

RCOMP - Redes de Computadores (Computer Networks)

2025/2026

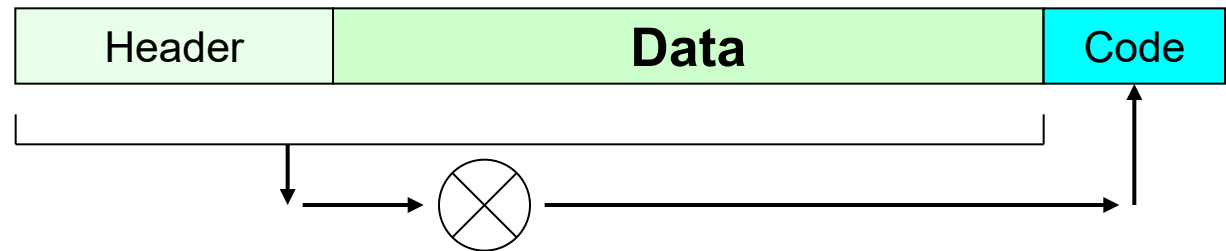
Lecture 02

- Error detection.
- Network delays.
- Flow control.
- Error control.
- Network architectures.
- The OSI model.
- The IEEE 802 Architecture (ISO 8802) .
- The TCP/IP Architecture .
- The client-server model.

Error detection code

While a packet is being transmitted, it will not be unusual a symbol getting the wrong interpretation at the receiver. Errors may occur, meaning part of the received data will be different from the originally sent data.

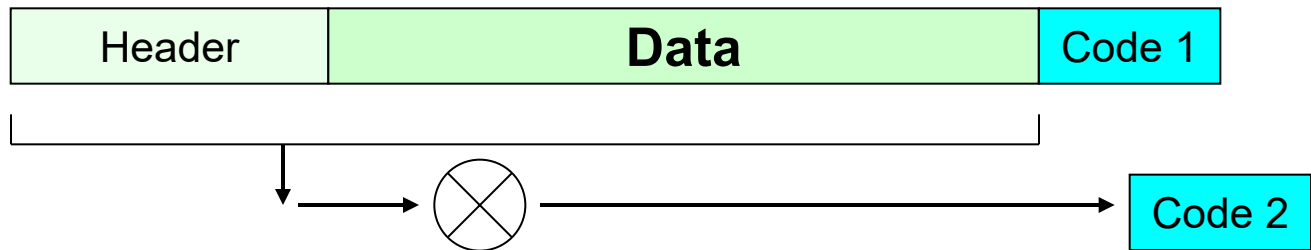
These errors may be detected as far as the sender adds a validation code to the packet. The code is generated by an appropriate function taking the content of the packet as input and producing a small fixed size code representing the content. This function will produce a different code if there is any small change in the input data.



The sender uses the function to calculate the code, and then, adds it to the packet. In some cases, the code is added to the packet header, but it can also be added to the tail as presented on the image above.

Error detection

The code produced by the sender is sent along with the packet. This gives the receiver the ability to repeat the process and confront the two codes it will then have:



At the image above, **Code 1** was calculated by the sender and **Code 2** by the receiver.

If the two codes differ, then the receiver knows **there has been an error**. If the two codes are equal, the receiver knows **probably there was no error**.

In fact, this method never absolutely guarantees there was no error. There is always a remote chance that a random change in the packet due to transmission errors may yet result in a packet and code that match each other. **Though this is highly unlikely to happen by chance.**

Forward Error Correction (FEC)

Detecting errors is just the first step, once detected, something must be done about that. Error correction, also called **error control**, handles the problem once an error is detected.

One approach for error control is **FEC**, it requires the use of more extended validation information (not a simple code) and special functions to handle it. For errors with up to some extension, FEC allows immediate and autonomous correction by the receiver.

The other approach is **Backward Error Correction (BEC)**, it requires the receiver to ask the sender for the packet to be retransmitted. This is also known as Automatic Repeat Request (ARQ).

Due to this dialogue, the network delays between the sender and the receiver have a significant impact on BEC efficiency.

FEC, though, is not a universal solution, it only solves errors if they are limited to a short number of bits. When used, FEC must be combined with BEC for the cases when errors are more extensive and can't be solved by FEC.

Network delays - packet delivery time

Transferring a packet, from the sender to the receiver, through a network takes some time.

The total time it takes for the transfer to be fully accomplished is called the **packet delivery time**. It measures the elapsed time since the sender emits the packet's first bit until the packet's last bit arrives at the receiver.

The **packet delivery time** is the sum of two parcels:

- **Packet transmission time** – this is the amount of time it takes to emit or receive a given size packet at a given data rate.

