The Document

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January 10, 2020

Abstract

This will be the document where I write everything I know about what I learn during my studies. I hope it serves me in the future as a "go to" document when I need to remind something.

Introduction

This document will focus on definitions, understanding concepts and transmitting intuitions/perspectives.

Each chapter will have the same structure. First the definitions and logic connections between everything with references to derivations and demonstrations that will be in attach at the end of the document. Second an overview of the derivations and how formula connect together to make a quick & easy way of finding the connections for hurry times. And third, simply one or two pages with the most important formulas from that chapter.

The first chapter will be about antennas. The second about microwaves. And the ones after that will consist of applications such as Microwave Links/ Hertzian Beams, Satellites and Radar. Maybe I'll join in a chapter about Probabilistic/Statistic Detection and Estimation of signals and one about some fundamentals of communications.

Please be aware that this is a document always at work. I'll write right here when I think a certain chapter is closed, otherwise it might be target of modifications.

Finally, there's a resource folder that I'll be refferencing frequently. It has many of my notes hand written, books, slides. Things I thought to be interesting or important. Probably all books mentioned in any section will be there.

This folder can be accessed through my Shared Drive Link

Enjoy

Contents

1 Telecommunication Networks - Transport Networks

From the courses Internet Networks and Services (RSI in portuguese) and Telecommunication Networks (RTel in pt) I've had an insight about how the whole network is multiplexed into optic fibers and many other interesting topics such as the triple play services and a bunch of protocols that are used in today's world to make everything communicate with everything. Therefore, I propose to write a sum up of the slides and bibliography of RSI and RTel in this section. I'll mainly give importance to RTel since it is what I'm studying at the moment, but I hope to go through the slides of RSI as well.

1. Introduction
   (2 lessons)
2. Fundamentals of networks
   (7 lessons)
3. Ethernet and data centre networks
   (5 lessons)
4. SDH transport networks
   (4 lessons)
5. Optical transport networks
   (4 lessons)
6. Access networks
   (3 lessons)

1. The Internet
2. Quality of Service on the Internet
3. IP Network Models
4. Next Generation Networks
5. The Telephony Network
6. Technologies for data transport
7. MPLS - Multi-Protocol Label Switching

1.1 Introduction

Definition *Telecommunications* : is the transmission of information at a distance through the use of electro-magnetic signals.

Definition *Telecom. Network* : collection of nodes and links with the purpose of interchanging these signals in order to have an information flow.

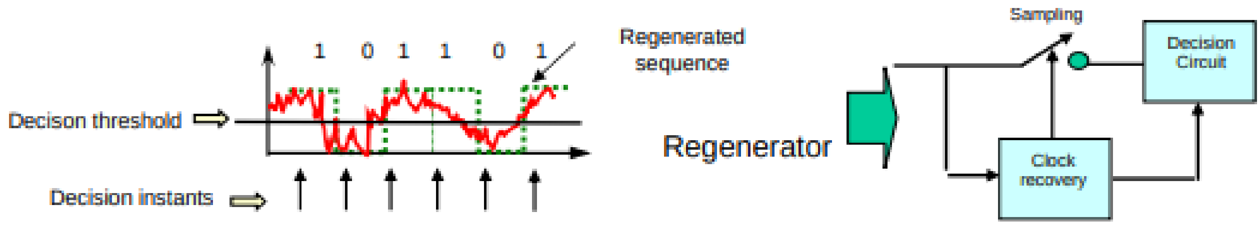
These Telecommunication Networks can be public, owned by Telecom. / Network Operators that use that network to provide services to the general public, or can be private, used by a company to connect infrastructures. Many of these private networks also rely on leased links by public networks.

There are mainly 3 layers in a network: the backbone or core, the metropolitan and the access layer. As expected, the access layer collects the traffic, connections to homes, offices, everywhere the internet is required. The metropolitan area connects different parts of the city that use that network, typically with a ring (made out of optic fiber). The core is the most extensive layer, with a mesh of nodes and very extensive links that connect cities of the whole world. Hundreds or thousands of km's is not atypical.

What makes possible to communicate with everyone connected to the internet is that the networks of both operators are also connected.

As a public service, the public networks must provide fidelity (transmit the information without loss of changes) and reliability (less than 3 minutes down per year).

Nowadays, most of transmission is digital. A series of pulses is transmitted through a channel with attenuation, dispersion, interference from other signals and noise. Therefore what reaches the other side is considerably different and has to be estimated what the original input was.



