Notes for ECE 46300 - Introduction To Computer Communication Networks

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Contents

Course Description

An introduction to the design and implementation of computer communication networks. The focus is on the concepts and the fundamental design principles that have contributed to the global Internet success. Topics include: digital transmission and multiplexing, protocols, MAC layer design (Ethernet/802.11), LAN interconnects and switching, congestion/flow/error control, routing, addressing, performance evaluation, internetworking (Internet) including TCP/IP, HTTP, DNS etc. This course will include one or more programming projects.

Computer Networks

The high-level question this course will answer is "how do computers reliably communicate?"

The answer is through computer networks, a group of interconnected nodes or computing devices that exchange data and resources with each other.

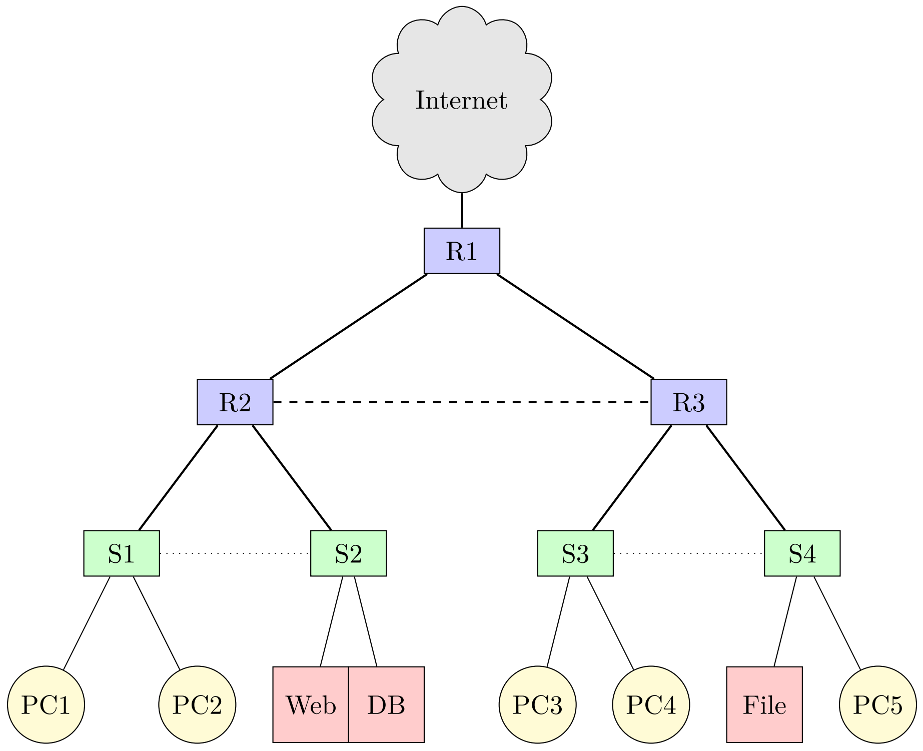


Figure 1: Computer Network

A computer network enables communication between users and their devices.

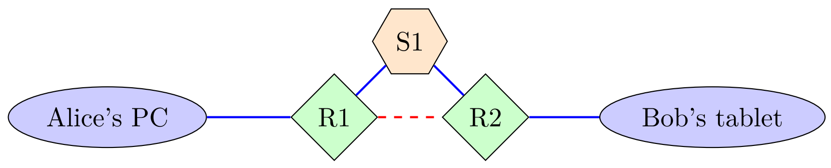


Figure 2: Simple Network

The first stop that most home devices take in connecting to other devices is the router. From there, the router can route data and find a path for Alice's PC to get to Bob's tablet. The core of any network is routers, which figure out the best way to get data from one device to another device.

This process can get complex, with cell tower connections, different ISPs, different edge devices, and more. Let's abstract the important elements of the network.

- Links: carry data from one endpoint to another
- End hosts: sitting at the edge of a network. Generate and receive data.
- Routers: forward data through the network.

Any network can be abstracted as a connection of links, end hosts, and routers. We can thus represent any computer network as a graph, in the mathematical sense, and apply all our graph algorithms to it.

A communication link can either be *full* or *half* duplex. In a half duplex, a link can carry data in only one direction at a time. In a full duplex, data transmission is bidirectional. If a link bandwidth is B, then a full-duplex link can carry B in both directions simultaneously.

In this class, all links are assumed to be full duplex unless otherwise stated.

Packets

In our abstraction of routers, we leave the problem of finding a path from end hosts unanswered. Since we can represent a computer network as a graph, a shortest-path algorithm like Dijkstra's is a relatively simple way to find a good path between Alice and Bob.

We wish to answer the question "how do users access shared network resources?" Assume no coordination between users and assume users initiate access.

There are two ways to answer this:

- Reserving resources (circuit switching).
- On-demand (packet switching)

Circuit Switching

The idea at the core of this method is that if Alice wants to communicate with Bob, then they reserve a path through the network to do that. The reserved path is called a *circuit.* They then send data along the reserved path.

A pro of circuit switching is that users are guaranteed to have a path through the network. No queuing, no waiting. The routers along the path also don't need to make any decisions in real time. They know the path in advance.

A con is that a resource could be reserved but not used. When users don't coordinate, circuit switching leads to very inefficient use of resources. Another con is that circuits need to be set up and torn down. Imagine the overhead of old telephone networks, where workers had to manually connect telephone wires from one caller to another.

In classical circuit switching, all the resources along the path are reserved for a single circuit and there is no path sharing amongst multiple circuits. In virtual circuit switching, each circuit reserves a subset of resources along its path. It's still reservation based, but two or more circuits can share a same resource (e.g. if two users want to send data that takes up half the bandwidth of a connection, they can both send it at once).

Packet Switching

To overcome the shortcomings of circuit switching, packet switching was invented. The way this works is data is broken into small units called *packets.* You send packets over the network whenever you have them and trust the routers to figure out the path your packets take on the fly.

Each packet needs to have metadata that describes things like the destination of the packets, the order of the packets, and so on.

A pro of packet switching is that we have much better resource utilization. Also, there's no overhead of circuit setup and teardown.

A con of packet switching is that there's no guarantee that a given user will have the resources to send their data. Now we also have overhead from the packet headers, and routers now need to process packet headers and find paths on the fly.

Although both have advantages and disadvantages, modern networks almost exclusively use packet switching and unless otherwise specified all networks in this class are assumed to be packet switched.

Packet Headers

Packet headers require, at least, the following:

- Destination address, used by network to send packet to destination
- Destination port, used by network stack at the destination for application multiplexing
- Source address, used by network to send packet back to source
- Source port, used by network stack at the source for application multiplexing

We'll learn more about the OSI model in a later section, but for now suffice to say that each layer has its own header that it stacks onto and strips from user data, starting with the application layer attaching its own header to user data, to the transport layer sticking a header on, and so on. As the package travels from host to destination the layers get added, and then stripped off in the order they were added.

Internet Architecture

We need to solve these problems:

- Name and addressing: identifiers for network nodes
- Destination discovery: finding the destination address
- Forwarding: sending received data to the next hop (neighbor)
- Routing: finding the path from source to destination
- Reliability: handling failures, packet drops, packet corruption
- Application multiplexing: delivering data from multiple host applications to the network and vice versa

Name and Addressing

Name is the human-readable name for each node (e.g., URL www.google.com). Address is where node is located (e.g., IP address 172.21.4.110).

Destination discovery

When you go to a web page, you type the name in your browser. You need to get the address still. The way names are resolved into addresses is via the Domain Name System (DNS), a set of global servers that maintain the mappings between host names and addresses. When you type a URL, the browser contacts a DNS server to get the address.

Forwarding

A router has many ports. Each port in a router acts as both input and output, i.e., you can both send and receive packets on each port simultaneously. When a packet arrives at a router, the router looks at the destination address in the header. Based on destination address, the router consults the routing table, determines the right output port, and sends the packet to that port's queue. For each output port in parallel, when the port is free, the router picks a packet from the corresponding output queue in some order (e.g. FIFO) and sends the packet over the output port.

Routing

How do a network of routers collectively find a path between each source and destination host? The answer is a routing protocol, a distributed algorithm that runs independently at each router.

Reliability

The goal of reliability is to ensure every packet sent will eventually reach the destination uncorrupted. Unfortunately, routers can drop packets or packets can get corrupted. The idea behind all reliability protocols is after sending data, await an ACK from the destination. If ACK was received, everything is good! If not, resend the data.

Application Multiplexing

Each app that communicates over a network needs some subset of basic functionalities, like a reliability protocol. We could ask each app developer to implement the protocol themselves, but this is burdensome on the developer and prone to errors. Instead, on modern computers, there is a common OS service that handles packing data into packets, creating packet headers, and handling ACKs.

Typically, each host has a program running inside the OS called "host network stack". In addition to the responsibilities above it also handles getting data from the network and sending it to the right application.

When an app wants to access the network, it opens a *socket* which is associated with a *port*. A socket is an OS mechanism that connects applications to the network stack. A port is a number that specifies a particular socket. The port number is used by the OS for app multiplexing to direct incoming packets to the right applications.

Modularity

The internet is inherently hierarchical. There are big networks at high levels, e.g. AT&T, which connect smaller networks at a lower hierarchy e.g. Purdue. We break the task of sending data into five layers, known as the *Open Systems Interconnect* (OSI) model.

1. Physical: bits sent over a physical connection. Involves signal processing, analog-to-digital conversions, etc.

2. Data Link: "best-effort" data delivery within a local network. Involves naming, addressing, destination discovery, routing, forwarding

3. Network: "best-effort" data delivery between networks. Involves naming, addressing, destination discovery, routing, forwarding

4. Transport: reliable end-to-end data delivery. Involves reliability and application multiplexing.

5. Application: network service to users or applications. Involves read/write data from applications, encrypt/decrypt, etc.

Each layer only talks with the layers directly above and below.
Where should these subtasks be implemented?

Sockets

A *socket* is an OS mechanism for inter-process communication. It allows two processes or applications on the same (loopback) or different machines to communicate with each other. The communication path goes through the network stack of machines running the processes.

The socket interface sits between the application and transport layer. It has send and recieve buffers for each app, which apps access via `send()` and `recieve()`. Each application is attached to a socket that has a unique port number and send-recieve buffers.

Connection

A *connection* is identified with a 5-tuple:

- Local IP address
- Remote IP address
- Local port
- Remote port
- Transport protocol type

These 5 things in combination make a connection. A local and remote socket make a connection.

Any address that begins with `127.` is on the local machine, i.e. on loopback.

Creating a socket is the first step in socket programming. Use the `socket()` system call, which returns a file descriptor. This is interesting, because it abstracts the difficulties of sending data over a network to the same process as writing to a file.