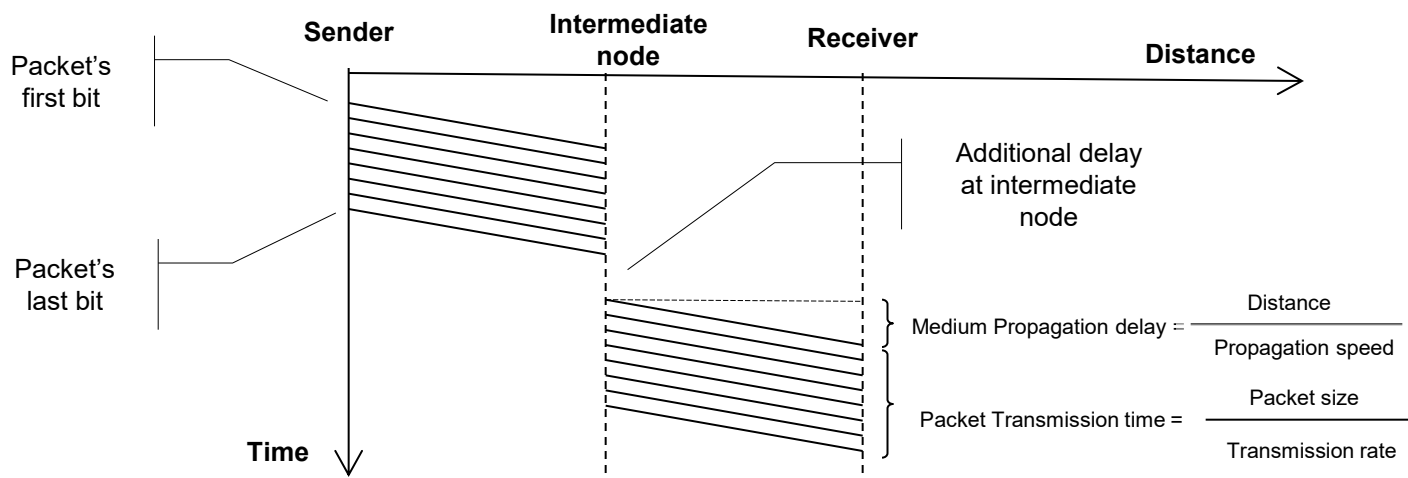
$$\text{Packet transmission time} = (\text{Packet size}) / (\text{Data rate})$$

- **Propagation delay** (network latency) – this is the amount of time it takes for one bit/symbol to travel through the network from the emitter to the receiver. If sender and receiver are connected to the same transmission medium (**no intermediate nodes**), then:

$$\text{Propagation delay} = (\text{Distance}) / (\text{Signal propagation speed})$$

Network delays – network latency

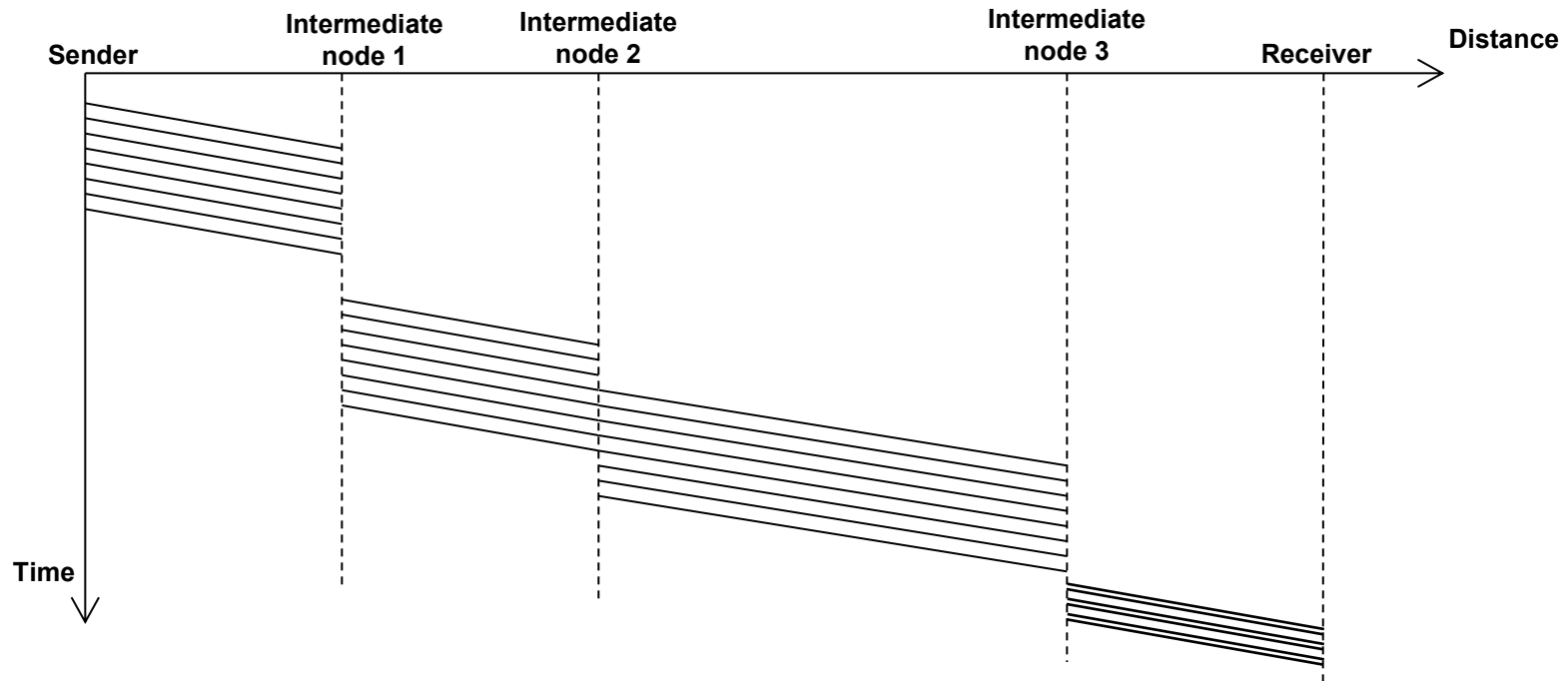
In a switching network, there are intermediate nodes between the sender and receiver, so things get far more complex. Each intermediate node will add some more time to network latency (propagation delay), it also receives and emits the packet, so the corresponding transmission time has to be added.