As having a dedicated physical infrastructure for each service would be far to expensive and messy, the big majority of services share the same channel, the optic fiber. It is easily shared because of the available bandwidth in it. Thus, the signals are multiplexed at the entrance, using different wavelengths (**Wavelength-Division Multiplexing (WDM)**) and de-multiplexed at the other end, to follow each one to their device that is requiring the service.

Note that WDM is exactly like **Frequency Division Multiplexing (FDM)** but in the optical domain. Technically they are exactly the same as changing the wavelength is nothing more than changing the transmit frequency.

A single mode optical fiber can reach throughputs of 10 Terabits per second.

Remember that there are many ways of scheduling frames in a multiplexer. With time slots or doing it statistically are two ways. Also, there are 2 types of switching: packet switching and circuit switching. Circuit is when a channel is constantly reserved for a certain application even if it is not being used. Packet switching allows a much better share of the resources. Packet switching principle is based on sending packets whenever there's a packet to transmit and use all the resources to do so as fast as possible. Therefore, the "speed" of the internet depends a lot on the amount of people that are accessing it.

Regarding the physical infrastructures for the transmission of data, those go from satellites, well the open space in general as microwave links are also a thing, twisted pairs, optical fibers and even a few more that are less common.

Finally, a look at the tendencies is pertinent. The traffic is increasing constantly, at a rate of 30% a year, therefore the network must be upgraded as the time passes as well, or else it won't be able to handle the traffic of the future. Not only are the links being upgraded since now we have fiber to the home, terminating really in our router, but each node must be upgraded as well to cope with the traffic resulting in new switches, larger datacentres, ect... However, all of this must be standardised to guarantee compatibility between countries, operators, manufacturers and users and to ensure minimum quality of service for all users. This standardisation is done by the International Telecommunication Union (ITU) that has two main sectors of interest: the -T sector regarding telecommunications in general and the -R sector for radiocommunications that is more focused in point-to-point, mobile, satellite links, ect... Additionally, ETSI, ISO, OSI, ANSI, IEEE are some of the main organisations that standardise technologies. IEEE is the best :)

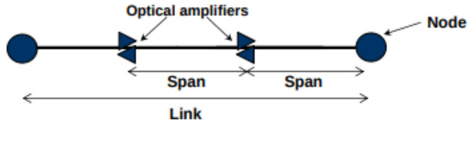
1.2 Networks Fundamentals

A network is composed of nodes and links and can be represented by graphs. However, a clear distinction between networks and graphs has been made in class: a network is a graph with a few more numbers that represent various network parameters. These parameters will be talked later, but can be delay of a link, distance, ...

The physical topology concerns the physical connections that are in place while the logical topology for a certain case concerns the actual flow of information. Even though every computer is connected in a network, maybe the information always flows from one to the others and the graph that represents that has much less links.

A link can be unidirectional or bidirectional. If the link is unidirectional, sometimes is referred to as an arc. $e_1 = (v_1, v_2)$ is the representation of a link, and the order of the nodes matter if it's an arc.

In optical fiber networks, or other networks that require amplifiers, the space between amplifiers (distance the signal has to go attenuating) is called a span.



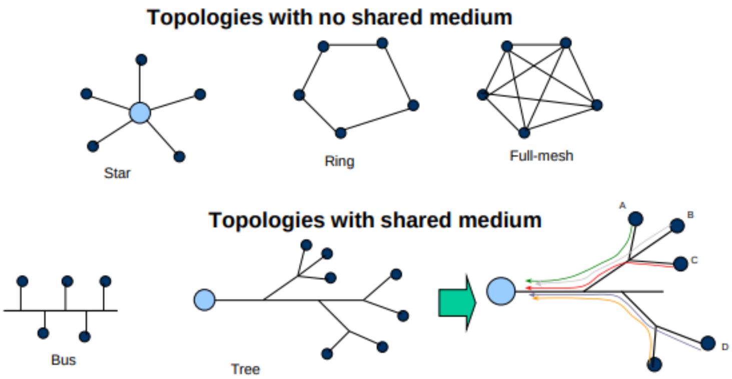
In a graph there's N number of Vertices(v_i), and L number of Edges(e_j). And the degree of the vertex is the number of edges it has. It's called the order of the graph, it's number of vertices, and the size of the graph it's number of edges.

Directed graphs only have unidirectional links, while undirected graphs only have bidirectional links.

