TCP/IP Issues



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A TCP/IP Tutorial

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- Fundamental networking protocols of the Internet
- TCP: Transmission Control Protocol (RFC 793, September 1981)
- IP: Internet Protocol (RFC 791, September 1981)
- There have been improvements, but the fundamental architecture remains
- N.B.: HTTP/HTTPS have an outsize role; more on this later
- We'll start with a brief, simplified overview of TCP, covering only the parts necessary for this class

- Support many applications—most older networks had baked-in (and limited) applications
- Support many kinds of networking hardware—most older networks were tied to particular link types
- Support many different operating systems—most older networks were vendor-proprietary
- No central core network

- A *network* is two or more hosts on a common medium
- Example: Ethernet (many hosts), point-to-point fiber (two hosts)
- The Internet is sometimes called a *catenet*: multiple networks joined together
- Networks are connected by routers



- Basic principle: "IP over everything and everything over IP"
- Multiple applications use multiple transport protocols—but there's only one IP
- IP speaks to multiple network devices
- Developed under DARPA (Defense Advance Reseach Projects Agency) sponsorship; replacement for the old ARPANET

The "Hourglass" Model



7	Application	
6	Presentation	
5	Session	
4	Transport: TCP, UDP, etc.	
3	Network: IP	
2	Link: WiFi, Ethernet, etc.	
1	Physical: radio, fiber, etc	

The Real Network Stack?

9	Political	← YOU ARE HERE
8	Financial	
7	Application	
6	Presentation	
5	Session	
4	Transport: TCP, UDP, etc.	
3	Network: IP	
2	Link: WiFi, Ethernet, etc.	
1	Physical: radio, fiber, etc	

Application Does what you really want, e.g., web TCP Presents a reliable byte stream to the applications IP Gets packets from "here" to "there" Link, Physical Deals with the hardware

- The underlying networks provide a possibly unreliable "datagram" sevice
- Unreliable: packets may be dropped, damaged, duplicated, reordered
- Packets are forwarded to the next hop to their eventual destination—or dropped if not deliverable
- Packets may be dropped because of network congestion
- Very little concern for the correctness of any packet
- *Stateless* forwarding—what happens with a packet does not affect what happens to the next packet
- Note: this is the *service model*—implementations can behave differently to optimize things if they wish

- A single message
- No call setup required; every message contains a source and destination
- Note: there are related terms: frame, packet, segment, etc. I'll use packet and datagram interchangeably for all of these concepts, but a networking course would be more precise

IP: The Internet Protocol

0	4	8	16			3	31		
Version	Header Length	Service Parameters		Length					
	Pack	et ID	0	D F	M F	Fragment Offset			
Time to Live Protocol			Header Checksum						
Source Address									
Destination Address									

Version The originally deployed Internet Protocol was Version 4. Version 6 is (very slowly!) being rolled out; I'll say very little more about it

Time to Live A hop count for packets, to prevent infinite forwarding loops

Protocol The next protocol: TCP, UDP, etc.

Header Checksum Validate the correctness of the IP header and only the IP header. Probably a mistake to have included it.

Source Address Where the packet came from

Destination Address Where it's going

Other This isn't a networking class; I won't discuss them...

- IP addresses are 32-bit numbers (for IPv6, 128-bit)
- They in fact have some structure, but that's for a later lecture
- Most IP addresses must be globally unique. Some (so-called "RFC 1918" addresses) are for local use and are translated at the site boundary to a global address

A Network Diagram



- A process on host A wants to send a packet to host B
- The IP layer on A examines the destination address and decides which node (generally a router) is the best next hop
- (How that decision is made will be covered in another lecture)
- It asks its link layer to send the packet there
- The next hop receives the packet via its IP layer. The IP layer either accepts it locally (and sends it to TCP) or forwards it another hop
- Eventually, it arrives

- Hosts are trying to send to IP addresses
- Link layers often have their own, very different addresses: for Ethernet and WiFi, these are 48-bit numbers. A mapping is necessary
- The host sends a *broadcast messsage* using the Address Resolution Protocol (ARP)
- All nodes on the local network receive it; the proper receiving machine replies with "here is my link-layer address"