There are three things needed to open a socket, which is done with a call like `int fd = socket(int family, int type, int protocol)`

- `family`: protocol family used for communication. E.g. `AF_INET`.
- `type`: defines the communications semantics used. There are two popular choices, `SOCK_STREAM` which is reliable, and `SOCK_DGRAM` which is unreliable.
- `protocol`: specifies a particular transport protocol. Setting to 0 will choose the default protocol implemented in the OS. E.g. TCP, UDP.

SOCK_DGRAM is connectionless, i.e. no handshake required before sending data. This is a packet or *datagram* abstraction. SOCK_DGRAM offers unreliable communication in the sense that there is no promise that every packet is delivered. For datagram abstraction, any data sent with `send()` just has a UDP header slapped on and that's sent off as a packet. Whatever transport layer on the other side just strips off the UDP header and forwards the payload to the recieving app.

SOCK_STREAM is connection-oriented, i.e. explicit handshake happens before data is sent. This is a byte stream abstraction. SOCK_STREAM offers reliable communication, which means that all the data reaches its destination reliably and in-order. For bytestream abstraction, there is a buffer. Bytes sent are stored in the buffer. The TCP protocol decides how many bytes go into packet.

In either case, excess bytes are dropped if the receive buffer is full. In the datagram case nothing is done. In the bytestream case, the sender is notified and presumably resends the bytes.

In the datagram abstraction, if the sending app makes five send calls, the receiving app should make five receive calls. Nothing similar is promised in the bytestream abstraction.

Note that the actual data going over the network is packets, regardless of if a byte stream abstraction or datagram abstraction is used. The abstraction is just for programming convenience.

A recap of important function calls:

- `socket()`: creates a socket
- `bind()`: bind socket to an address <IP, port>
- `connect()`: initiate connection to server
- `listen()`: listen for and queue incoming client connections
- `accept()`: accept incoming client connections
- `send()`: send data over a connected socket (TCP)
- `recv()`: receive data from a connected socket (TCP)
- `sendto()`: send data to a specific address (UDP)
- `recvfrom()`: receive data and sender's address (UDP)
- `close()`: close the socket and release resources

Network Performance Metrics

When evaluating if a network is good or bad, there are two key metrics: *bandwidth* or *throughput* and *delay* or *latency*. A good network should have high bandwidth/throughput and low delay.

Bandwidth and Throughput

Bandwidth is the max number of bits that can sent through a network per unit time, typically measured in bits per second (bps). Bandwidth is a property of the physical network components. For instance, a copper cable may have a bandwidth of 500 Mbps or a fiber optic cable may have a bandwidth of 100 Gbps.

Throughput measures the bandwidth utilization of a network. It's also measured in bits per second.

$$0 \leq \text{throughput} \leq \text{bandwidth} \tag{1}$$

Throughput is a function of the mechanisms used to communicate over the physical network. Network throughput can be lower than network bandwidth due to inefficiencies of communication mechanism, but a good network will make throughput as high as possible.

Latency

Suppose you are sending a series of bits from point A to B. Delay or latency is the total time between first bit leaving point A and last bit reaching point B. It is measured in units of time, typically seconds or milliseconds. Note that, in a packet-switched network, the entire packet must be received by the router. This "store-and-forward" model introduces delays. An alternative with smaller delay is "cut-through", where the router begins processing as soon as the destination address is received. The tradeoff is that routers can't check for packet corruption, since it necessarily requires the entire packet.

Internet for outer space is currently an area of research. Latency can be minutes, hours, or days in such a case.

We represent delays with a timing diagram, as in Figure 3.

There are different delays when packet routing:

- Transmission Delay: Packet Size / Link Bandwidth
- Propagation Delay: Link Length / Link Propagation Speed
- Queuing Delay: Sum of Transmission Delay for all packets ahead in the queue
- Processing Delay: Depends upon the router hardware processing speed, nature of processing, etc.

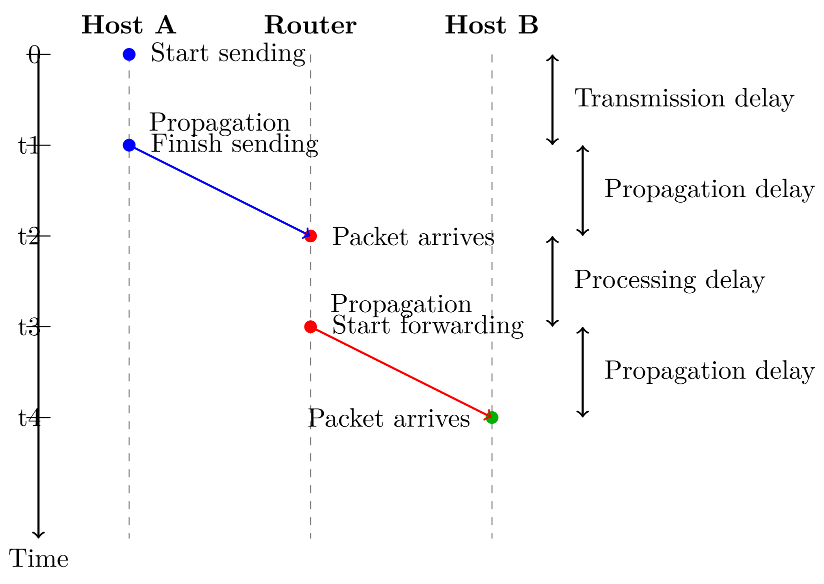


Figure 3: Timing Diagram

The total delay for a single packet going over 2 hops in a store-and-forward packet-switched network is

$$2(TD + PD) + QD + PrD \tag{2}$$

For M hops ($M - 1$ routers) the end-to-end delay is

$$M(TD + PD) + (M - 1)(QD + PrD) \tag{3}$$

For a store-and-forward model with N packets over 1 hop, the time from 1st bit leaving A to last bit of last packet reaching B is

$$N \times TD + PD \tag{4}$$

For a store-and-forward model sending N packets over M hops, the total end-to-end delay is

$$M \times (TD + PD) + (M - 1) \times (QD + PrD) + (N - 1) \times TD \tag{5}$$

Queuing occurs at the router when the packet arrival rate exceeds the packet drain rate.

OSI Model

The Open Systems Interconnection (OSI) model describes communications from the physical implementation of transmitting bits across a transmission medium to the highest-level representation of data of a distributed application. Each layer has well-defined functions and semantics and serves a class of functionality to the layer above it and is served by the layer below it.

Physical

The realm of electrical and computer engineers. Deals with converting between digital and analog signals or electrical and optical signals. Beyond the purview of this course.

Data Link

The data link layer runs on top of the physical layer. It transfers data between nodes on a network segment across the physical layer. Whereas the internet as a whole runs on a global standard (IP) to allow subnetworks to communicate, the data link layer allows autonomy within each local area network (LAN). Each LAN can run its own network protocol for communication within LAN, e.g., Ethernet, Wi-Fi, 5G, CSMA, Sonet, etc. The data link layers handles addressing, destination discovery, forwarding, and routing within a local network. We will study data link layer mechanisms in the context of the most popular data link layer protocol called “Ethernet” Ethernet is an example of a wired data link layer protocol, i.e., nodes are connected using physical cables

Another very popular data link layer protocol is Wi-Fi, which is an example of a wireless data link layer protocol. Wi-Fi is not covered in this class

Network

The network layer runs on top of runs on top of the “best-effort” local area delivery service data link layer. While the data link layer allows information to be relayed within a local network, the network layer connects different local networks. As with all layers, we must solve the problems of addressing, destination discovery, forwarding, and routing. The problems are the same, but the scale is different so the solutions must be different.

Transport

The transport layer provides end-to-end communication services for applications. It ensures complete data transfer, error recovery, and flow control between hosts. This layer segments and reassembles data for efficient transmission and provides reliability with error detection and correction mechanisms. Common implementations are *User Datagram Protocol* (UDP) and *Transmission Control Protocol* (TCP).

Application

MAC Addresses All network devices are connected to the network via a “Network Interface” or “Port”. A network interface can be “physical” (wired or wireless), such as an actual connection on a server in some closet, or it can be “virtual”, i.e., a piece of software emulating a network interface. Each network interface, physical or virtual, has a Media Access Control (MAC) address. MAC addresses are 48 bits or 6 bytes long and typically represented in hexadecimal format, e.g., `ab:00:05:2c:e4:34`. MAC address of each interface within a given network must be unique, but MAC addresses are not necessarily globally unique.

There are three ways to transmit information from a sender to a recipient:

- Unicast: one-to-one transmission
- Multicast: many-to-many transmission
- Broadcast: one-to-all transmission

Naive implementations of broadcast might have the sender send its packet to every of the $N - 1$ hosts in the network, but a more efficient implementation is to send the packet to the router and have it send it out to everyone else. A special destination MAC address of `ff:ff:ff:ff:ff:ff` is used to indicate a broadcast packet.

To see the list of interfaces on your computer, run the following command in the terminal: `ifconfig` (mac/Linux) or `ipconfig` (Windows).

Ethernet

In Ethernet, the Ethernet data (payload) and header is carried in an "Ethernet Frame". The structure of an Ethernet frame is as follows: All Ethernet packets start with a "Preamble" - 7 bytes of alternating 1s and 0s used for clock synchronization between sender and receiver. This is followed by the Start Frame Delimiter (SFD), the one byte `10101011`, then the destination MAX address, 6 bytes, and then the source MAC address, 6 bytes. These are followed by the Ethernet type, 2 bytes, which specifies the protocol carried in the payload of the packet (e.g. IP), and finally the data itself. The data has a minimum size of 46 bytes and a maximum size specified by the Maximum Transmission Unit (MTU), which is configurable. Everything is capped off with a Frame Check Sequence (FCS) of four bytes, used for bit error correction and detection, and an Inter Packet Gap (IPG), which is minimum 12 bytes of all 0s.

