

Computer Networks

Lecture 1 Introduction to protocols and protocol layering

This course covers the protocols used in the internet.

What is a protocol?

It is a set of rules for transmission and formation of data.

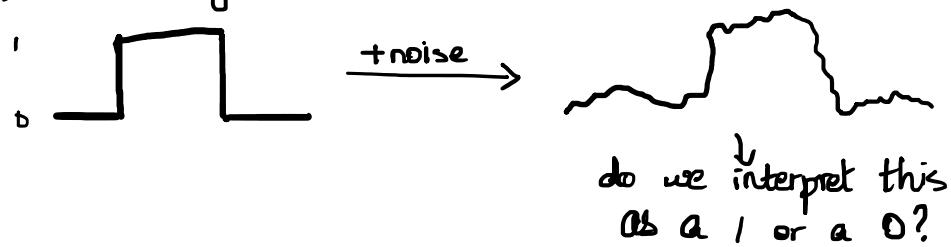
Textbook: Peterson & Davie - Computer Networks: A System's Approach
(6 ed.)

The internet started out in the late 1960s as a government project ARPANET.

In different places there were different types of media (wired/wireless) and applications (telnet/rlogin / FTP). The internet is a network that works despite the massive amount of diversity. How would one even design this?

Say you have just two people A and B. You want a way to communicate data. We usually convert it into bits and send the signal across the wired or wireless medium.

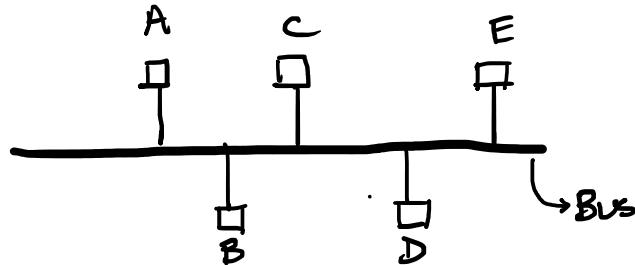
However, there might be (electronic) noise in the medium.



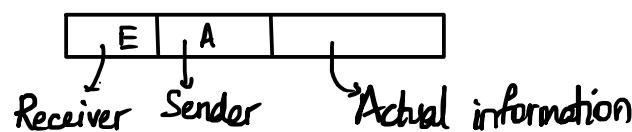
The noise is higher in wireless media.

Denoising, signals and other similar things come under the physical layer of the network.

Now take a slightly more complicated situation where you have 5 people. A simple solution is to have a bus like so:



Say A wants to talk to E. Then A should be able to indicate that the message is meant for E. If two-way, then they should also be able to identify themselves.



What if two pairs, say A→E and B→C want to communicate at the same time? The signals may get superimposed and both the informations completely garbled.

This issue is known as **medium access**.

see: MAC address
(Medium Access Control)

↑ unique for every device,
hardcoded in hardware

Medium Access

Data Link Layer

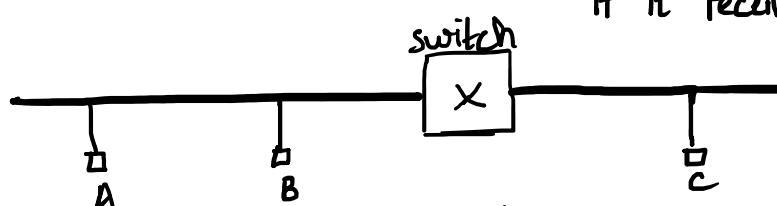
Things like medium access and framing come under the **data link layer**.

(DLL)

↓
demarcating where the information begins/ends.