- The purpose of TCP is to convert an unreliable set of packets to a reliable, unstructured byte stream
- That is: IP hands possibly-damaged packets to TCP. TCP handles ordering, damage detection, and retransmission
- TCP also separates IP packets into multiple connections
- In other words: IP deals with unreliable datagrams; TCP produces a set of reliable *connections* (sometimes call *circuits*)

TCP: The Transmission Control Protocol

0 8 1						16	24 31				
Source Port						8 16 24 31 Port Destination Port Sequence Number Sequence Number Colspan="4">R S F N N Window Sum Urgent pointer Options, if any					
Sequence Number											
Acknowledgement Number									Number		
Data offset	Reserved $\begin{bmatrix} c & e & v & A & P & R & S & F \\ w & c & e & G & K & H & T & N & N \\ c & R & e & G & K & H & T & N & N & Window & C & C & C & C & C & C & C & C & C & $										
Checksum									Urgent pointer		
Options, if any											
User Data (if any)											

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Source Port Part of the connection identifier

Destination Port Part of the connection identifier

Sequence Number The sequence number of the first data byte in this packet

Acknowledgment Number The sequence number of the last byte succesfully received

- ACK The Acknowledgment Number field is valid
- RST Reset this connection
- SYN Technically, "synchronize sequence numbers"; more intuitively, part of creating a connection
- FIN "Finish", i.e., start tearing down this circuit

Checksum Error detection—is this packet correct?

- A TCP server *listens* on a given *port number*
- Most of the time, this is a "well-known", i.e., standardized number: HTTP is port 80, SMTP (for email) is port 25, HTTPS is port 443, etc.
- (To a first approximation, you can tell what a connection is used for from its server port number)
- A TCP client is (generally) assigned a random, unused port number by the kernel; it attempts to *connect* to a server by the server's IP address and port number
- The (source address, source port, destination address, destination port) is the *connection identifier*
- Multiple connections can exist between a single pair of hosts: the client's port number will be different

- The client sends a packet to the server with the SYN bit set, saying "this is the sequence number of my first byte"
- This creates client-side state
- The server's reply has the SYN bit set (and "this is the sequence number of my first byte"), the ACK bit plus an acknowledgment of the client's initial sequence number
- Note that at this point, the server has to create state for this half-opened connection
- The client's reply has only the *ACK* bit set, plus n acknowledgment of the server's initial sequence number
- This is called the three-way handshake, which takes 1.5 round trips
- The connection is not fully open until the server receives this last message

The Three-Way Handshake



- When TCP receives a packet, it validates the checksum
- If the checksum is invalid, i.e., if the packet is damaged, it is silently dropped
- No, TCP doesn't send negative acknowledgments
- If the sequence number is for the next byte expected, the receiving machine sends an *ACK* packet, probably with no data, but with an acknowledgment number of the next expected byte
- If there's a "hole", it silently saves the received packet until the hole is filled
- Senders wait a certain amount of time for acknowledgments. If they don't get one, the packet is retransmitted.

• Why doesn't the IP layer drop damaged packets?



- Why doesn't the IP layer drop damaged packets?
- IP could check (and many link layers do check)—but that's redundant
- TCP has to check anyway
- UDP might not want a check-think OFB encryption
- This is the end-to-end principle
- Worth noting: because most links are very reliable (and many have their own checksums), very, very few packets are dropped because of TCP checksum issues

- When one side is done sending data, it emits a *FIN* packet
- This packet must be acknowledged by the other side—*FIN* bits count as bytes in sequence number space
- The other side must send an appropriate ACK packet
- The connection is completely torn down when both sides have received *ACK*s of their *FIN* bits
- Until then, both sides have to retain state

- When a host sends a packet and doesn't receive an acknowledgment, it waits a while and then retransmits it
- If too long goes by, it can declare the connection dead and notify the application
- If a host receives a packet and there is no matching connection, it immediately replies with a *RST*—reset—packet