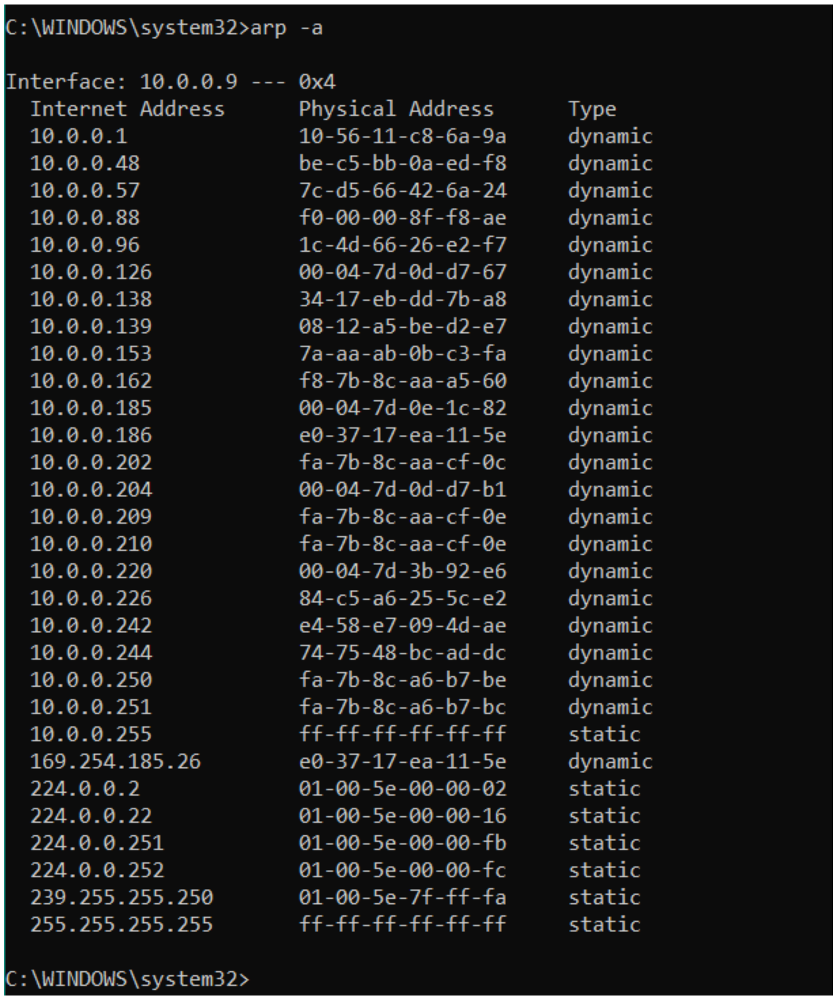
ARP The Address Resolution Protocol (ARP) is used to get the MAC address of a destination host within the same local network as the source host. It assumes you know the IP address of the destination host. Each host maintains a local table called ARP table which stores a mapping between an IP address and MAC address, as in Figure 4. Run `arp -a` on mac/Linux to view the table, If the entry is found in the table, done! Else run ARP to get the MAC address.

The ARP protocol has three stages. Say a host needs the MAC address of some machine that it has the IP address of. The host broadcasts an Ethernet frame with an ARP request. The structure of an ARP request is as follows:

1. Hardware type
2. Protocol type
3. Hardware size
4. Protocol size
5. Opcode
6. Sender MAC
7. Target MAC (all 0s)
8. Target IP

Everyone on a local network gets the request. If the target IP matches the host IP, it sends an ARP reply packet.

The structure of an ARP reply is



```
C:\WINDOWS\system32>arp -a

Interface: 10.0.0.9 --- 0x4
  Internet Address      Physical Address      Type
  10.0.0.1              10-56-11-c8-6a-9a     dynamic
  10.0.0.48             be-c5-bb-0a-ed-f8     dynamic
  10.0.0.57             7c-d5-66-42-6a-24     dynamic
  10.0.0.88             f0-00-00-8f-f8-ae     dynamic
  10.0.0.96             1c-4d-66-26-e2-f7     dynamic
  10.0.0.126            00-04-7d-0d-d7-67     dynamic
  10.0.0.138            34-17-eb-dd-7b-a8     dynamic
  10.0.0.139            08-12-a5-be-d2-e7     dynamic
  10.0.0.153            7a-aa-ab-0b-c3-fa     dynamic
  10.0.0.162            f8-7b-8c-aa-a5-60     dynamic
  10.0.0.185            00-04-7d-0e-1c-82     dynamic
  10.0.0.186            e0-37-17-ea-11-5e     dynamic
  10.0.0.202            fa-7b-8c-aa-cf-0c     dynamic
  10.0.0.204            00-04-7d-0d-d7-b1     dynamic
  10.0.0.209            fa-7b-8c-aa-cf-0e     dynamic
  10.0.0.210            fa-7b-8c-aa-cf-0e     dynamic
  10.0.0.220            00-04-7d-3b-92-e6     dynamic
  10.0.0.226            84-c5-a6-25-5c-e2     dynamic
  10.0.0.242            e4-58-e7-09-4d-ae     dynamic
  10.0.0.244            74-75-48-bc-ad-dc     dynamic
  10.0.0.250            fa-7b-8c-a6-b7-be     dynamic
  10.0.0.251            fa-7b-8c-a6-b7-bc     dynamic
  10.0.0.255            ff-ff-ff-ff-ff-ff     static
  169.254.185.26        e0-37-17-ea-11-5e     dynamic
  224.0.0.2             01-00-5e-00-00-02     static
  224.0.0.22            01-00-5e-00-00-16     static
  224.0.0.251           01-00-5e-00-00-fb     static
  224.0.0.252           01-00-5e-00-00-fc     static
  239.255.255.250       01-00-5e-7f-ff-fa     static
  255.255.255.255       ff-ff-ff-ff-ff-ff     static

C:\WINDOWS\system32>
```

Figure 4: ARP Table

1. Hardware type
2. Protocol type
3. Hardware size
4. Protocol size
5. Opcode
6. Sender MAC
7. Sender IP
8. Target MAC
9. Target IP

On getting the ARP reply packet back, the originating host updates its ARP table with a new mapping from the target IP address to target MAC address.

There are two ways of connecting nodes.

- Shared medium: All nodes connected via single common medium, such as a wire or space itself in the case of wireless transmissions. Each packet sent over the shared medium is received by each host. On receiving a packet, a host checks destination MAC address and discards if it does not match host's MAC address
- Point-to-Point: Dedicated pairwise connections.

An issue with shared medium is that there can be collisions. The solutions are somewhat technology-dependent, so we will discuss the solution in the context of Broadcast Ethernet where the shared medium is a wire.

There are two classes of techniques:

- Reservation, including frequency division multiple access (FDMA) and time division multiple access (TDMA)
- On-demand, including random access

In FDMA, we divide the medium into frequencies. Each source is assigned a subset of frequencies and sends on its assigned frequency. With TDMA, divide time into time slots. Each source is assigned a subset of time slots and sends on its assigned time slot. The benefit of these strategies is that we avoid collision. However, if source is idle, then its frequency/time slot is wasted. In FDMA, noise interference may cause disruption In TDMA, if source has data to send, can't send immediately, wait for its slot. TDMA also requires clocks of all hosts to be synchronized.

With random access, when a source has a packet to send, it sends it out. Unfortunately this leads to corruption when two packets collide.

There are methods to detect mitigate corruption. One is to have the sender keep listening to the medium while transmitting. If sender senses collision (e.g., excess current on the wire), it aborts transmission and broadcasts a "Jam" signal. A Jam signal is a random 32-bit signal that ensures that all receiving nodes fail CRC checksum and discard the frame. The sender then waits for a random time and resends to avoid instantly colliding again.

Another method to mitigate corruption is Carrier Sense Multiple Access (CSMA). In CSMA, the sender listens to the shared medium before transmitting. If idle, it starts transmitting. If busy, it waits. This does not eliminate collision because of non-zero signal propagation delay.

Together this collision detection and CSMA are called CSMA/CD. CSMA listens to the medium and waits for it to be idle before transmitting. CD sends a Jam signal if a collision is detected. For re-transmission, most implementations use random exponential backoff. After a packet collision, sender tries to re-transmit packet after a wait time. After k collisions for a packet, choose wait time randomly from $\{0, ..., 2^k - 1\}$ time slots. k resets to 0 after a packet transmission succeeds. This gives exponentially increasing wait time with each attempt, but also exponentially larger success probability with each attempt.

CSMA/CD does not scale to large number of hosts. It gets a higher collision rate, wastes more bandwidth re-transmitting the same packets, and has high and unpredictable delay due to variable back-off times. In practice, shared LANs don't have more than 1000 hosts Another issue is that CSMA/CD assumes hosts send packets intermittently. If everyone is sending steadily at all times there will be more collisions. In addition, for CD to work, the sender must be able to detect collision (if it happens) before it is finished transmitting the entire packet. If that's not true then the sender might have sent out multiple packets before receiving the Jam signal. On re-transmit, some receivers might receive duplicate packets. This imposes a constraint on the minimum packet size or maximum network length. At high bandwidth, CSMA/CD requires either large min packet size (wasted bandwidth when less data to send!) or small network length.

So how do we overcome the scalability issue? With a point-to-point forwarding device, as in Figure 5. A point-to-point forwarding device typically comprises multiple ports (or network interfaces). Each port connects to a single host or multiple hosts sharing a medium or some other forwarding device, using point-to-point links. It forwards packets received on one port out on some other port.

There are three layers for forwarding devices, although a given

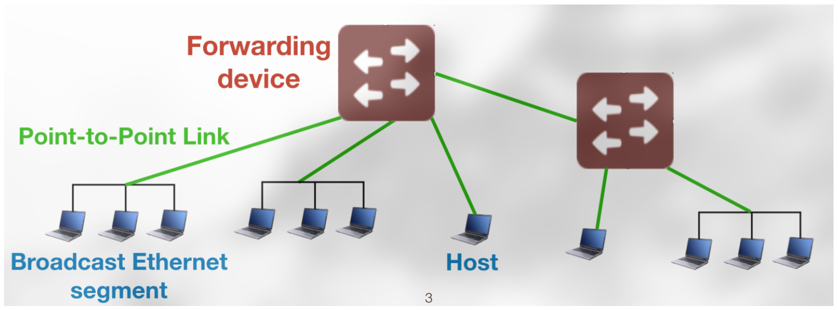


Figure 5: Forwarding Device

device can function at multiple layers:

- Layer 1: operates at Physical layer, i.e., forwards using physical layer packet header fields. Example: a Hub
- Layer 2: operates at Data Link layer, i.e., forwards using Data Link layer packet header fields (e.g., using destination MAC address). Example: a bridge or switch.
- operates at Network layer, i.e., forwards using Network layer packet header fields (e.g., using destination IP address). Example: a router.

Layer 1 A layer 1 device, the hub, is the simplest possible forwarding device. It broadcasts frame received on a given port to all other ports except the port the frame was received on. Physical layer headers contain no address information, so broadcast is the only option. Typically a hub connects multiple Broadcast Ethernet segments. and helps extend Broadcast Ethernet to larger distance by providing signal amplification and signal re-generation. Nobody really uses hubs these days because they just create bigger Ethernet segments, which still have collisions.

Layer 2 The simplest layer 2 device is a bridge. It understands MAC addresses and creates two half-duplex point-to-point connections between its two ports. A generalized version of the bridge, the switch, is the most commonly used device. A switch is a multi-port bridge, i.e., has $N > 2$ ports. It creates N half-duplex point-to-point connections (a *matching*) between input and output ports using a crossbar, which is just a bunch of wires going between input and output ports. A matching is a bipartite graph where every edge has a unique endpoint.