Repeater

Now say we have 1000 people. The medium access problem grows massive if we use a single bus. Since the bus will be so long, we would also need a **repeater** (an amplifier) to ensure the signals don't die out. There would also be a lot of interference. We could use a switch instead:

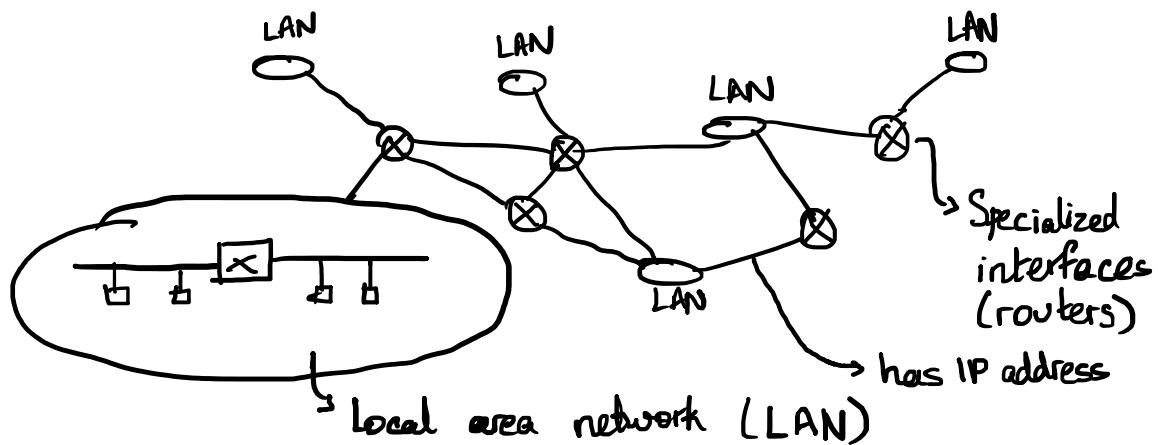


if it receives a message from A→B, then it ignores it. If say A→C, it repeats.

basically a selective repeater

This comes under the data link layer as well.
A switch used in this scenario is called a Layer-2 switch. (L2)

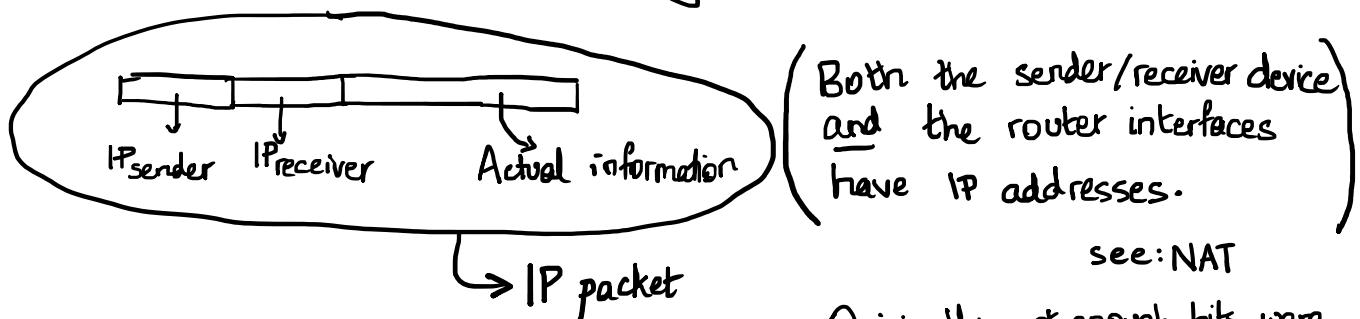
Now say I have a million people.



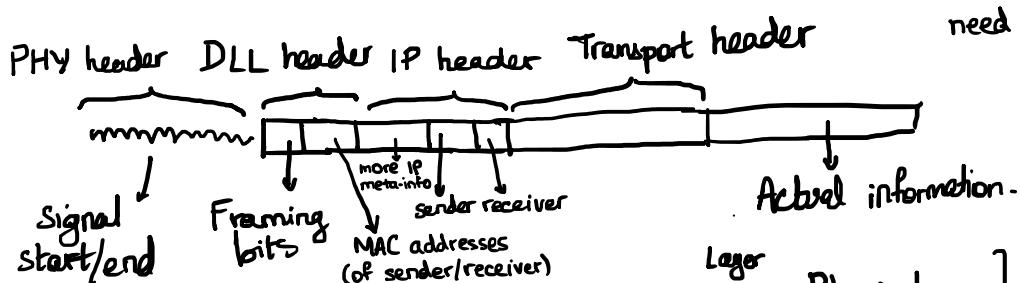
We need some common protocol so that the diversity does not cause issues
IP protocol is the common one

Network Layer (IP)

Each of these interfaces will have IP addresses (one for each link)
If someone wants to communicate, they can use the IP address.



