level UDP / TCP/IP

Network Programming

Overview

- UDP (User Datagram Protocol) is unreliable
 - Potentially delivered out of order (or not at all!)
 - Connectionless
- TCP (Transmission Control Protocol) provides reliability
 - In-order
 - Connection-oriented
- Both sit on top of another protocol

Internet Protocol (IP)

IP is the network layer

Essentially, responsible for host to host packet delivery (routing)

Translation between multiple data link protocols such as ethernet, wifi, etc.

IP Datagrams

 IP provides connectionless, unreliable delivery of IP datagrams

 Connectionless: All datagrams are independent of each other

 Unreliable: no guarantee datagrams are even delivered, let alone ordered

IP Addresses

- IP at network layer but must be able to talk to other devices on different mediums! (ex: iPhone to wired server)
 - Also why MAC at different layer than IP

- Must provide some degree of network information
 - Allows for efficient routing of datagrams

IP Addresses

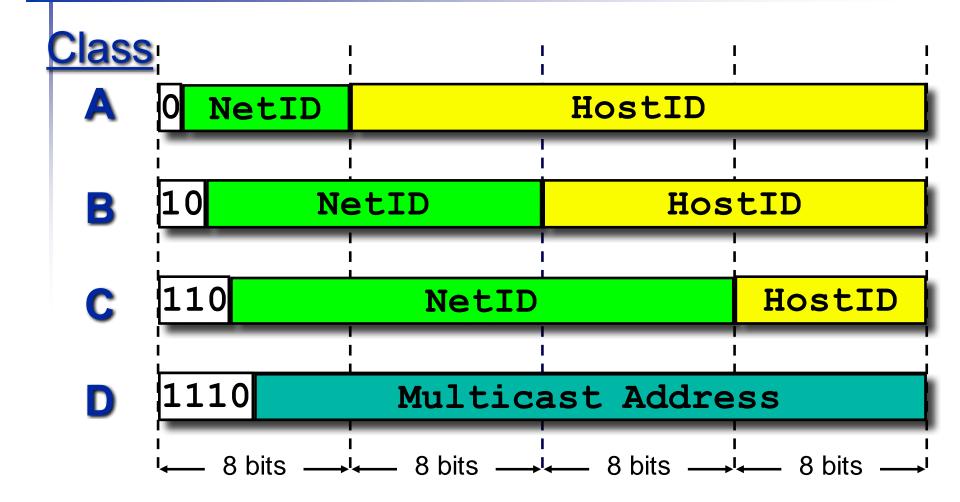
IP addresses are logical addresses

Four octets (32 bits) [IPv4]

Includes a network portion and a host portion

All IPs must be unique

IP Address Format



Ramifications

- Class A: 128 network IDs
 - 16M host IDs per network ID

- Class B: 16K network IDs
 - 64K host IDs per network ID

- Class C: 2M network IDs
 - 256 host IDs per network ID

Network / Host IDs

- Network IDs are assigned by a central authority
 - ICANN, IANA

 Host IDs are assigned locally by a systems administrator

Network ID and host ID are used for routing purposes

IP Address Format

IPs are often written in dotted decimal notation

- For example, 128.113.0.2 (<u>www.rpi.edu</u>)
 - **1**0000000.01110001.00000000.00000010