An algorithm called iSLIP decides the matching configuration. iSLIP looks at the current *demand*, i.e., for each input packet what is the output port. iSLIP then configures the crossbar to create the matching that satisfies the most demands. It repeats the above two steps iteratively till all demands are satisfied.

Ethernet running inside a switch is called "Switched Ethernet". Modern Ethernet LANs run Switched Ethernet instead of Broadcast Ethernet.

In switched ethernet, each switch maintains a "Forwarding Table" Which keeps track of which hosts are reachable via each output port. For the network in Figure 6, the forwarding table is given by the table below.

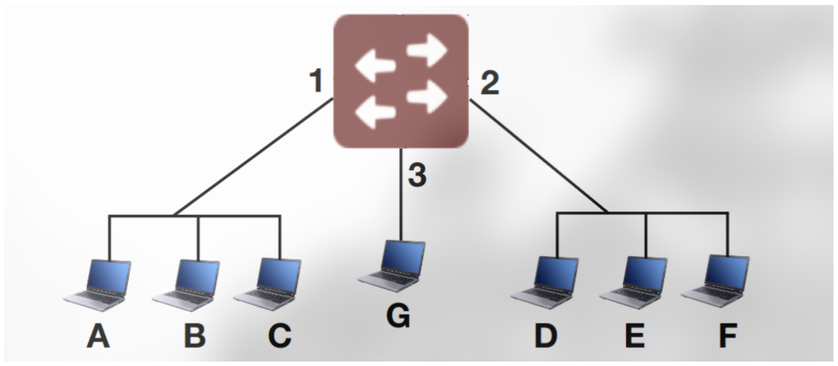


Figure 6: Switched Ethernet Network

Destination MAC Address	**Output Port**
A.mac	1
B.mac	1
C.mac	1
D.mac	2
E.mac	2
F.mac	2
G.mac	3

To populate the forwarding table, use the following algorithm. When a switch receives a frame on port p, it checks the source MAC address in the frame. Let it be s. The switch learns that host with MAC address s is reachable via port p, so it adds the entry $< s, p >$ to its forwarding table. If a switch never receives a frame sourced from a host, it will never learn which port the host is reachable on.

The way MAC forwarding works is that, upon receiving a frame with destination MAC d and if `d = ff:ff:ff:ff:ff:ff`, copy and send frame on every port except the port on which the frame was received on. This is a broadcast. If d is not `ff:ff:ff:ff:ff:ff`, then check the forwarding table for an entry $< d, p >$. If entry $< d, p >$ exists, then if p is same as the port on which frame was received, then drop the frame. Otherwise send frame out on port p. Else if no entry corresponding to d exists, the switch copies and sends frame on every port except the port on which the frame was received on, again a broadcast. If there is a loop in the network graph, then a broadcast packet can cause infinite loops and consume all resources in a broadcast storm. The way to avoid this is to structure networks as spanning trees with no loops. Doing this in a distributed way is extremely complex, and MAC learning by itself can't handle finding a route with no loops.

Spanning Tree Protocol

Luckily, the *Spanning Tree Protocol* (STP) extends MAC learning to detect and remove loops from the graph. The task of STP is, given a network of switches and end hosts, to create a graph where vertices represent switches/hosts and edges represent links, then to remove all loops from the graph. The way STP accomplishes this is by, unsurprisingly, building a spanning tree, a subgraph that includes all vertices but no cycles. In a spanning tree there is exactly one path between vertex pairs, which means there will be no loops.

Unfortunately, no single switch has full view of the graph, so we need to build the spanning tree in a distributed manner. In STP, switches exchange messages to build the spanning tree. These messages are carried in special packets called *control packets*. These packets are distinct from normal Ethernet packets and have STP-specific info. Control packets in STP are only exchanged between directly connected neighbor switches and are never forwarded beyond that. Control packets are only used for STP and are not used for MAC learning and forwarding.

STP has three steps:

1. Pick a root: pick the switch with the smallest MAC address.
2. Compute the smallest cost path to the root. For each switch, calculate the smallest cost path to the root. Where there are multiple smallest cost paths to the root, choose the path via neighbor switch with the smaller MAC address.
3. Make links not on any smallest cost path "inactive". For link A-B, if neither A nor B uses it on its smallest cost path, then it's "inactive". Switches do not forward any data packets on that link and ignore any received data packets on that link. However, continue to send and receive control packets on "inactive" links to handle failures where they may need to be reactivated.

Each switch maintains a view $(R, \text{cost}(X, R), X, H)$. R is the current root according to X. $\text{cost}(X, R)$ is the cost from X to R. H is the next hop neighbor via which X reaches R.

A control packet from neighbor X to Y carries X's current view, $(R, \text{cost}(X, R), X, H)$ where X is proposing R as root and advertising a cost of $\text{cost}(X, R)$ from X to R via neighbor switch H (next hop to root from X). The algorithm converges when no switch view changes on receipt of a control packet.

```
// propose this switch as root
send (X, 0, X, X) to all neighbors

// on recieve (R, cost(Y, R), Y, H)
// from neighbor Y
```

```
when control packet recieved
    if advertisement from current next hop to root
        if R < X
            view = (R, cost(X, Y) + cost(Y, R), X, Y)
    else
        // also update if same root, same cost, smaller MAC
        if smaller root or cost path advertised
            update
```

The protocol must react to failures and link cost changes in the network. Switches, including the root, can fail. Links can fail or their cost can change. Each switch X tracks the status of its current next hop H and link $X - H$. On detecting failure of X or $X - H$, X updates its view to $(X, 0, X, X)$. Each switch X periodically sends its view $R, \text{cost}(X, R), X, H)$ to all its neighbors. This is needed for convergence under certain failure scenarios and makes STP robust to control packet corruption and loss.

The final Spanning Tree Protocol operates as follows: Initially, each switch proposes itself as the root, setting its view to $(X, 0, X, X)$ and sending a control packet $(X, 0, X, X)$ to all neighbors. Upon detecting a failure of its next hop (H) or the link $X - H$, switch X updates its view to $(X, 0, X, X)$ and advertises this new view to all neighbors. Each switch periodically sends its current view $(R, \text{cost}(X, R), X, H)$ to all neighbors. When receiving a control packet $(R, \text{cost}(Y, R), Y, H)$ from neighbor Y, switch X updates its view according to predefined cases (Case 0, 1, and 2). If X's view changes, it sends its new view in a control packet to all neighbors. Additionally, X checks the next hop of neighbor Y each time a control packet is received from Y. A link $X - Y$ is made "inactive" if and only if the next hop of X is not Y and the next hop of Y (H) is not X, at which point X stops forwarding data packets on that link, removes related entries from the forwarding table, and ignores data packets received on $X - Y$ for MAC learning and forwarding. The algorithm converges when no switch updates its view upon receiving a control packet.

Internet Protocol

The *Internet Protocol* (IP) address replaces the MAC address for the network layer, IP forwarding replaces MAC forwarding, network routing protocols (DV, LS, BGP) replace MAC learning with STP, and destination discovery with DNS replaces ARP.

As the most popular protocol with the data link layer is Ethernet, the most popular protocol with the network layer is *internet protocol* (IP).

While the MAC tells the identity of the host, the IP address tells the location of the host. The MAC address of a host does not change (in most cases), and so can provide host-based services such as allowing access to a network service to an authorized computer, regardless of the location of a computer. IP addresses allow reaching distant devices. IP address do this by constructing a hierarchy so that each router only needs to store the IP address of the networks immediately above and below it in the network hierarchy.

STP doesn't work as a routing protocol at the network since it finds the shortest path to the root and not the destinations. STP also has higher latency and wasted bandwidth. The pros of simplicity and quick convergence outweigh the cons of higher latency and wasted bandwidth for small networks, but not for large. With network routing protocols, each router finds the shortest path to each destination, and there is no root. The pros are better latency and better bandwidth utilization, but at the cost of higher complexity and taking longer to converge. The pros outweigh the cons for large networks.

ARP doesn't work for large networks because it requires broadcast requests to every host on the network. DNS requires dedicated infrastructure and extra mechanisms for fault tolerance, but doesn't require broadcasts, which is good for large networks.

IP is really the only protocol for the network layer. This is in stark contrast to every other layer, which have a proliferation. In this class we'll focus on IPv4 instead of the newer IPv6. In IPv4, IP addresses have 32 bits and are represented as `X.X.X.X` where `X` is an 8-bit decimal, e.g. `192.168.3.29`.

Hosts within a subnet share the same subnet address prefix. So perhaps devices in the same subnet have IP addresses `X.X.X.92`, `X.X.X.23`, and `X.X.X.01`. Any hierarchy can be encoded in the IP address; perhaps the first n_0 bits correspond to a given country, the next n_1 to an internet provider in that country, the next n_2 to an organization, the next n_3 to a location, etc.

We use a subnet mask to extract the subnet address from the IP address. The subnet mask a 32 bit long series of 1s followed by a series of 0s. For example, `255.255.240.0` is 20 1s followed by 12 0s.

The subnet address is the bitwise AND of the IP address and subnet mask. For the previous example, with an IP address of `192.168.3.29` the subnet address would be `192.168.3.29 & 255.255.240.0 = 192.168.0.0`.

Under classful addressing, IP addresses are divided into a set of classes. Each class has n bits statically allocated for a subnet address and the remaining $32 - n$ bits are for the host identifier. Depending on the value of n, there are three popular classes.

- A: $n = 8$, MSB `0`, 8 subnet bits, 24 host bits, 128 subnets, $2^2$4 hosts.
- B: $n = 16$, MSB `10`, 16 subnet bits, 16 host bits, 16K subnets, $2^1$6 hosts.
- C: $n = 24$, MSB `110`, 24 subnet bits, 8 host bits, 2M subnets, 2^8 hosts.

The issue with classful IP addressing is that the division of bits between subnet and host addresses are not flexible. So to support 129 subnets one should need only 8 bits for subnet address, but with classful IP addressing, one will need a B class address, which uses 16 bits for the subnet address, wasting 8 bits.

The more widely implemented method is called *Classless Inter-Domain Routing* (CIDR). CIDR allows a flexible number of bits to be allocated for the subnet address. In CIDR notation, a subnet address is represented as `X.X.X.X/n`. The first `n` bits are allocated for the subnet address, meaning the subnet can support 2^{32-n} IP addresses. For example, the subnet address `128.32.0.0/11` can support $2^2$1 IP addresses in the range `128.32.0.0` to `128.63.255.255`.