As we can see on the image above, the latency added by an intermediate node equals the **packet transmission time**. However, intermediate nodes may not be able to send the packet as soon as it's received, this introduces an **additional delay**. Intermediate nodes use queues (FIFO) to store received packets before emitting them, these queues operate as buffers and are essential because the intermediate node may not be able to emit packets at the same rate they are being received.

Network delays – network latency

Intermediate nodes may introduce other features affecting network latency calculation, the graphic represents a packet path through a switching network:



Intermediate nodes 1 and 3 fully receive packets before starting to emit them, this is called **store & forward**. Some types of networks allow, however, another operation mode called **cut-through**, that is the case of intermediate node 2. Another factor to take account for is that intermediate nodes may receive and send at different data rates, that is the case of intermediate node 3.

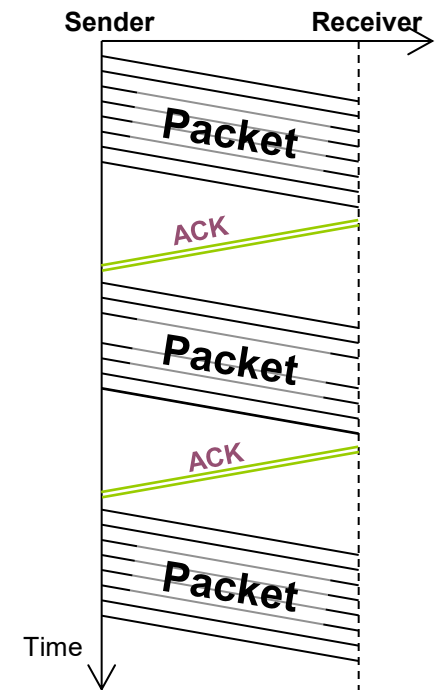
Flow control

The flow control goal is regulating the data flow between the sender and the receiver, and thus, avoiding an overflow at the receiver (receiving more packets than it can cope with).

The best way to achieve this is letting the receiver control the flow. In the technique called **stop and wait** the sender is forced to wait for the **ACK signal** from the receiver before sending the next packet:

The image on the right shows how inefficient this flow control technique may become due to propagation delays. The network may be idle, nevertheless, the sender cannot use it until the ACK is received.

To overcome this **stop and wait** flow control issue, a variant known as the **sliding window protocol** exists. Instead of the emitter having to wait for an ACK before sending the next packet, it can send a burst of W packets. The W parameter is the **window size** and it's configurable.



Flow control – sliding window protocol

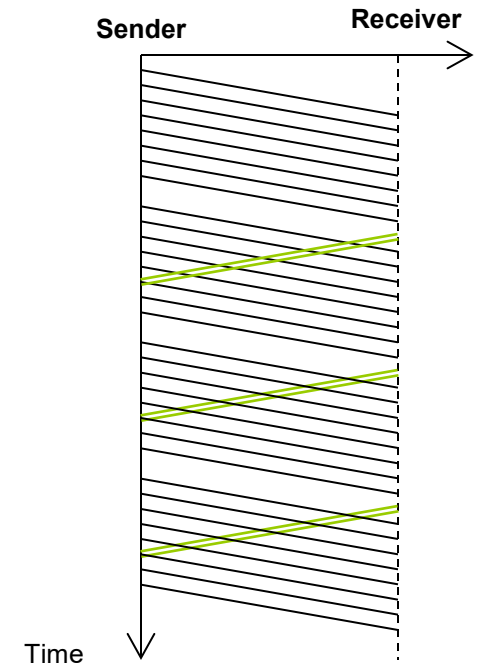
With the sliding window protocol, the sender is allowed to send a burst of W packets (window size) without receiving any ACK from the receiver:

After sending W packets, the sender must then wait because the window is exhausted, however, for every ACK that meanwhile arrives from the receiver, an additional packet can be sent.

Under ideal conditions (appropriate W value), there are no stops in packet transmission. This will happen if the ACK for the first packet arrives before the window is exhausted (W packets are sent).

It's important to point out that the absence of stops (100% efficiency) is possible only in a full-duplex link (simultaneous transmission in both directions).

Notice that switching networks are inherently full-duplex because even if some links are half-duplex, packets traveling in opposite directions can cross each other at intermediate nodes.



Error control

The purpose of error control is correcting errors found by error detection. Although self-correcting mechanisms are available (FEC - Forward Error Correction), in most cases the receiver will ask for the retransmission of another copy of the same packet (BEC - Backward Error Correction).

Retransmission requesting is known as ARQ (Automatic Repeat Request), it may be implemented together with flow control. For this purpose, the receiver can reply to the sender either with ACK or NACK. Then the NACK means an error has been detected, and as such, the packet in reference must be retransmitted.

When used together with the sliding window protocol, this retransmission request mechanism, becomes known as **continuous ARQ**.

To use the sliding window protocol packets will have to be tagged (usually with a sequence number). This is required because in case of an error the receiver will have to tell the sender which packet is to be retransmitted. For instance, the receiver will send NACK 5 to request the retransmission of packet number 5.

Network Architectures

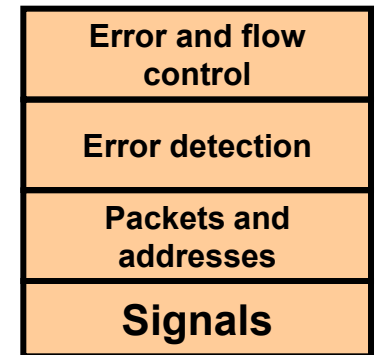
Communication between applications located in physically apart systems is a complex process, it involves many issues that must be addressed.