So overall,



Originally, not enough bits were allocated for IP, so they need not be unique.

see: private and public IP address

Layer	Physical
1	Physical
2	Data Link
3	Network (IP)
4	Transport
5	Application

5 layer model of protocol stack

Routers **Routers are Layer-3 switches.**

The router's job is to see the IP header and decide where to forward it (if at all). How do they work together to figure this out?

↳ There are many protocols for this.

BGP, ISIS, OSPF

autonomous system
level routing

Layer-3 protocols take care of routing.

Routers have queues to temporarily buffer packets when the input rate is higher than the maximal output rate.

Packet loss If the queue gets filled, then **packet loss** occurs (packets are just thrown away).

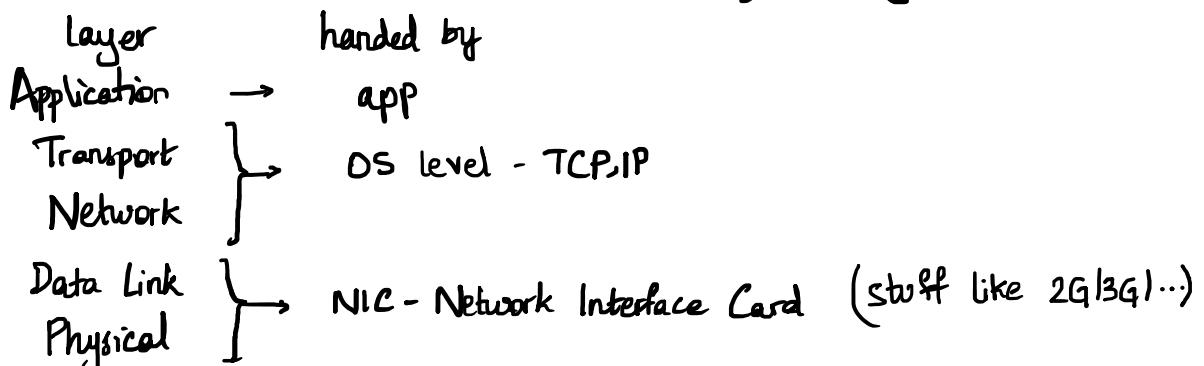
We do retransmission of lost packets. In ARPANET then, the retransmission starts from the original sender.* This isn't always the best solution, but ARPANET was a defense network so it makes sense. (This relic is even in today's internet?)

So loss must be detected and the message must be retransmitted until the receiver receives it.

↳ To know this, there should be a returned acknowledgement packet from the receiver. If it doesn't come for a long time, package lost?
when we send a lot of packets, we needn't worry if a few acknowledgements are lost

Transport Layer (TCP) This layer is known as the **transport layer**. The common protocol used is TCP. (In ARPANET, the rudimentary UDP was used)

Some other models use more than 5 layers. (Eg. OSI-7 layer)



Headers are added on as the message moves down.