The reason to make the distinction between directed and undirected graphs (unidirectional and bidirectional edges): in case of optical fiber, which is what connects most of long distance networking, is required to use more than one fiber. Because and optical emitter can't receive as well (at least in the same fiber). Also, if amplifiers are required, note that they are directed as well.

A path can be represented by a set of links, starting at some node. Source and sink are the names for the first and last vertex of that path.

1.2.1 Network Topologies



Bus, Ring and Star are the main physical topologies. Tree as well.

A tree is simply a graph with no cycles.

1.2.2 Network representative Matrices

A graph can be represented with an **Adjacency matrix (A)** , with $a_{ij} = 1$ if there's a direction from the vertices i to j.

The average node degree is given by the average of each node's degree which won't be more than the sum of all links, times 2 divided by the number of nodes. Times 2 because each link contributes for the degree twice, once at each end.

$$\delta_i = \sum_{j=1}^{N} a_{ij}$$

The average node degree is given by

$$<\delta> = \frac{1}{N}\sum_{i=1}^{N}\delta_i = \frac{2L}{N}$$

The Network diameter (D_R) is the maximum number of links between nodes through the shortest path between them. D_R is the longest of the shortest paths between every node.

Every link can have an associated cost (a function of distance, delay, reliability, actual cost, or other parameters) and a capacity (u_e denotes the capacity of node e).

The link capacity can be measured in any traffic unit that is appropriate for the problem. In packet networks: bit/s; in SDH networks: STM-N; in OTN networks: OTU-k; in WDM networks: number of wavelengths

Despite similar to an Adjacency Matrix, the **Demand matrix (D)** is slightly different. $d_{ij} = 1$ if the traffic flows from the vertex i to the vertex j.

Note that the diagonal of this matrix should be empty, or else it would mean that a certain node would receive information from himself, which makes no sense.

The mean number of demands is the number of demands divided by the number of nodes.

$$<d> = \frac{1}{N}\sum_{i=1}^{N}\sum_{j=1}^{N} d_{ij}$$

The number of unidirectional demands (edges) in a case of full mesh logical topology is $D_1 = N(N-1)$. N nodes x the other N-1 nodes. Note that One other way of seeing it is that the D matrix is $N \times N$ but we need to take N away due to the empty diagonal. $D_2 = \frac{D1}{2}$ is the number of bidirectional demands, which is when only the top triangle of the D matrix is considered. This is usual because with bidirectional links the D matrix will always be symmetric.

Another interesting matrix is the **Traffic matrix (T)** and it's used to denote traffic intensities. It only has entries different from 0 in the exact same places the Demand matrix has. It's used for static traffic designs.

In transport networks the traffic units is the type of client signals: Ex: E3 (34 Mb/s), STM-1 (155.51 Mb/s), GbE (1 Gb/s), 10 GbE (10 Gb/s) ,etc. This traffic units must be converted to traffic units appropriate to be used in network design. These traffic units must be VC-n in SDH networks and in OTN networks ODU-k signals.

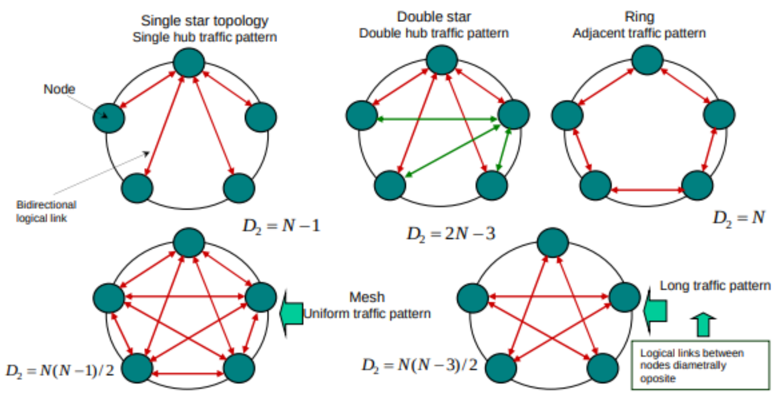
Considering now a Dynamic Traffic case, a few concepts arise:

- Average intensity of data flow between two nodes;
- Traffic bursts are time intervals where the flux of data is considerably higher than the average rate;
- Peak rate is the maximum instantaneous intensity.

Aggregation level or multiplexing level reduces the traffic *burstiness* because there less often flows get fully used and is more likely that the information can flow at a constant pace instead of in bursts.