Due to this complexity, since the dawning of computer networks, in the 70s, a strategy of successive modules was adopted, usually known as layers. Each layer solves one part of the problems.

How these layers are organized and how they interact with each other is known as the network architecture or network model.

The main principle is, each layer uses the layer below and adds new features that will be available to the layer above it.

The image on the right summarizes some of the main networking issues we have addressed so far in this course.

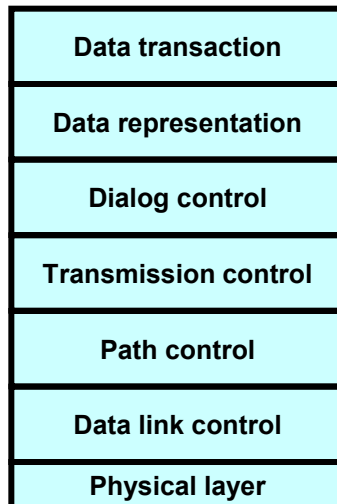


Proprietary Architectures

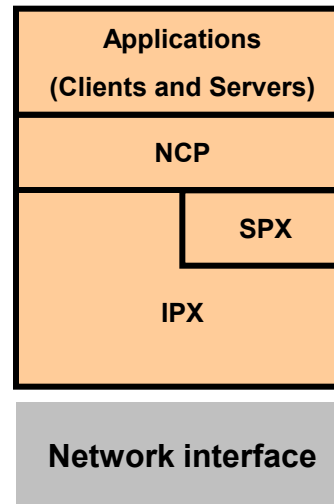
Computer networks have started to arise spontaneously in the early 70s. At that time, each manufacturer developed its own closed system, following a patent culture to avoid being copied by others. These architectures are known as proprietary architectures, some of them are:

IBM

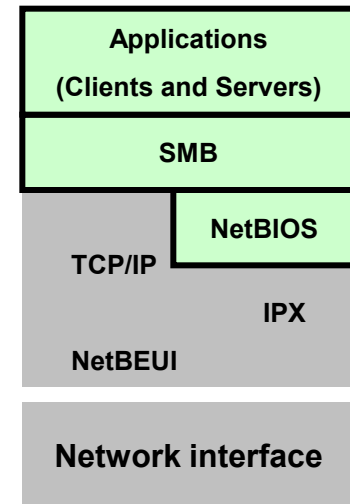
SNA (Systems Network Architecture)