Hosts have two options for configuring their IP addresses. They can either do it manually and pick whatever IP address they want, or they can implement the *Dynamic Host Configuration Protocol* (DHCP) and have an IP automatically assigned to them. The way this works is that DHCP typically runs on the router and maintains a pool of allowed IP addresses for a network, and when a host connects, the router assigns it an IP address.

There are public IP address, used for routing over the internet, and private IP addresses, for communication within a private network. Public IP addresses are assigned by the Internet Corporation for Assigned Names and Numbers (ICANN). ICANN allocates large IP blocks to regional internet registries, e.g. the middle East, Europe, central Asia. Each internet registry allocates address blocks to large *internet service providers* (ISPs) within that region. The ISP allocates addresses to individuals and smaller institutions. For instance, ICANN allocates some `X.0.0.0/8` addresses to ARIN, which allocates one /8 address to AT&T, which allocates one /16 address to Purdue, which allocates one /24 address to ECE, which gives your computer an IP.

Hosts over the internet need a unique public IP address for correct routing, but there are only $2^{32} \approx 4000000000$ unique IPv4 addresses. The number of modern devices would exhaust this very quickly if not for private IP addressing.

Private IP addresses can be used for communication only within a private network, and packets with private IP addresses as destinations are dropped by public internet routers. Hosts in different private networks can have the same private IP address, which helps with the problem of IPv4 address exhaustion. The reserved private IP address ranges are

- `10.0.0.0/8` (`10.0.0.0` to `10.255.255.255`)
- `172.16.0.0/12` (`172.16.0.0` to `172.31.255.255`)
- `192.168.0.0/16` (`192.168.0.0` to `192.168.255.255`)

Network Address Translation (NAT) enables hosts on private networks to communicate with hosts on the Internet. A NAT device sits at the boundary of a private network and the public Internet (typically implemented inside a gateway router) and manages a pool of public IP addresses allocated to the private network. When a host from the private network wants to send an IP packet to a host in public Internet, NAT picks a public IP from the pool and re-writes the (private) source IP in the packet with public IP. It also stores the private IP to public IP mapping in a table to re- translate the (public) destination IP of an incoming reply packet from the Internet with the corresponding private IP. On its own, this doesn't help with the problem of IP address exhaustion. Ideally, NAT should share a small number of public IPs between a large number of private hosts. This is done with IP masquerading. With IP masquerading, a single public IP is mapped to multiple hosts in a private network. Hosts mapped to the same public IP are assigned different port numbers to distinguish them from one another. A port number is 16 bits, meaning one can support 2^16 hosts using a single public IP address.

IP masquerading is also called Network Address and Port Translation (NAPT) or Port Address Translation (PAT).

This comes with a cost. NAT destroys universal end-to-end reachability of hosts on the Internet. A host on public Internet cannot initiate direct communication to a host with a private IP address. Applications that carry IP address in the payload of the application data (e.g., HTTP) generally do not work across a private-public network boundary.

Some NAT devices inspect the payload of widely used application layer protocols, and if an IP address is detected, they do the translation.

An IP packet, also called an IP datagram, is the fundamental unit of data exchanged at the network layer. Each packet consists of a header and a payload. The header contains all necessary information for routing and delivery, while the payload carries the data from the transport layer (e.g., TCP or UDP segment).

The structure of an IPv4 packet is as follows:

- **Version (4 bits)**: Specifies the IP version. For IPv4, this value is `4`.
- **Header Length (4 bits)**: Specifies the length of the header in 32-bit words. The minimum value is `5`, corresponding to 20 bytes.
- **Type of Service (8 bits)**: Indicates the priority and quality of service desired for the packet, such as low delay or high reliability.
- **Total Length (16 bits)**: The total size of the IP packet in bytes, including both header and payload. The maximum value is `65535`.
- **Identification (16 bits)**: A unique identifier assigned to each packet so fragments of the same packet can be reassembled.
- **Flags (3 bits)**: Controls or identifies fragments. The most important flag is the "More Fragments" bit, which is set if more fragments follow.
- **Fragment Offset (13 bits)**: Indicates the position of the fragment relative to the start of the original packet.
- **Time to Live (TTL) (8 bits)**: The maximum number of hops (routers) the packet can traverse before being discarded. Each router decrements the TTL by one; if it reaches zero, the packet is dropped.
- **Protocol (8 bits)**: Specifies the protocol carried in the payload, such as `6` for TCP or `17` for UDP.
- **Header Checksum (16 bits)**: Used for error detection on the header only. Each router verifies and recomputes it.
- **Source IP Address (32 bits)**: The IP address of the originating host.
- **Destination IP Address (32 bits)**: The IP address of the intended recipient.
- **Options (variable length)**: Optional field used for features such as record route, timestamp, or security.
- **Padding (variable length)**: Added to ensure the header length is a multiple of 32 bits.
- **Data (variable length)**: The payload, usually a transport-layer segment such as TCP or UDP data.

A typical IPv4 header without options is 20 bytes long. The payload size depends on the Maximum Transmission Unit (MTU) of the underlying data link layer. If an IP packet is larger than the MTU, it is divided into smaller fragments. Each fragment is then reassembled by the destination host using the Identification, Flags, and Fragment Offset fields.

IP provides a best-effort delivery service, meaning it does not guarantee reliability, ordering, or data integrity beyond basic error detection on the header. These functions are handled by higher-layer protocols such as TCP in the transport layer.

Network routing protocols are used to populate the routing tables. They calculate the best path from each router to all other routers/hosts, unlike STP which calculates the best path from each switch to a root switch.

Networking routing protocols come in two flavors: *distributed path* computation and *distributed topology* computation.

Distributed path computation is similar to STP in that each router computes paths using a distributed algorithm, oblivious of network topology. The distance vector (DV) and border gateway (BG) algorithms fall into this category.

The distance vector algorithm is a distributed version of the Bellman-Ford algorithm that calculates the least cost path from each router to all the n routers and hosts. Each router maintains a vector of views of its least cost path to all other routers and hosts. It shares its vector with all its neighbor routers and updates the vector and receiving vectors from its neighbors. The most popular distance vector implementation is the *routing information protocol* (RIP).

DV does not prevent loops under all scenarios. We mitigate this with split horizon, which dictates that if a router gets a vector that states the next hop to a destination is itself, to drop the vector. This prevents loops involving two routers, but cannot prevent loops of three or more. To detect and remove all loops, set the *max infinity counter* to some value larger than the largest path cost in the network. When the cost to a destination reaches or exceeds infinity, the router detects count-to-infinity. On detection, the router removes the destination entry from its routing table or sets the next hop for the entry to NULL. This is used alongside split horizon, which tries to prevent loops.

Distributed topology computer is when routers use a distributed algorithm to learn the entire network graph. Each router runs a local path computation algorithm on the learned graph. This category includes link state (LS).

In link state, each router runs a distributed algorithm to learn the global network graph. It then runs a local computation to find the shortest paths and populate its routing table. The most popular link state implementation is Open Shortest Path First (OSPF).

Each router X maintains a view of the global network graph G. Initially, G is empty. For each neighbor Y, X adds link $X - Y$ and link cost $\text{cost}(X, Y)$ to G. X creates a control packet called a link state announcement (LSA) and sends it to all neighbors. The LSA contains the list of all neighbors of X and the cost to neighbors, i.e. $(Y, \text{cost}(X, y))$. On receiving node Y's LSA $(Z, \text{cost}(Y, Z))$ X updates its graph G by adding or updating link $Y - Z$ and the corresponding link cost. X then broadcasts the received LSA on all ports except the port it arrived on. Every time G changes, X reruns the route computation algorithm locally. It computes the least cost path from itself to all other nodes by using, for example, Dijkstra's algorithm.

LSAs are sent when a router learns of a change to a neighbor, like a neighbor or link to neighbor failed, a neighbor was added, or a link cost to neighbor changed. LSAs are also sent periodically to correct possible corruption of previous LSA data or previous LSA packet drops.

LSA comes with a big problem. Suppose X sent an LSA at time t with neighbors $\{W, Z\}$. At time $t + 2$, a new link $X - V$ is added and X sends a new LSA $\{W, Z, V\}$. At router Y, LSA $\{W, Z, V\}$ arrives before $\{W, Z\}$ because the new link allowed the LSA to be propagated faster. After $\{W, Z\}$ arrives, Y will conclude that X only has 2 neighbors. This wasn't an issue with STP and DV, because control messages were only exchanged between directly connected neighbors.

One way to mitigate this could be to include a timestamp with each LSA sent, and for Y to ignore LSAs from X whose "sent" timestamp is older than any LSA that Y has already received. This is essentially what routers do. Each router X maintains a local sequence number S_x and while generating an LSA, the router includes the current value of S_x in the LSA before broadcasting. With each LSA broadcast X increments S_x by 1. Each router also maintains a list of the highest sequence number that it has received from each of the other routers. This prevents loops in all scenarios, although there can be transient loops. While LS has not converged, loops can still be formed. To handle these transient loops each IP packet contains an 8-bit *time-to-live* (TTL) field in the header. The sending host initializes the TTL with some value. TTL is decremented by one by the router at each hop. When TTL reaches 0, the packet is dropped by the router. TTL is not a substitute for routing protocols because it does not prevent packets from getting in a loop, only reduces the amount of time a packet spends in a loop.

The key difference between DV and LS is that DV advertises your paths to everyone to your neighbors, while LS advertises your paths to neighbors to everyone.

Border Gateway Protocol

Recall that the internet is a hierarchical network of networks. An autonomous system (AS) or domain is a network under a single administrative domain. Each AS is assigned a unique identifier.

Intra-domain routing refers to routing within an AS. Examples: RIP (distance vector), OSPF (link state).

Inter-domain routing refers to routing between ASes. Example: border gateway protocol (BGP).

We can't simply use DV or LS for inter-domain routing because ASes want freedom to pick routes based on custom policy. This can't be expressed as a least cost path like in DV or LS, but rather depends on the business relationship between ASes.

There are three basic kinds of relationships between ASes.

- Customer, e.g. Purdue is AT&T's customer. The customer pays the provider.
- Provider, e.g. AT&T is Purdue's provider.
- Peer, e.g. AT&T and Verizon are peers. Peers don't pay each other. Peers exchange roughly equal levels of traffic.

A customer is multi-homing if it has multiple providers. Provider ASes connect customer ASes to the rest of the Internet, and customer ASes pay the provider ASes for this service. Peering is needed to connect ASes at the top, which have no providers. These ASes are known as Tier 1 ASes and include ISPs like AT&T, Sprint, Verizon. Peering can also be done at lower levels for various reasons, such as two top secret institutions not trusting a provider and exchanging information directly.

To set up inter-domain routing, destinations are subnets. Nodes are ASes, and the internals of each AS are hidden. Physical links between ASes are tagged with the corresponding business relationship. Border gateway protocol (BGP) is user for routing.