- RPI must have a class B address!
 - Leading digits are 10

Network / Host Addresses

Hosts aren't assigned addresses, their network interface is

Hosts may have multiple NICs

If the network addresses are the same, they share the network

Broadcast / Network Addresses

Broadcast address: host ID all 1's

 Broadcasts may be implemented however the underlying layer sees fit

 Network address: host ID all 0's, refers to entire network

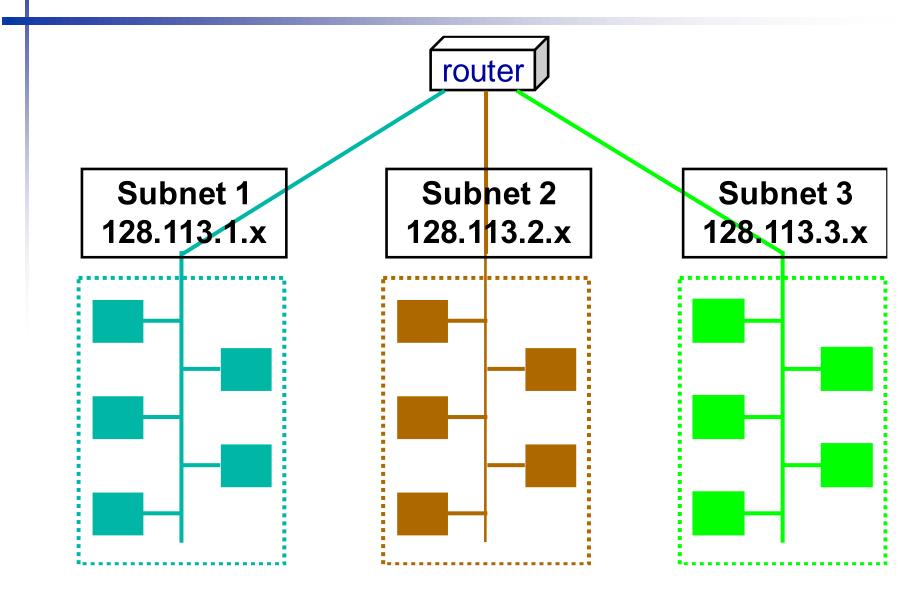
Subnet Addresses

 Organizations are able to further subdivide its available address space into "subnets"

- For example, clump nearby machines into their own subnet
 - Could also do this logically

10 NetID SubnetID HostID

Subnetting

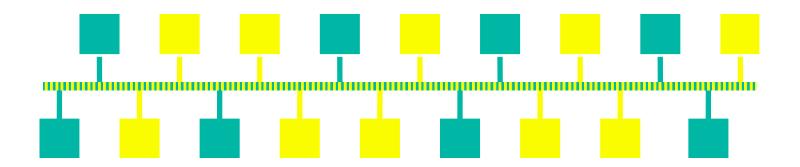


Subnetting

IP subnet broadcasts have host ID of all 1s

Subnets can simplify routing

Multiple subnets can share a single wire!



IP

- Connectionless delivery
 - Each datagram handled individually
- Unreliable
 - No guarantee
- Fragmentation + reassembly
 - Hardware MTU
- Routing

Error detection

IP Datagram

https://tools.ietf.org/html/rfc791#section-3.1

← 1 byte →		← 1 byte →	— 1 by	yte — 1 byte —	
VERS	HL	Service	Fragment Length		
Datagram ID			FLAG	Fragment Offset	
TTL		Protocol	He	eader Checksum	
Source Address					
Destination Address					
Options (if any)					
Data					

Datagram Fragmentation

Each fragment has same structure

 IP requires reassembly done at destination only, not at intermediate routers

 Any lost fragments require ICMP error message be sent and entire datagram discarded

TCP / UDP over IP

Application Layer Process Process TCP UDP Transport Layer Network Layer IP Data-Link Layer Hardware

UDP

- UDP is a transport protocol
 - Communication between two processes

 UDP uses IP to deliver datagrams to the proper host

Uses ports to provide additional specification

UDP Format

https://tools.ietf.org/html/rfc768

Source Port	Destination Port			
Length	Checksum			
Data				

TCP

TCP is a transport protocol

- In addition to all things provided by UDP, TCP provides:
 - Reliability
 - Full-duplex
 - Connection-oriented
 - Byte Stream

TCP Segments

The chunk of data that TCP requests IP to transmit is called a Segment

- Each segment contains:
 - Data bytes from byte stream
 - Control information identifying data bytes

TCP Segment Format

https://tools.ietf.org/html/rfc793#section-3.1

<u></u> 1 t	oyte —	— 1 byte —	← 1 byte ↓ 1 byte →			
Source Port			Destination Port			
	Sequence Number					
Request Number						
offset	Reser.	Control	Window			
Checksum			Urgent Pointer			
Options (if any)						
Data						