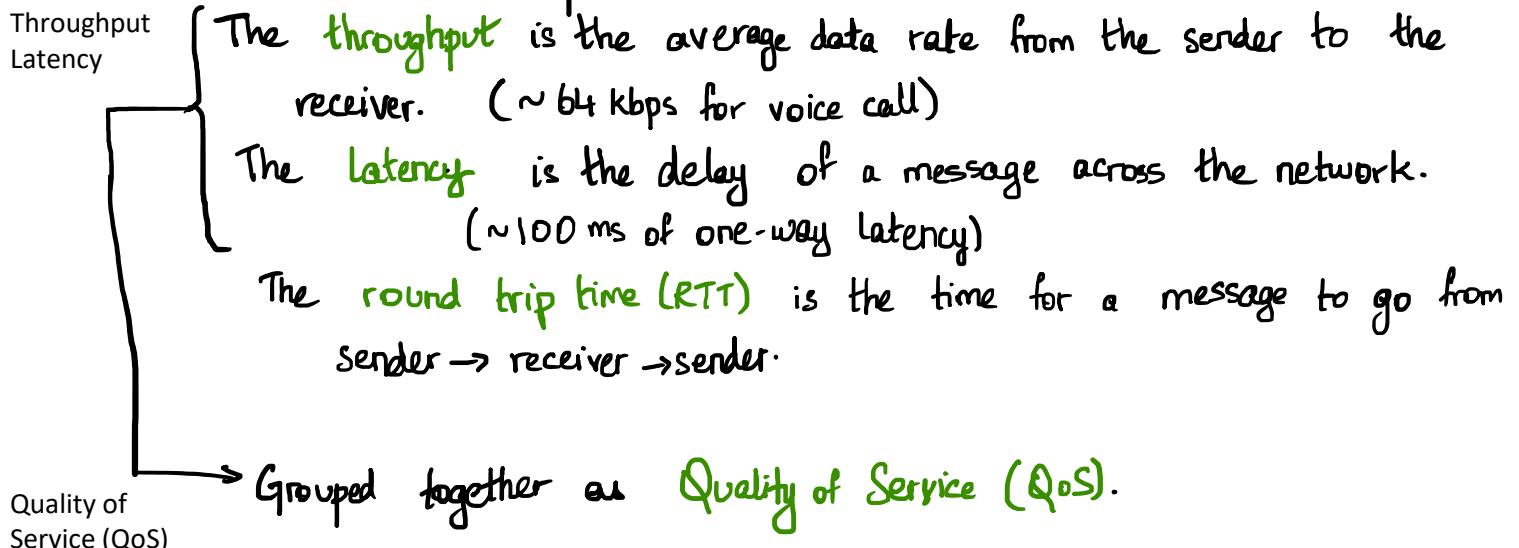
The message is transmitted to the base station
↓
only looks at network/DLL/PHY in general

Repeater only looks at PHY.
L2 switches only look at DLL/PHY.

Why is protocol layering done?

- + The complicated networking problem is decomposed into smaller, more manageable ones.
- + Modular design so you can change individual protocols easily.
- It may help if TCP knew where the lost information was dropped.
The lack of interaction between distinct layers makes this difficult.

How well does it perform?



Say video streaming → throughput ~1 Mbps needed
latency doesn't really matter, couple of seconds would also be fine.

When we build a network, we specify the QoS and then design the protocols. We don't have this freedom when using the internet however, we can only interact with the application layer meaningfully.

Throughput is the number of bits that can be pushed from one end to another.

Latency is the time taken. (Property of network, not application)

Say file transfer → the higher the throughput, the more preferred.
 if there is no such "critical" required throughput,
 it is said to be **best effort**. No QoS guarantees
 from the network.

The internet is designed as best effort.

We are trying to move away from this and guarantee QoS, but it's difficult. (Mobile-mobile communication does guarantee QoS)
 of transmission

In a router, the maximum data rate through a certain link is called **bandwidth** (link capacity).