In BGP, each AS advertises to neighbor ASes its best paths to one or more subnets. Each AS selects the "best" path(s) from the set of advertised paths to a subnet destination.

BGP is inspired by distance vector. It has per-distance route advertisements to its neighbors. There is no global learning of network topology. It's iterative and uses distributed convergence to final paths. But, there are three key differences.

- While distance vector shares the distance and next hop to a destination, BGP shares the entire path to a destination to enable ASes to apply more complex policies. This makes loop detection trivial.

- BGP does not always select the shortest path.
- For policy reasons, an AS may choose not to advertise a route. Thus reliability is not guaranteed even if the graph is connected.

BGP policy is imposed in how route are selected and advertised. Selection is which path to use to send data. It controls whether/how traffic leaves the network. Advertisement is which path to advertise to other ASes. It controls whether/how traffic enter the network. Traffic flows in the reverse direction of the advertisement.

Typical selection policies are, in decreasing order of priority,

1. Make or save money by preferring a path through the customer, then a path through a peer, then a path through a provider
2. Maximize performance by selecting the shortest path
3. Minimize use of network bandwidth aka "hot potato" routing

The typical advertisement policy is Gao-Rexford advertisement policies, whose goal is to avoid being transit when there is no monetary gain. The customer advertises to providers, peers, and other customers. The peer advertises to customers. The provider advertises to costumers. The AS provides transit service to all its customer traffic, because that's what customers pay for. It does not provide transit service between two providers or peers, since these entities are not paying the AS.

eBGP refers to BGP sessions between gateway (border) routers in different ASes, used to learn routes to external destinations. iBGP refers to BGP sessions between gateway routers and other routers (both gateway and interior) within the same AS, used to distribute externally learned routes internally.

IGP, Interior Gateway Protocol, is an intra-domain routing protocol that provides internal connectivity.

Basic messages in BGP are

- Open message: establishes BGP session, relayed on a reliable transport layer like TCP.
- Update message: Advertises new routes or route changes to neighbors and updates neighbors of any old routes that have become inactive.
- Keep-alive message: informs neighbor that BGP session is still active.

Update messages have the format {Destination IP prefix: Update type | Route attributes}. There are two update types: announcement, which heralds new route or changes to existing routes, and withdrawal, which announces the removal of routes which no longer exist. Route attributes are used in route selection and advertisement choices. They include `ASPATH`, a vector of ASes a BGP message has traversed, and `LOCAL PREF`, preference value for `ASPATH`, among others.

In BGP, it's very easy to manipulate route advertisements. Reachability is not guaranteed even if a graph is connected. More seriously, there are attacks which an AS can perform by manipulating advertisements. An AS can advertise an IP prefix that they actually don't have a route to, or advertise a more specific IP prefix which, due to longest prefix match, all traffic to the sub-prefixes will be redirected. Thus ASes can act as black holes causing all traffic to the hijacked prefix to be discarded, they can snoop by inspecting traffic to the hijacked prefix, and redirect traffic to a hijacked prefix to bogus destinations. BGP path advertisements can also be manipulated to show shorter paths or more connected ASes. These attacks happen relatively infrequently since one needs access to BGP routers to launch most attacks and it's easy to detect the culprit. Most BGP mishaps are the result of unintentional misconfiguration.

Domain Name System

Domain Name System (DNS) provides the mapping from the human-readable name to IP address. A user asks DNS something like "what is the IP address of www.google.com?" and DNS responds "8.8.8.8".

The DNS infrastructure consists of local DNS servers subordinate to centralized DNS servers managed by ICANN.

DNS has a hierarchical infrastructure. For example, `ece.purdue.edu` (the ECE's domain) is a subdomain of `purdue.edu` (Purdue's domain), which is itself a subdomain of `edu` (the top level domain for American educational institutions). Thus, DNS infrastructure reflects this hierarchy. `ece.purdue.edu` has a DNS server, `purdue.edu` has a DNS server, and `edu` belongs to the set of generic domain servers. The general hierarchy is:

1. Root server: address hardwired into other DNS servers. Managed by ICANN/IANA
2. Top Level Domain (TLD) servers: `.com`, `.uk` etc. Managed by ICANN/IANA
3. Local DNS servers: `.purdue.edu`, `.google.com`. Managed by local service providers

Every DNS server knows the address of the root server. Every DNS server knows the address of its immediate children. A DNS server only stores the name-to-address mappings of the domain it has authority for. The DNS hierarchy is designed this way so that each server only needs to store a subset of the DNS database, ensuring scalability, and so that any server can discover servers for any portion of the hierarchy.

Each DNS server is typically replicated for availability and scalability. The DNS service is available if there is at least one replica up. Queries can be load balanced among replicas using *anycast*.

A comparison of anycast and other destination IP addressing methods follows:

- unicast: one destination host
- broadcast: desintation is all hosts in the network
- multicast: subset of hosts in the network
- anycast: a group of receivers are associated with the same destination IP. A routing algorithm sends to a single receiver from the group based on some metric (e.g. least loaded receiver, geographically closest). Typically used to balance the load across multiple servers each running the same service

Say you, an entrepreneur, wants to start a university named "The People's Northeastern University of Singapore". You want to register the domain "pnus.com". The first step is to get a block of IP addresses from your ISP, such as `112.110.117.128/25`. Next, you register "pnus.com" with a registrar like GoDaddy, which you provide with the name and IP of your local DNS server(s). The registrar will insert what you provide into the `.com` TLD server. For instance, you might provide your registrar with `dns1.pnus.com`, `112.110.117.115`. You then store mappings for your domain in your local DNS server `dns1.pnus.com`, such as

- `www.pnus.com`: `112.110.117.110`
- `mail.pnus.com`: `112.110.117.100`
- `math.pnus.com`: `112.110.117.0`

Hosts on a network have a DNS client which triggers the resolver code via, for instance, the command `gethostbyname()`, which then sends the DNS query to the local DNS server. If the DNS doesn't have the associated IP, it may contact other DNS servers to resolve the query before sending back the response to the client.

In the DNS protocol there are *query* and *reply* messages, which are carried over unreliable transport such as UDP port 53. The DNS specification supports reliable transport (TCP) too, but this is not always implemented. Reliability is typically implemented via repeating requests on timeout.

There are two ways to resolve a DNS query, as illustrated by Figures 7 and 8.

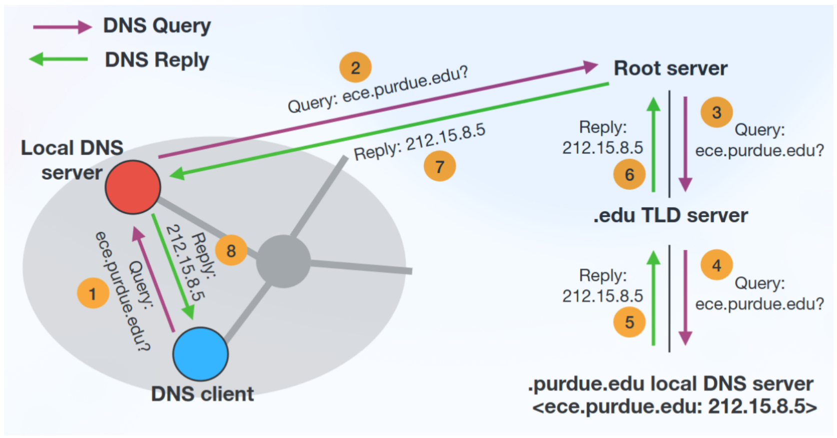


Figure 7: Recursive DNS query resolution

All DNS servers at every level of the hierarchy reply to queries. DNS replies include a *time to live* (TTL) field. The DNS server delete

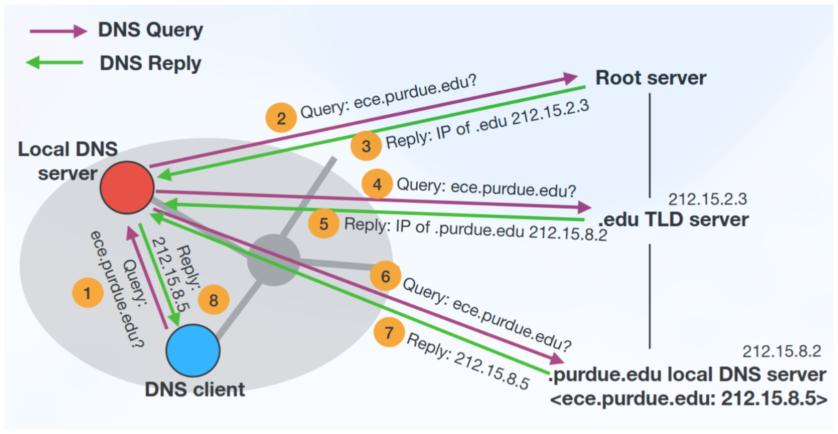


Figure 8: Recursive DNS query resolution

cached entries after the TTL expires. Most popular queries hit the cache in the local DNS server, which enables a faster reply to clients. DNS caching can be done at the host for even faster reply. Windows implements this by default, and applications such as browsers can also implement their own DNS caching.

Transport Layer Reliability

For the transport layer to be *reliable*, it must be correct and performant in a specific sense.

A transport mechanism is "reliable" if and only if it resends all dropped or corrupted packets and it attempts to make progress. Making progress in this context means if there is data to send, the transport layer eventually attempts to send that data. Performant means that whatever implementation utilizes the network and host bandwidth efficiently.

Reliability is a difficult problem. Consider a naive protocol that sends each packet as often and fast as possible till the packet is received at the receiver. The sender will not make progress because there is no way for the sender to know whether the packet was dropped or received, so it will continue sending the same packet at all times. We need feedback from the receiver to indicate whether the packet was received.

The naive protocol is updated to send ACK for each received packet. The sender keeps resending the same packet until ACK is received. This is suboptimal because the source is unnecessarily sending the same packet multiple times.

Finally, the protocol is updated so the receiver sends an ACK for each received packet. The sender sends a packet and waits for the ACK. If an ACK is received back, the sender sends the next packet. If no ACK is received in some timeframe the same packet is resent because the ACK or sent packet may have been lost. This is still suboptimal because the sender has to wait between ACKs, so why not keep sending packets while waiting to receive the ACK?

A sender can have at most w unacknowledged packets in flight, where w is known as the window. The window size it set such that the sender can maximize use of the available bandwidth. The sender sets a timer for each sent packet and waits for the ACKs. It sends the next packet in line after each received ACK and if it doesn't receive an ACK and the timer has expired, it resends the packet. To match packets and ACKs and ensure data doesn't get out of order, each packet is tagged with a sequence number.