Novell NetWare

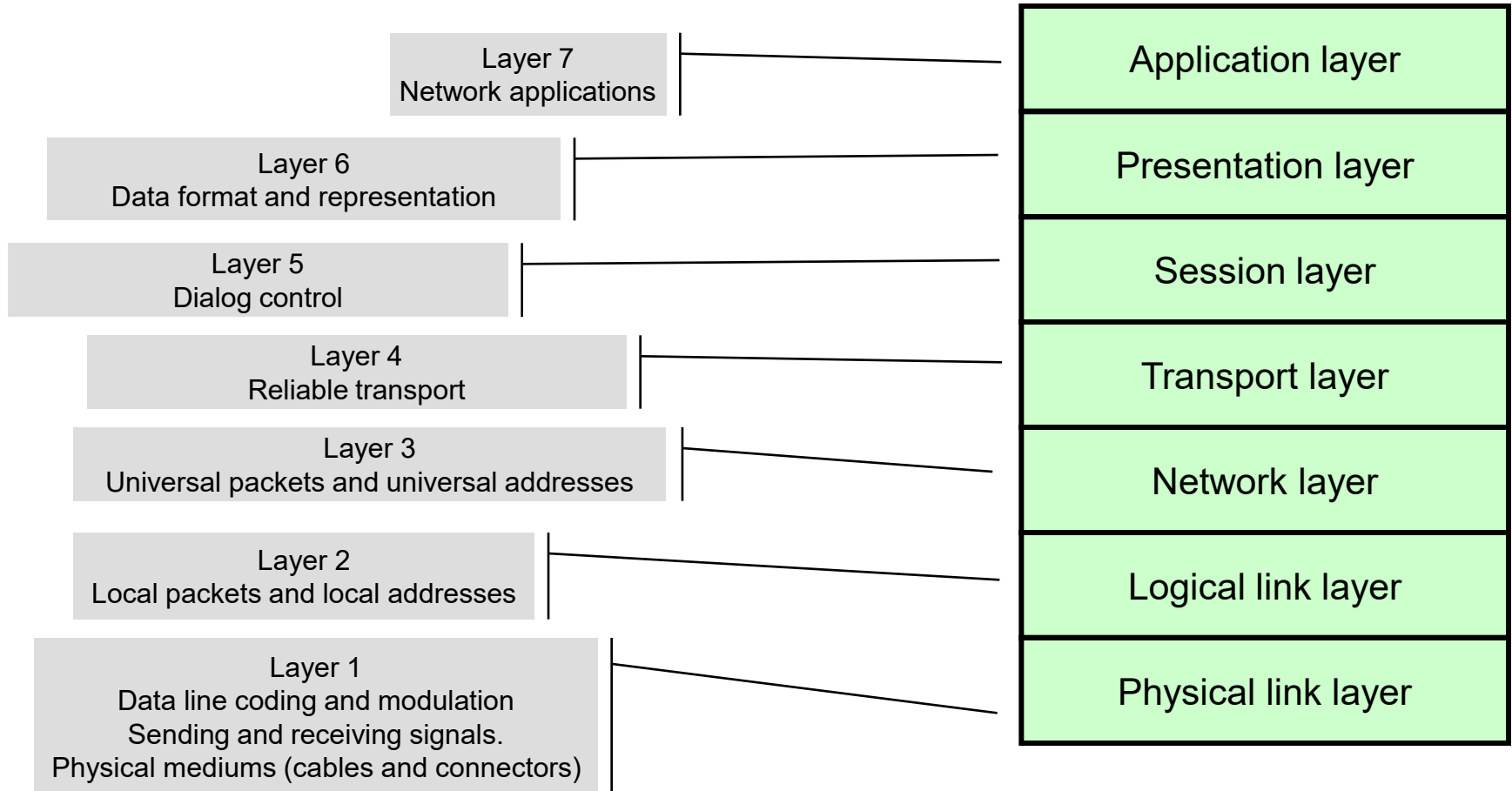


Microsoft NetBIOS/SMB



The OSI model

In an effort to normalize network architectures, ISO (International Organization for Standardization) has established the OSI (Open Systems Interconnection) model. This model describes seven layers:



The OSI model - layers

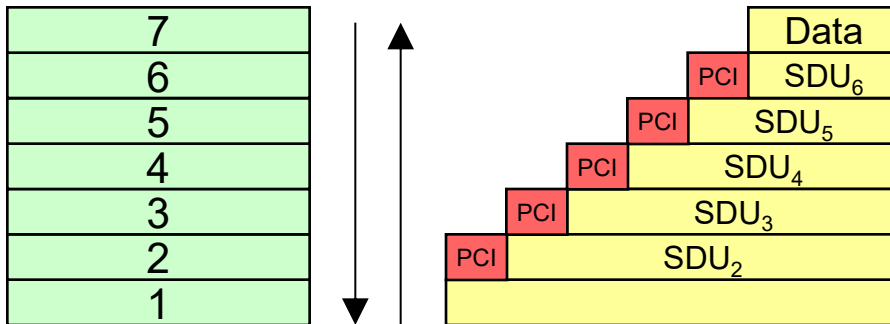
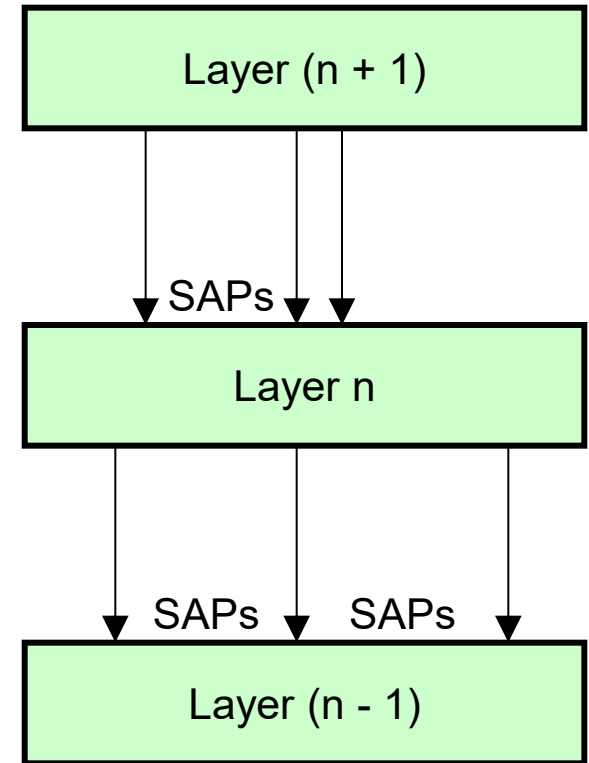
Successive layers of the stack interact with each other according to a downward service call model through service access points (SAP).

Each layer uses the services provided by layer below and adds them new features.

Typically, the new features implemented by each layer require the addition of control information (PCI - Protocol Control Information).

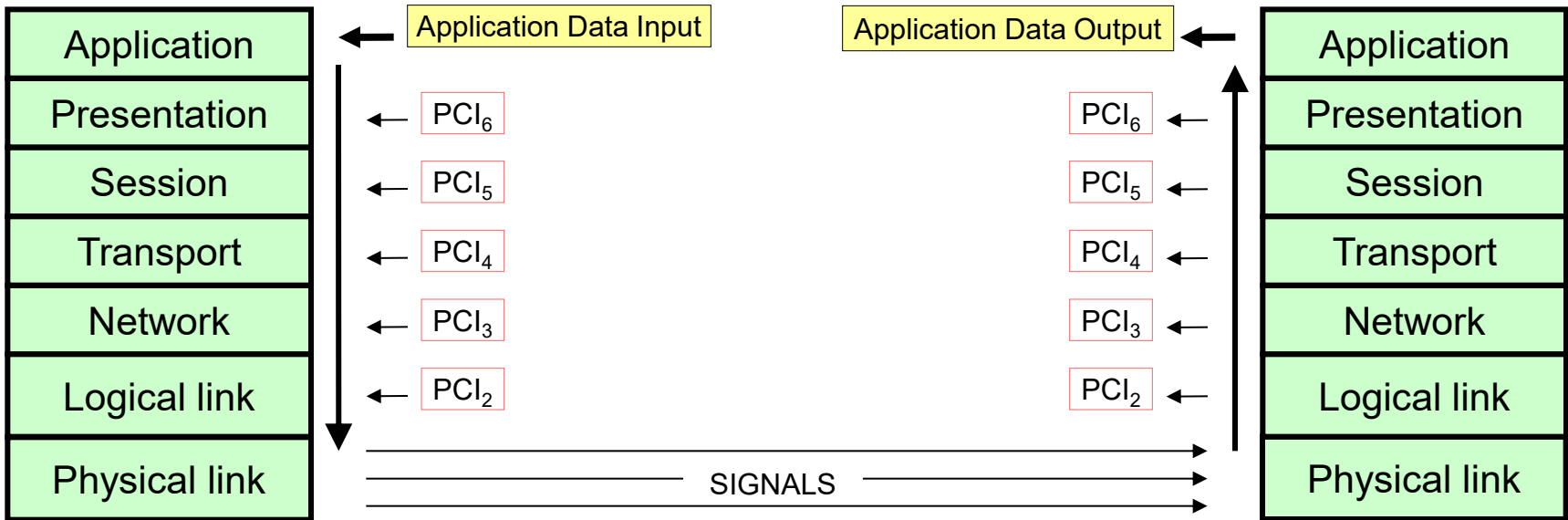
The PCI is added to the data (SDU - Service Data Unit) coming from the upper layer, i.e., the payload.