UDP Sockets

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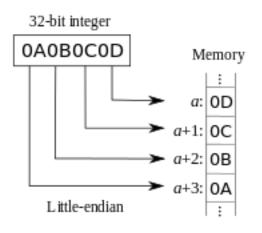
UDP Format

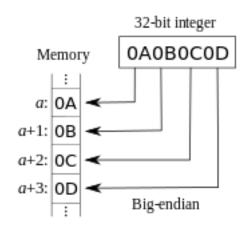
https://tools.ietf.org/html/rfc768

Source Port	Destination Port			
Length	Checksum			
Data				

But first... Endianness

 Some machines went big-endian (68K), others little-endian (x86)





From Gulliver's Travels

Network Issues

- When hosts exchange single-byte data types, no problem
 - What about 32-bit word?
 - Are these the same value?
 - 0x00000001 and 0x01000000
 - Problem!

Values sent from big-endian machine would be interpreted incorrectly on the little-endian machine!

Network Byte Order

- Network defines big-endian to be the byte order
 - May be different from host byte order

- Translation always required
 - Even on big-endian machines
 - How do you know what type of machine your code may be compiled on in the future?

Byte Order Functions

#include <netinet/in.h> uint16 t htons(uint16 t hs); uint32 t htonl(uint32 t hl); uint16 t ntohs(uint16 t ns); uint32 t ntohl(uint32 t nl);

Sockets

- Berkeley sockets implementation, originally from 4.2BSD (1983!)
 - Effectively became POSIX sockets

Building blocks for modern network-enabled programs

Simple API

socket

- #include <sys/socket.h>
- int socket(int domain, int type, int protocol);

- Just creates an endpoint, nothing more!
- domain typically PF_INET / AF_INET
- type: SOCK_[STREAM,DGRAM,RAW]
- protocol: just use 0 for system default for given domain / type

bind

- fd: must be returned by socket()
- sa: sockaddr containing IP / port
- len: length of passed-in sockaddr

Servers call bind upon startup

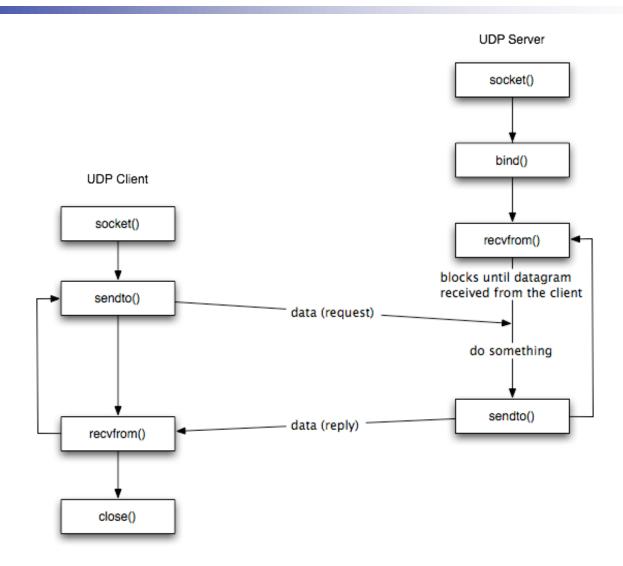
struct sockaddr_in

```
struct sockaddr in {
     uint8 t
                    sin len;
   sa family t
                    sin family;
   in port t
                   sin port;
   struct in addr sin addr;
                    sin zero[8];
   char
struct in addr {
   in addr t
                    s addr;
```

Typical usage

```
struct sockaddr in saddr;
/* Zero out the memory */
bzero(&saddr, sizeof(saddr));
saddr.sin family = PF INET;
saddr.sin port = htons(1234);
saddr.sin addr.s addr =
  htonl(INADDR ANY);
```