Sequence numbers must be a monotonically increasing value carried in each sent packet. It tells the receiver the order of packets within the sending packet stream. It can be used by the receiver to figure out which packets were received and which were lost. It also helps with in-order packet delivery.

The acknowledgement number carries the sequence number of packets in the received stream. It tells the sender which packets have been received. It can be used by the sender to detect and retransmit lost packets.

For individual ACK there are two retransmission strategies, *go-back-n* which retransmits all packets starting from the smallest sequence number for which an ACK was not received, and *selective retransmit* which retransmits only those packets for which an ACK was not received. The former is simpler, while the latter is more efficient.

In selective ACK, each ACK contains the information about all received packets thus far, the highest contiguous number plus any additional sequence numbers. This is to help with lost ACKs. The loss of an ACK does not necessarily require a retransmission, because each ACK carries info for all the previous ACKs.

Cumulative ACK contains just the highest contiguous received sequence number. This avoids the overhead of selective ACK but keeps most of its benefits. The condition to retransmit here is if a timer expires or the sender receives duplicate ACKs. Note, however, that network delays could result in duplicate ACKs, so receiving a duplicate ACK only indicates that packet loss might have occurred. If the sender receives multiple duplicate ACKs in a row (typically 3), then retransmit.

Key design consideration for reliability are

- What sequence number should the receivers send as ACK to the sender?
- Under what conditions should a sender retransmit a packet?
- What packets should a sender retransmit?
- What is the right window size for a sender at any given time?

Recall that a *window* is the maximum number of unacknowledged data the sender can have in flight, which directly affects the rate at which the sender can send data. The goal of sizing a window is to maximize available network and host bandwidth. Senders should send as fast as they can without overloading the network or receiver, i.e. such that no queuing occurs either in the network or at the receiver.

When sender sends a data packet, assume it takes t_1 time to get to destination and another t_2 time for the ACK to arrive at the sender. The *round trip time* (RTT) is $t_1 + t_2$. Let B be the available bandwidth on the path from source to destination. Then the window w should be $B \times RTT$ bits, also known a the *bandwidth delay product* (BDP). To maximize bandwidth use the sender should always send BDP window of bits.

In practice it is difficult to estimate the right values for B and RTT because the available bandwidth depends on how much bandwidth other hosts are using. Additionally, The RTT is the sum of transmission delays, propagation delays, processing delays, and most importantly queuing delays, which are non-deterministic. So in theory, the best value of a window is $B \times RTT$, but in practice requires careful consideration of congestion and flow control.

User Datagram Protocol

One of the most popular transport layer implementations is *User Datagram Protocol*, UDP. UDP is unreliable, connectionless, datagram-abstracted transport. Unreliable in the sense of lacking an ACK, re-transmission mechanism, and window sizing. Datagram-abstracted in the sense of using packets. Connectionless in the sense of sending data without any explicit connection with the receiver.

A UDP header includes the following fields, each 16 bits:

- Source port.
- Destination port.
- Length, total size of UDP header + payload. Max size is $2^{16} - 1$ - IP header size bytes.
- Checksum, used for error detection. Set to 0s if unused.

Transmission Control Protocol

Another popular transport layer protocol implementation is *transmission control protocol*, TCP. TCP is reliable, bytestream abstracted, and connection-oriented in exactly the opposite sense of UDP. Reliable means that TCP implements application multiplexing, ACKs and retransmissions, flow control, and congestion control. Bytestream is the familiar concept of communicating using a byte stream abstraction instead of packets. Connection-oriented means a pairwise sender to receiver connection is established before sending data and used to maintain connection-specific state like initial sequence number, window scaling factor, window size, and more.

TCP Header

A TCP header includes the following fields:

- Source port.
- Destination port.
- Sequence number.
- Acknowledgement.
- HdrLen, length of the TCP header in number of 4-byte words. Minimum value of 5.
- 0.
- Flags.
- Advertised window.
- Checksum. 16 bits for error detection caused by bit corruption.
- Urgent pointer.
- Options.

The checksum for TCP is interesting. It's calculated over a pseudo IP header, the TCP header, and the payload. The pseudo IP header is extracted from the IP header and includes the source IP, the destination IP, a fixed 8 bit string of 0s, the 8-bit protocol field, and the 16-bit TCP packet length. TCP connection state is defined using a 5-tuple of source IP, destination IP, source port, destination port, and protocol, so the purpose of the pseudo IP header is to ensure the source and destination IP, plus the protocol fields are not corrupted.

Receive Buffers

The TCP receiver allocates a receive buffer to hold incoming bytes until the application reads them. The advertised window field in the TCP header tells the sender how much free space remains in that buffer and is used for per-connection flow control. TCP uses cumulative ACKs: the acknowledgement number reports the highest contiguous byte received and delivered to the application. Segments that arrive out of order can be buffered by the receiver, but they do not advance the cumulative ACK until the gap is filled; the receiver will commonly send duplicate ACKs (repeating the same acknowledgement number) to indicate the missing byte range.

On the sender side, transmitted but unacknowledged bytes are kept in a send buffer. Loss detection uses two mechanisms: duplicate ACKs and timers. Receipt of three duplicate ACKs triggers a fast retransmit of the missing byte(s) (avoiding the slow timeout path) because repeated ACKs imply a hole in the receiver's byte stream; waiting for three helps avoid retransmitting for transient reordering. If no ACKs arrive (for example when a single packet or the final packet is lost) a retransmission timeout (RTO) fires and the sender retransmits the oldest unacknowledged byte. Most TCP implementations maintain one retransmission timer per connection, reset it when new data is ACKed, and compute RTO from measured RTTs using conservative estimates to avoid spurious retransmits. With TCP's cumulative-ACK semantics, retransmission typically follows a Go-Back-N behavior: the sender restarts transmission from the first unacknowledged byte and proceeds from there.

TCP Connection

A TCP connection is identified with a 5-tuple of source IP, destination IP, source port, destination port, and protocol (which is always TCP). Connection set up is used to exchange the *initial sequence number* (ISN), as well as exchange the window scaling factor. A receiver needs to know the ISN of the sender to figure out which sequence number marks the start of the byte stream being received. Each endpoint chooses its ISN randomly.

A TCP connection setup is a 3-way handshake. Host A sends a synchronize packet (SYN) to B. Host B returns a SYN acknowledgement packet (SYN-ACK) to A. Host A sends an ACK packet to acknowledge the SYN-ACK.

The reason for this three-way handshale is to establish the ISNs.

- SYN contains sequence number equal to ISN of A, X. Its ack field is irrelevant

- SYN-ACK contains sequence number equal to ISN of B, Y. Its ack field is X.
- ACK has sequence number of $X + 1$, ack is $Y + 1$.

The special packets SYN and SYN-ACK are distinguished through flags in the header. The `flags` field is nine bits, each of which corresponds to a certain TCP flag.

Bit no.	Flag
0	FIN
1	SYN
2	RST
3	PSH
4	ACK
5	URG
6	ECE
7	CWR
8	NS

The ACK flag bit is always set if the header contains a valid acknowledgement number. All TCP packets, except SYN and RST-REPLY, have an ACK flag bit set.

TCP by default can do data batching on both send and receive side. Batching results in fewer packets sent, but adds delay. Setting the PSH flag disables batching.

If the URG flag is set, the urgent data in the TCP packet is delivered to an application immediate, bypassing the receive buffer. The urgent pointer in the TCP header stores the offset to the last urgent byte in the TCP segment, starting from the first byte in the segment.

ECE and CWR are used by congestion control protocols that use explicit congestion notification. NS is experimental.

NS stands for nonce sum, and despite the name is not a headcount of the ultra powerful but instead a cryptographic measure to protect against accidental or malicious concealment of marked packets from the sender.

When A wants to close the connection, it sends a FIN packet. B replies with a FIN-ACK. A replies with an ACK. B closes the connection upon receipt of the ACK, otherwise it sends the FIN-ACK again. A waits a little longer, seeing if there's going to be another FIN-ACK, then times out and closes the connection if none was received. In the case of the FIN packet, the `flags` field in the header would be `000010001`.

TCP endpoint can close a connection any time by sending a RST packet. There are two kinds:

- RST-ABORT: Generated when a connection is explicitly aborted by an endpoint e.g., the socket at the endpoint is being killed
- RST-REPLY: Generated on receipt of certain kinds of invalid packets (e.g., data packet for a connection that does not exist anymore)

If an endpoint receives a RST-ABORT, it closes the connection with no ACK. If an endpoint receives data for a closed connection, it generates RST-REPLY.