- Per the last slide, a host can only tell that a connection is dead if it tries to send something that is never acknowledged
- What if a server is waiting for client input but it *never* comes?
- The server has to retain state the entire time!
- Answer: *keep-alives*
- A keep-alive is (effectively) a NOP packet whose sole purpose is to elicit a response. They're sent after a suitable period of silence—generally several minutes to several hours
- If the keep-alive packet does not receive an acknowledgment within the proper interval, the connection can be declared dead and the state discarded

- Some services, e.g., Voice over IP (VoIP) are better suited to a datagram model
- That is, they can tolerate occasional lost or damaged packets, but cannot deal with delays for retransmissions
- UDP provides the same datagram model as IP; the port numbers are used for demultiplexing
- The checksum is optional; if it's received as zero, the validation is skipped
- Note well: no kernel state is created by the receipt of a UDP packet; it's delivered to a listening application, if any, or discarded





- For a number of reasons, especially firewalls and encryption, many new protocols are layered on top of HTTPS
- Example: Dropbox, Apple's iMessage, and more
- In other words, HTTPS is not just web servers

Client struct sockaddr_in dest; /* (Somehow) put the IP address and port number of the server into "dest" */ fd = socket(AF_INET, SOCK_STREAM, 0); connect(fd, &dest, sizeof dest);

Server

struct sockaddr_in us, src;
/* (Somehow) put the service port
 number into "us"; zero the rest
 of it
*/
listenfd = socket(AF_INET, SOCK_STREAM, 0);
bind(listenfd, &us, sizeof us);
listen(listenfd, 5);
newfd = accept(listenfd, &src, sizeof src);
fork();

- Yes, this is a rapid, deep dive into complex material
- Unfortunately, most of the tutorials I've found online go into far more depth on stuff I've skipped, e.g., the TCP state diagram
- Most of the rest of the semester depends at least in part on this material

TCP State Transition Diagram



Basic Network Security



- The remaining lectures cover what is sometimes called "network security"
- Most of the time, that's wrong—there's nothing wrong with the network
- The network is the mechanism by which bad applications are attacked
- (Highway robbery generally does not involve stealing a piece of the road...)

Well, Sometimes it Does...

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NEWS								
Home US Election	Coronavir	us Video	World US	5 & Canada	UK Busi	ness		
Entertainment & Arts		In Pictures	;		World News	TV		
Health		Reality Ch	eck	1	Newsbeat			
World Africa Asia	Australia	Europe	Latin Ameri	ca Middle	e East			
Russia highway robbery: Official 'stole 50km road'								

Claim: The Designers of the Internet Ignored Security

There are many claims about the Internet being designed without regard to security. Most are incorrect.

Bad design **False**. Most of the problems are due to buggy applications, not the design of the Internet

Authentication **False**. The Internet couldn't mandate authentication, because different operating systems do things different ways—and why should you trust the sysadmin of a random remote site?

Encryption **Partly true**. It was assumed that encryption would be provided outboard to the hosts—but that assumed that link-layer and network-layer encryption was all that was necessary. This is wrong

Ignorance **Unknown**. We don't know what attacks were known or thought of back then. Some were definitley known to the NSA; others were known but (wrongly) considered infeasible.

Simple Attack: Eavesdropping

- We're worried about eavesdroppers—let's encrypt the link from A to N2
- What if the attacker targets a different link, e.g., *N2—N4*?
- Will traffic flow N10—N14—B or N10—N13—B?
- What if the operator of N2 is corrupt or the node is hacked?
- We need end-to-end encryption, not link encryption
- But link encryption is sometimes useful anyway



- Suppose that host D1 (somehow) steals D2's IP address
- How will A1 know if it's talking to the real D2?
- Or suppose that B4 answers an ARP query intended for B1? ARP isn't authenticated
- Remember when I talked about the standard cryptographic threat models...?



State Consumption

- Suppose that B4 is down and F impersonates it, sending a SYN packet to A2
- The SYN+ACK—the second message in the three-way handshake—will go towards host B4, but will be dropped
- A2 has created state for the half-open connection, but the connection will never fully open
- Eventually, it will time out, but that will take a few minutes
- Suppose that *F* does that repeatedly...



Questions?



(Great horned owl, Central Park, November 17, 2019)

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