Typical UDP client/server



recvfrom / sendto

```
#include <sys/socket.h>
ssize t recvfrom(int fd, void
  *buf, size t nbytes, int flags,
  struct sockaddr *from,
  socklen t *len);
ssize t sendto(int fd, void *buf,
  size t nbytes, int flags,
  struct sockaddr *to, socklen t len);
```

Oddities

- Sending 0 bytes is completely fine
 - 8 byte UDP header (no data)

- recvfrom() can return 0
 - Different from TCP where a 0 means peer has closed connection

- Both functions can also be used w/ TCP
 - But why???

udpserv01.c

```
#include
            "unp.h"
int
main(int argc, char **argv)
{
   int
                        sockfd;
    struct sockaddr_in servaddr, cliaddr;
    sockfd = Socket(AF_INET, SOCK_DGRAM, 0);
    bzero(&servaddr, sizeof(servaddr));
    servaddr.sin_family = AF_INET;
    servaddr.sin_addr.s_addr = htonl(INADDR_ANY);
    servaddr.sin_port = htons(SERV_PORT);
    Bind(sockfd, (SA *) &servaddr, sizeof(servaddr));
    dg_echo(sockfd, (SA *) &cliaddr, sizeof(cliaddr));
}
```

dg_echo.c

```
#include "unp.h"
void
dg_echo(int sockfd, SA *pcliaddr, socklen_t clilen)
{
   int
        n;
   socklen_t len;
   char mesg[MAXLINE];
   for (;;) {
       len = clilen;
       n = Recvfrom(sockfd, mesg, MAXLINE, 0, pcliaddr, &len);
       Sendto(sockfd, mesg, n, 0, pcliaddr, len);
}
```

udpcli01.c

```
#include "unp.h"
int
main(int argc, char **argv)
{
    int
                        sockfd;
    struct sockaddr_in servaddr;
    if (argc != 2)
        err quit("usage: udpcli <IPaddress>");
    bzero(&servaddr, sizeof(servaddr));
    servaddr.sin_family = AF_INET;
    servaddr.sin_port = htons(SERV_PORT);
    Inet pton(AF INET, argv[1], &servaddr.sin addr);
    sockfd = Socket(AF_INET, SOCK_DGRAM, 0);
    dg_cli(stdin, sockfd, (SA *) &servaddr, sizeof(servaddr));
    exit(0);
}
```

dg_cli.c

```
#include
           "unp.h"
void
dg_cli(FILE *fp, int sockfd, const SA *pservaddr, socklen_t servlen)
    int n;
            sendline[MAXLINE], recvline[MAXLINE + 1];
    char
    while (Fgets(sendline, MAXLINE, fp) != NULL) {
        Sendto(sockfd, sendline, strlen(sendline), 0, pservaddr, servlen);
        n = Recvfrom(sockfd, recvline, MAXLINE, 0, NULL, NULL);
        recvline[n] = 0; /* null terminate */
        Fputs(recvline, stdout);
```

Not reliable in any way!

- What if messages get lost?
 - Client datagram?
 - Server datagram?

- How do we add reliability?
 - Timeouts (and retransmissions)
 - Sequence numbers
 - We'll have an assignment about this later in the semester

UDP connect()

- We can call connect() on a UDP socket
 - No longer able to use sendto() but rather write()/send()
 - Similarly for recvfrom(); replace with recv()/recvmsg()

May improve performance if communicating with same host repeatedly

UDP Clients: DNS

 Quicker, no three-way handshake required (we'll revisit this later)

 Connectionless means less burden on the nameservers (we may revisit DNS later)

 Loss isn't terrible, just send another query some time later

DNS Resource Records

A: hostname -> IPv4 address

AAAA: hostname -> IPv6 address

PTR: IP address -> hostname

MX: Mail Exchange

CNAME: canonical name

Skipping: Zeroconf/Bonjour

Builds on DNS (and other tech as well)

- SRV records added to DNS
 - https://tools.ietf.org/html/rfc2782

Designed to simplify networking

Skipping for now because configuring issues (avahi)