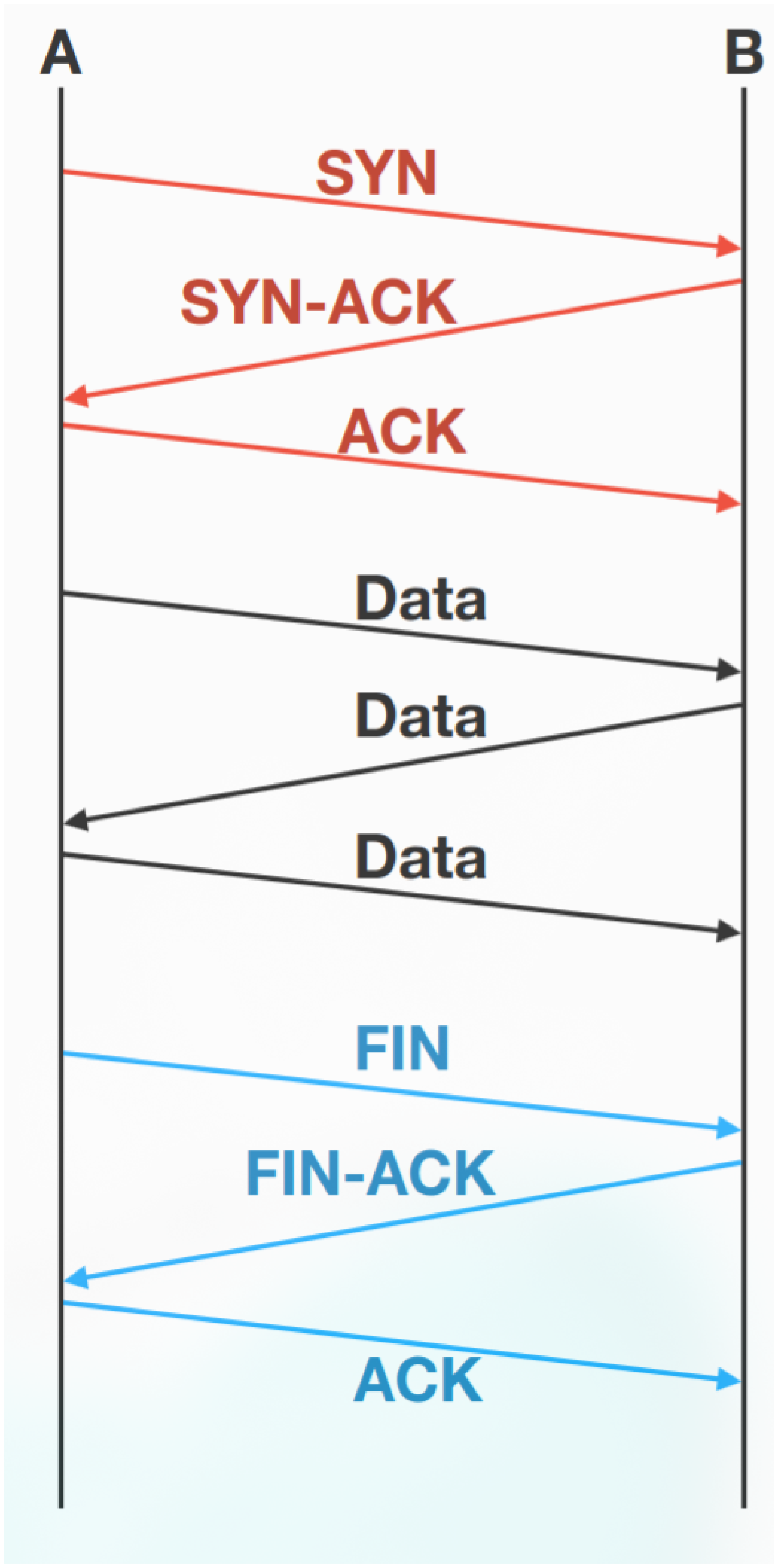


Figure 9: TCP Timing Diagram

Flow and Congestion Control

Senders in TCP maintain a sliding window of size W, as seen in Figure 10. The sender can have up to W bytes of unACKed data in flight.

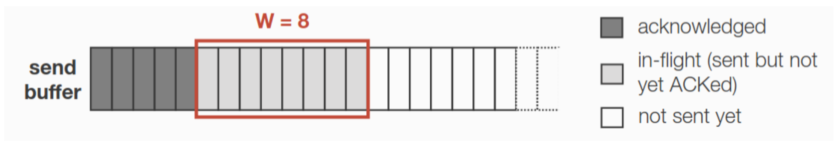


Figure 10: Sliding Window

The left edge of the window represents the beginning of in-flight data, which has been sent but not yet ACKed. The window slides when some data is ACKed to allow more to be sent. W determines the max rate at which the sender can send, the rate is approximately $\frac{W}{RTT}$ bytes per second.

The problem of sending data as fast as possible without overloading the receiver is *flow control*. The advertised window is a field in the TCP header, and the receiver sets it to the available receive buffer at the receiver. That is, the advertised buffer is set to recv buff size − ((next expected byte num − 1) − last byte num read by app). That is, the receive buffer can drop bytes that have already been read by the app.

Every sender uses the advertised window from the receiver to set its own window size W. Senders maintain a window called RWND. RWND is set to the advertised window value. The sender's window size W can be at most RWND, meaning the sender can have at most RWND unACKed bytes in flight, and can send at most $\frac{RWND}{RTT}$ bytes per second.

On the receiver side, if the buffer is too big then a lot of data may get queued at the receiver. If the buffer is too small, then the sender won't be able to fully use its bandwidth. The optimal size for the buffer is bandwidth delay product.

The advertised window field in the TCP header is only 16 bits long, limiting the sender window size to 65,535 bytes. This underuses bandwidth for networks with large BDP. Therefore, TCP endpoints also exchange a window scaling factor in the SYN packet during connection set up, and RWND is set to the scaling factor times the advertised window.

The problem of sending data as fast as possible without overloading the network is *congestion control*. The sender maintains a congestion window CWND, the maximum number of unACKed bytes a sender can have in flight. The CWND is increased upon lack of congestion, and decreased on detecting congestion. Congestion is detected by the number of packet losses.

The sender's slider window W is set to the minimum of the CWND and RWND to avoid overflowing the receiver buffer or router/switch buffers.

We talk of CWND in units of maximum segment size, the maximum amount of payload data in a TCP packet. Real implementations maintain CWND in bytes.

An oversized window is worse than an undersized window, since too many packets in the network affects everyone, while a slower sending rate only affects the sender. Ergo, the appropriate thing to do is gently increase when uncongested, and rapidly decrease when congested. If duplicate ACKs are received, then some packets are going through and the window size should be decreased aggressively, but less aggressively than in the case of a timeout, which means there were certainly several losses.

The balance that modern TCP implementations take is *additive increase, multiplicative decrease*. That is, on the success of the last window of data, increase the CWND by 1 MSS. On loss detected by 3 duplicate ACKS, set $CWND = \frac{CWND}{2}$. On loss detected by timeout, immediately drop CWND to one MSS. The factors aren't importance, but an increase factor of 1 and decrease factor of 2 works best in practice.

An issue with additive increase is that it ramps up very slowly when there's no congestion. For instance, consider an RTT of 100 ms and MSS of 1000 bytes. If there is 1 Mbps of available bandwidth, then the ideal window size (recall that the ideal window size is BDP) is 12.5 MSS. If the bandwidth increases to 1Gbps, then it takes 12500 RTTs to get to the new ideal window size of 12500 MSS.

The mitigation for this is to start with a small congestion window and increase exponentially, only for as long as CWND is small. After that transition back to AIMD. We introduce a slow start threshold `ssthresh` initialized to a large value, and when the CWND outgrows it, switch from slow start to additive increase.

To summarize slow start, TCP uses slow start for fast ramp ups when CWND is very small, and otherwise uses AIMD. Starting with a CWND of 1 and doubling every RTT, TCP leaves slow start when CWND >= `ssthresh` and enters the additive increase phase, where CWND is increased by one MSS on success of last window of data. Cut CWND to half on three duplicate ACKs, set `ssthresh` to the new CWND. Cut CWND all the way to 1 MSS on timeout, and set `ssthresh` to the old CWND divided by 2. Never drop CWND below 1 MSS.

Another problem is that congestion avoidance is too slow in recovering from an isolated loss of three duplicate ACKs. To optimize, we implement fast recovery. The idea behind fast recovery is to grant the sender temporary credits for each duplicate ACK to keep packets in flight. Each duplicate ACK is received because a new packet is arriving at the receiver, so the congestion isn't that bad and the loss was isolated. When three duplicate ACKs are received, TCP:

- enters fast recovery mode,
- cuts `ssthresh` to half of CWND, and
- sets CWND to half of CWND plus three (the number of credits for duplicate ACKs received thus far).

While in fast recovery, TCP increments CWND by one for each additional duplicate ACK and exits fast recovery after receiving a new ACK. On exit, it sets CWND to `ssthresh`, and starts additive increase back up.

Assuming RWND » CWND, throughput of a TCP flow is a function of congestion window size and round-trip time. The classical formula for throughput T with loss probability p is

$$T \approx \frac{MSS}{RTT}\frac{C}{\sqrt{p}}, \tag{6}$$

where C is a constant. The derivation assumes constant RTT and that the link delivers $\frac{1}{p} - 1$ consecutive packets followed by one drop.

Given a bottleneck link bandwidth B and N flows sharing the link, if we assume flow f_i demands bandwidth d_i, what is flow f_i's fair share of bandwidth b_i? We answer the problem with *max-min fairness*, where all flows demanding less than fair share $\frac{B}{N}$ get their demand, and remaining flows divide remaining bandwidth equally. In a stable state (no dynamics, all flows infinitely long, no failures, etc.) TCP congestion control (AIMD) guarantees max-min fairness.

There are other ways to ensure fair scheduling. For instance, packet round robin sends one packet per non-empty queue in a round robin manner. Weighted round robin sends w_i packets per non-empty queue i in a round robin manner. This approach fails with variable packet lengths, because queues can get more service with bigger packets. Another method is bit-by-bit round robin, where 1 bit per non-empty queue is sent in a round robin manner. The clock ticks by 1 once a bit from all non-empty queues has been transmitted, which is referred to as a *round* or *virtual time*. A packet's virtual finish time is independent of future packet arrivals. Say there is a router which can send one bit per unit real time, and there are two queues with the second initially empty and the first with 200 bits. Then the real finish time for the first is 200, and the virtual finish time is 200. If a packet arrives in the second queue of 100 bits, then the real finish time for A becomes 300 but the virtual time is maintained by sending two bits from A per unit virtual time. This results in max-min fairness regardless of packet sizes, but it isn't practical. The best overall algorithm is fair packet queuing (FQ), a packet-based version of bit-by-bit round robin. The idea is to determine the virtual finish time of each packet at enqueue into a queue according to the bit-by-bit round robin scheme, then send packets in order of increasing virtual finish time.

Shortcomings

TCP is:

- susceptible to DDoS attacks
- not designed for short flows
- hard to load balance
- biased against long RTTs
- designed for a cooperative environment, making it easy to cheat

A *denial of service* (DoS) attack is an attack where the perpetrator makes a resource unavailable to its intended users by exhausting some scare resources, such as CPU cycles or network bandwidth. *Distributed denial of service* attacks occur when an attacker is spread out and attacks from many endpoints. Such attacks include:

- SYN flood
- LAND attack
- UDP flood
- DNS amplification attack

DDoS attacks can be countered with firewalls or CAPTCHAs.

The second issue is that TCP is not designed for short flows. Internet traffic is a mixture of elephant and mice flows. Elephant flows carry most of the bytes (more than 95 percent) but are very few (less than 5 percent). Mice flows carry very few bytes, but comprise the vast majority of all flows. Elephant flows care about throughput, pushing as many packets per unit time from A to B as possible. The latency of the connection is amortized over the duration of the flow. Mice flows care about latency. When you send only one or two packets, the amount of time it takes a packet to go from A to B matters a lot. The four key characteristics of TCP congestion control, the 3-way handshake, gradual increase in window size/sending rate, duplicate ACKs as loss signal, and decreasing the rate only when loss detected are all designed with elephant flows in mind.

The third issue is that it's hard to load balance TCP. It is desirable that packets within the same TCP connection take same path, because this avoids packet re-ordering so 3 duplicate ACKs are only generated on packet loss and hence TCP only slows down when there is actual congestion/loss. However, assume there are three paths from source to destination. It would be wonderful to use all three and multiply the throughput, but TCP isn't designed for this.

Multipath TCP is an extension of TCP that allows a single TCP stream to take multiple paths, but it's a lot more complex.

The fourth issue is that flows get throughput inversely proportional to RTTs. Flows with long RTTs will achieve lower throughput.

The fifth and final issue is that it's very easy to cheat in TCP. For example, a cheating client could increase CWND by a large amount each ACK. A cheater could also break a single connection into many connections to hog more bandwidth.