In each layer, the PCI and the SDU together make the PDU (Protocol Data Unit), i.e., the packet.



The OSI model - protocols

Direct interaction happens only between successive layers, and in the physical layer. However, the same level layers, hosted in different network nodes will communicate with each other using the specific PCI of that same layer.



The exchange of information via each layer's PCI is related to that layer's specific features and follows a set of rules known as protocol. Thus, **each layer has its own protocol.**

The OSI model as a reference

Beyond the architectural description, OSI also tried to force some concrete protocol implementations for some layers.

However, most of such concrete implementations proposed in OSI were never widely adopted in real computer networks. One reason for that is manufactures where not willing to give away their proprietary architectures and restart implementations from scratch.

While under the point of view of concrete implementations it has not been a success, the OSI model was a very important step because it includes a set of concepts, standards, nomenclature, techniques and ideas that have become a reference point for any discussion in the computer networks area. Thus, it's often referred to as the **OSI reference model**.

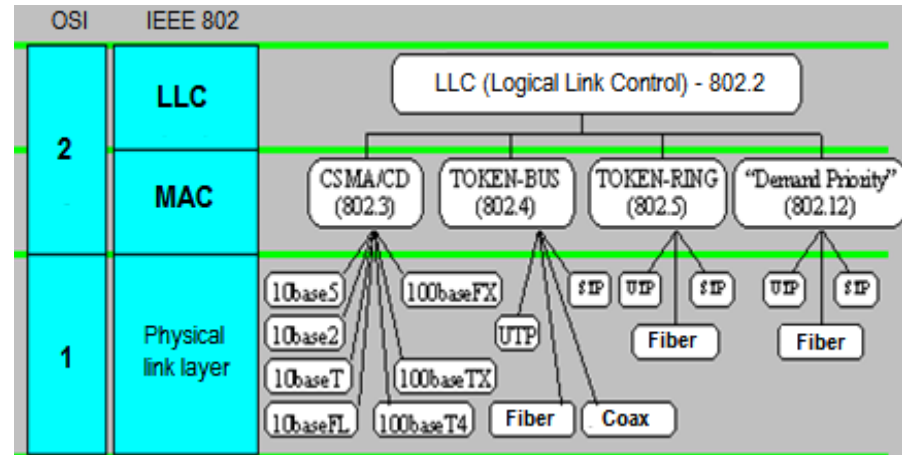
One OSI main motivation: unite all networks architectures to allow all nodes to talk to each other, was ultimately achieved by another architecture: TCP/IP.

TCP/IP architecture achieved it by becoming global through the internet, by doing so, once users everywhere start demanding an internet connection, manufacturers were compelled to adopt it.

The IEEE 802 Architecture (ISO 8802)

Most LAN technologies are standardized by IEEE and ISO, these technologies match layers 1 and 2 of the OSI model.

Each standard is identified by numbers and letters, for example Ethernet networks have the identifier IEEE 802.3 (ISO 8802-3). Whenever there are technical developments in these standards additions are made identified by lowercase letters.



Ethernet networks at 100 Mbps are defined in the 802.3u standard and the same Ethernet networks at 1 Gbps in the 802.3z standard.

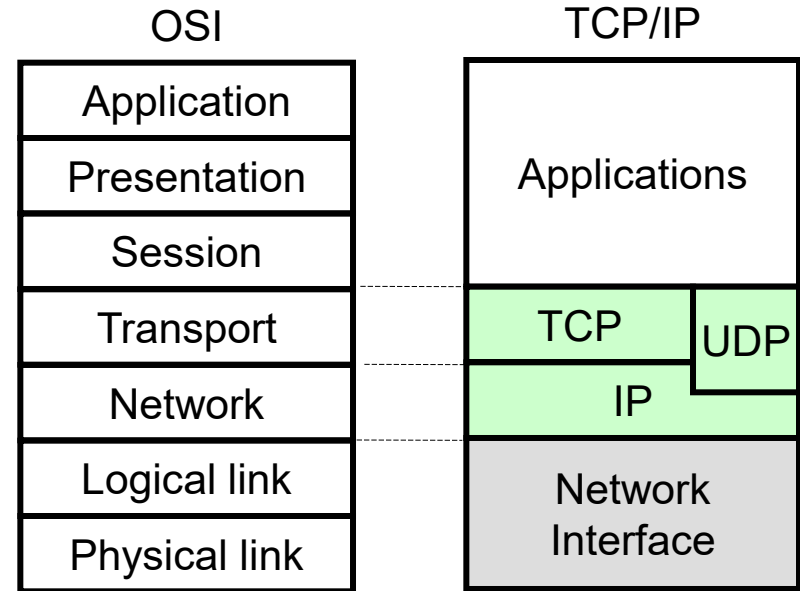
Most real implementations don't use the LLC layer; they interact directly with the MAC layer. This is possible because, beyond the **medium access control** mechanism itself, the MAC layer also implements all basic features to transfer data packets locally, it establishes node addressing and error detection.