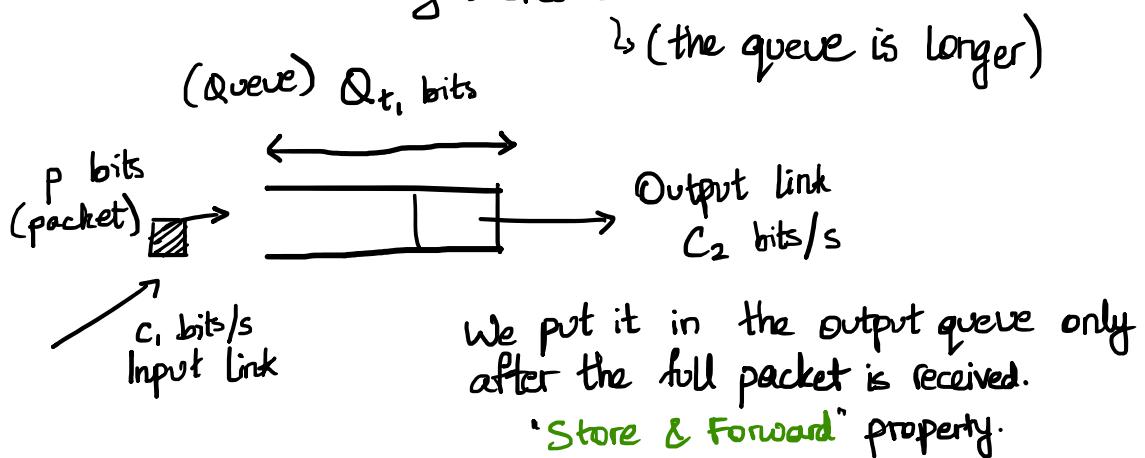
If you have multiple people using the same link, they are competing.

The protocols don't usually guarantee/reserve any bandwidth.

(some protocols do but making every router guarantee stuff is non-trivial)

If people are competing, throughput decreases.

latency increases.



In **Cut-Through** switching, we send it to the Output queue immediately.
 This may waste resources though (what if the packet is corrupted?).

Say t_0 → time of first bit at input

t_1 → time of last bit at input

Then $t_1 = t_0 + \frac{P}{c_1}$. (In S&F, we wait till t_1)

$t_2 \rightarrow$ time of first bit at output.

$$t_2 = t_1 + \frac{Q_{t_1}}{c_2}$$

t_3 → time of last bit at output

$$t_3 = t_2 + \frac{P}{c_2}$$

$$\Rightarrow t_3 = t_0 + \frac{P}{C_1} + \frac{Q_{t_1}}{C_2} + \frac{P}{C_2}$$

↓ ↓ ↓
 Waiting due to Queuing Transmission
 Store & Forward delay delay

Quite variable, depends
 on traffic

↳ number of people competing.

Cross-Traffic

Cross-Traffic → Traffic other than your own packets.

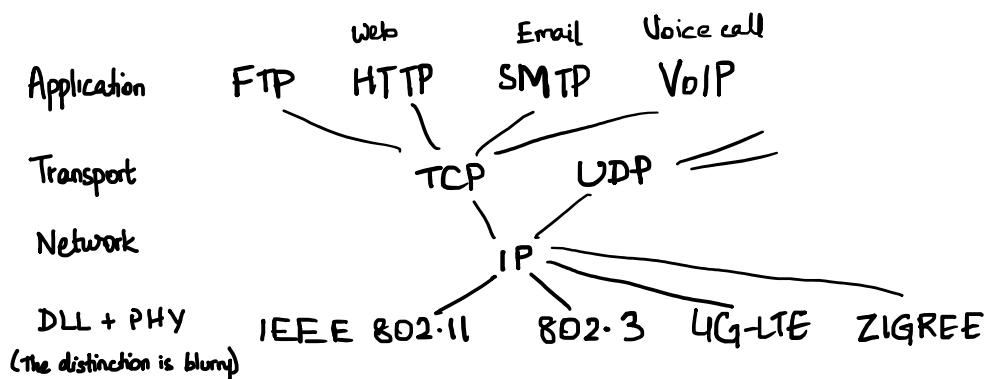
Congestion control → When the queue builds up at a router, "congestion"

Congestion Control is said to occur. We want it to continue working somehow—the way this is handled by the protocols is congestion control. Mainly handled by Transport layer (TCP).

(if packets being dropped or RTT very high, back off a bit)

May also be "handled by lower layers" (DLL, PHY).*

If you want your own congestion control in the application layer, you might use UDP (which does nearly nothing special)



- * if collision of signals, back off a bit and then retransmit. DLL has some stuff to do with retransmission as well

Since everyone uses IP, hourglass like figure.

Designed such that layers only need to talk to higher and immediate lower layer.