The local networks evolve quickly to switching and access control mechanisms are now mostly not used. The big exception is IEEE 802.11 wireless networks.

The TCP/IP architecture

The TCP/IP protocol stack resulted from a pragmatic approach, in opposition to the OSI model it was developed without much theoretical planning, using a minimalist approach in which problems are solved as they arise in practice.

Nevertheless, TCP/IP was able to reach some of the initial OSI goals. Due to the internet generalization and the resulting requirement to use IP (Internet Protocol), there seems to be an irreversible migration trend of all systems to the TCP/IP stack, abandoning all other protocols. In this context, the interconnection of systems is certainly resolved.



The TCP/IP stack consists of several protocols in addition to the IP, the most important are **UDP (User Datagram Protocol)** and **TCP (Transmission Control Protocol)**.

UDP – **connectionless** packet protocol without reliability, only simple error detection.

TCP – **connection-oriented** reliable protocol with sliding window flow and error control.

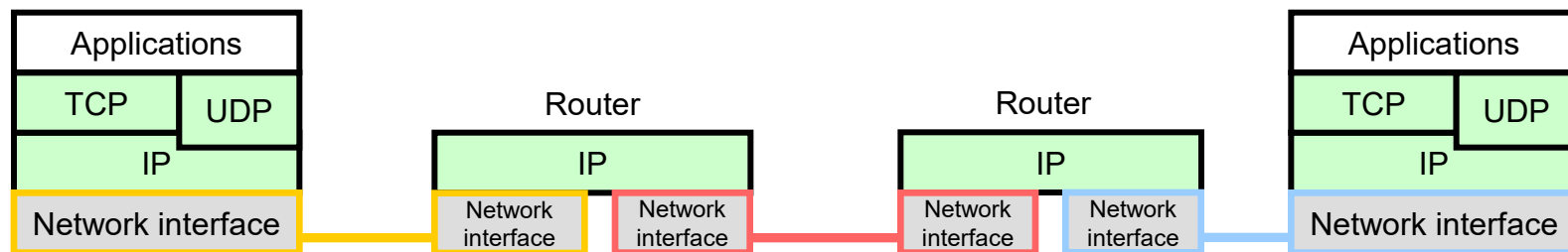
Of these two, the most widely used over the Internet is TCP.

IP routers

One advantage of TCP/IP is it doesn't define layer one neither layer two, the IP protocol is expected to use any existing layer two packet transmission technology. IP defines a universal packet format and universal node addressing, thus allowing the interconnection of different network types. This allows the building of a global network such as the internet.

However, when an IP node uses some specific layer two technology (LAN) it will be able to deliver only to other IP nodes directly connected to that same technology (LAN). To overcome this, routers come in play.

Routers are the intermediate nodes of IP networks; they are usually connected to more than one layer two network (network interfaces), and each may use different technologies. Thus, a router is able to transfer IP packets between two networks, even if each uses a different layer two technology.



The diagram above represents three different layer two technologies being used: yellow, red, and blue. By using IP, a node connected to the yellow LAN is able to communicate with a node connected to the blue LAN.

IP network addresses

IP also introduces the **network address** concept; the aim is simplifying the routing task. Instead of node addresses based (like switching in layer two) routing is network address based. By just looking at a node's destination IP address it's possible to determine to which network it belongs, each IP node address is made of two parts, one part identifies the network it belongs to, the other uniquely identifies the node within that network.

For applications, all these routing issues across heterogeneous networks are completely transparent, applications just use IP addresses along with the UDP and TCP protocols. It's up to routers analysing the destination IP address of each packet and find the appropriate path to reach such destination.

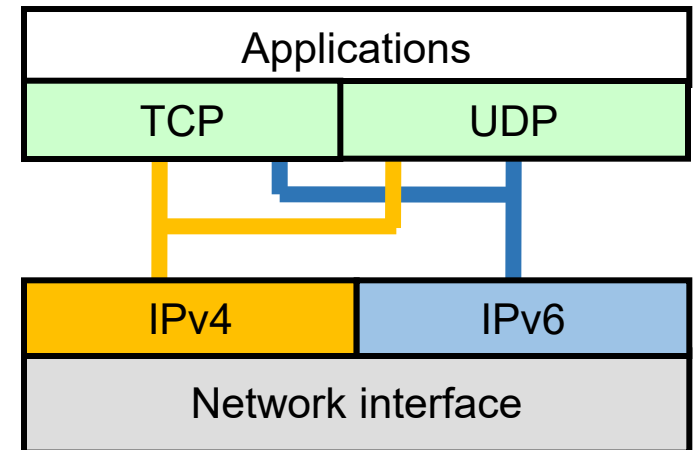
Although for IP the final destination for data is an IP node, for UDP and TCP protocols, final destinations for data are applications. They use 16-bit numbers to tag data, knowing thereby to which specific application in the destination node it should be delivered. These labels are known as port numbers or service numbers. Port numbers can be seen as addresses inside the node.

Internet protocol version 6

Since the internet has expanded, the most widely-used version of the Internet Protocol has been version 4, however, there is a new version 6, it's very slowly being introduced on the internet. One of the major differences between IPv4 and IPv6, is the increase on the size of node addresses, from 32 to 128 bits. Nevertheless, TCP and UDP protocols use IPv6 the same way they use IPv4.

To keep the internet running, the transition will necessarily be gradual, certainly, we are still going to have IPv4 on the internet for many years.

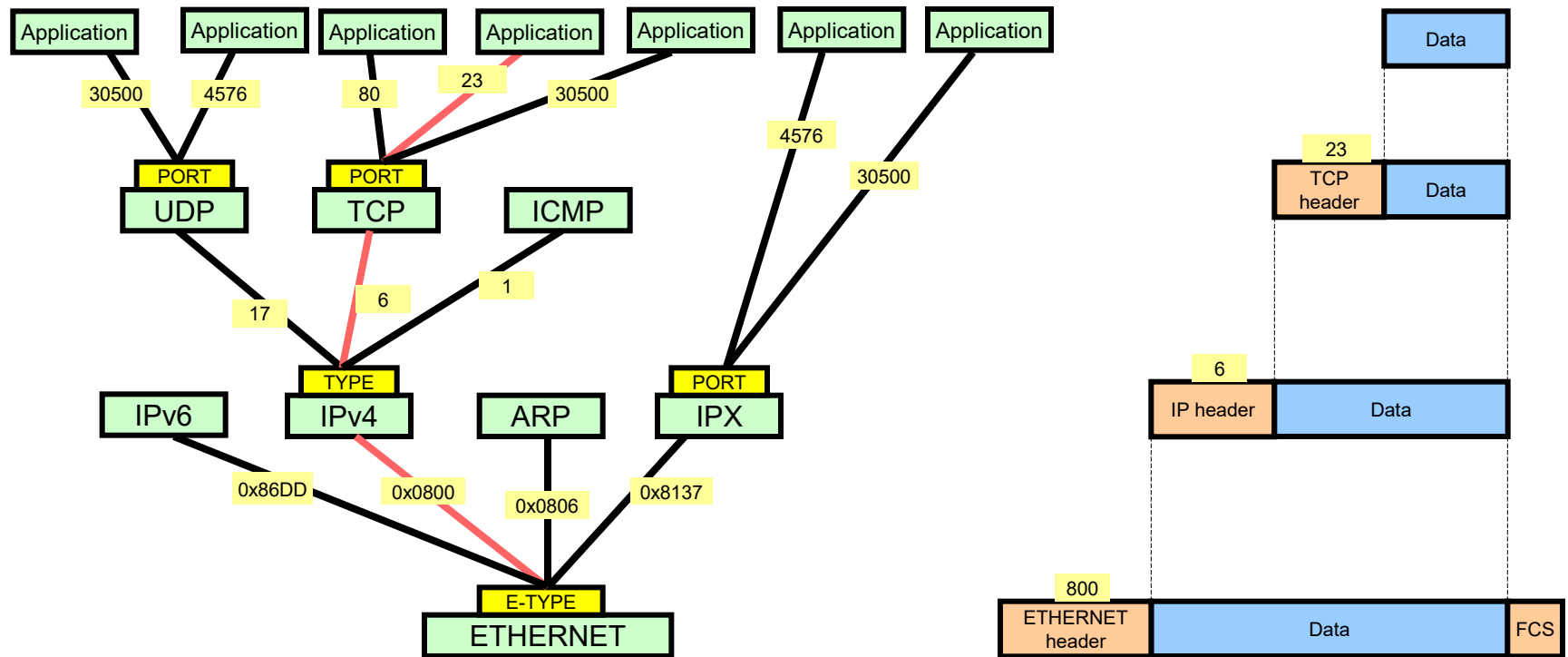
There are several co-existence strategies for IPv4 and IPv6, one of which is implementing both protocols in end nodes, this is known as **IPv4/IPv6 dual-stack** (right image).



As we can notice in the figure above, in each layer, there are frequently several protocols running. This is possible thanks to tag-based multiplexing. Tagging ensures data from a layer implementation X in one node, ends up being delivered to the same layer implementation X on the destination node.

Multi-protocol layers

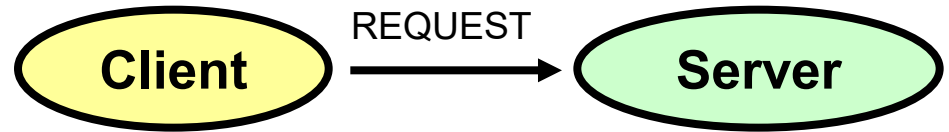
The coexistence of several protocols at the same layer means that there are parallel data streams which diverge (upward) and converge (downward). In order for the converged flows to diverge elsewhere, each of them has to be tagged, this is known as multiplexing. This proceeding is used several times along a protocol stack:



The client-server model

Almost all network communications follow a very simple dialog template known as the **client - server model**.

FIRST: the client sends a request to the server, usually following a user action. The client must know how to reach the server, hence, the server network node address and port number. The network address is usually provided by the user, possibly in the form of a hostname. The port number is fixed for each type of service.



SECOND: after receiving the request, the server processes it. Meanwhile the client is waiting for a response.



THIRD: after processing the request, the server responds to the client. To know the client's node address and port number, the server simply checks the request's source address.



These information exchanges follow the formats outlined in the application protocol. For certain services, a reply may not be required, for others this dialogue may be repeated several times.