Note one important distinction between Logical and Physical Topologies that we haven't done yet is that there can be many logical topologies on one physical topology. This can happen in the following way:

- **In a ring network there are diferent ways how the traffic can flow in a network leading to different network topologies (single star, double star, ring, mesh, etc.). *N*: number of nodes; D_2: number of bidirectional (two way) logical links.**



Routing is how a packet travels in a network. Therefore, routing ends up being the map between logical and physical topologies.

In optical networks the path between two nodes is usually called "lightpath".

We can also define a **Cost Matrix (C)** where each element c_{ij} represents the costs between nodes i and j.

The path can be performed manually (static routing - Demand matrix is time invariable) or dynamically, through routing algorithms (dynamic routing - Traffic matrix is time dependent, with constant arrival and termination of new demands).

Additionally, if the a given traffic demand(connection) is able to use more than one route, it is called a multipath routing process, else it is a mono-path process. Because there are usually many paths connecting two nodes, some metrics are taken into account when choosing which to follow:

- 1) Minimizing the network cost;
- 2) Minimizing the traffic in the most loaded link;
- 3) Minimizing the number of hops (number of links in the path);
- 4) Minimizing the path distance;
- 5) Maximizing the protection capacity, etc.

Most of the routing strategies incorporate some sort of shortest path algorithm to determine which path minimizes a particular metric, which can be for example 1), or 3), or 4). Note that the same algorithm used to 4) can be applied to 3) making all link distances identical.

From the physical topology, described by a graph G(V,E) and the traffic matrix T, describing all the traffic demands to be routed, one can perform shortest path algorithms such as Dijkstra's algorithm.

Order the demands according to a certain sorting strategy:

- Shortest-first: The demands with the lowest number of nodes in its path come first in the list;
- Longest- first: The demands with the highest number of nodes in its path come first in the list;
- Largest-first: The demands with the highest number of traffic units come first in the list;
- Random ordering: the demands are not known initially.

Route de demands according to the orderings. To break a tie choose the path that minimizes the load in the most loaded link.

Dijkstra's Algorithm

Consider a generic node *i* in a network with *N* nodes from where one wants to determine the shortest path to all the other nodes in the network. Lets l_{ij} be length (cost) of the link between node *i* and node *j* and d_{ij} the length of the shortest path between node *i* and *j*.

Algorithm:

1) Start with the source node i in the permanent list of nodes, i.e. $S=\{i\}$; all other nodes are put in the tentative list labeled S'. Set $d_{ii}=0$ and $d_{ij}=\infty \quad \forall j \neq i$.
2) For all the adjacent nodes to i set $d_{ij} \leftarrow l_{ij} \quad \forall j$ adjacent to i.
3) Identify the adjacent node j (not in the current list S) with the minimum value of d_{ij} (permanent node), add it to the list S ($S=S\cup\{j\}$) and remove it from ($S'=S'\setminus\{j\}$). If S' is empty stop.
4) Consider the list of neighboring nodes of the intermediate node j (but not consider nodes already in S) to check for improvement in the minimum distance path by setting $d_{ik} \leftarrow \min(d_{ik}, d_{ij}+l_{ij})$. Go to Step 3.

Now is pertinent to introduce yet another matrix, the **Hop Matrix (H)** where each element h_{ij} denotes the minimum number of hops from node i to node j.

The average number of hops per demand is nothing more than the sum of the hops of all demands divided by the number of demands. The number of demands was set as the unidirectional links between two nodes. Therefore, if 2 nodes share information between themselves (don't need to have a physical link, a logical one is enough) then there's a demand.

Therefore, the average number of hops can be computed:

$$<h>=\frac{1}{D}\sum_{i=1}^{N-1}\sum_{j=i+1}^{N}h_{ij}$$

Note that coherence is key here. If the amount of demands are the bidirectional demands, then the number of hops considered should only be the top half of the hop matrix. **If links are bidirectional, then there will always be a symmetry in these matrices** and we should compute the average with amount that mean the same thing.

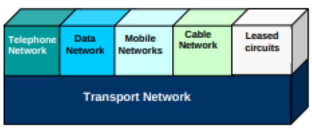
As means of simplifying this calculation, because hops and demands can be dynamic, a way of having a notion on the order of magnitude and get a fairly good approximation, when the number of nodes N is $4 \leq N \leq 100$ and the average node degree $<\delta>$ is $2.5 \leq <\delta> \leq 5$, is by computing the semi-empirical relation:

$$<h> \cong 1.12\sqrt{\frac{N}{<\delta>}}$$

1.2.3 Layers

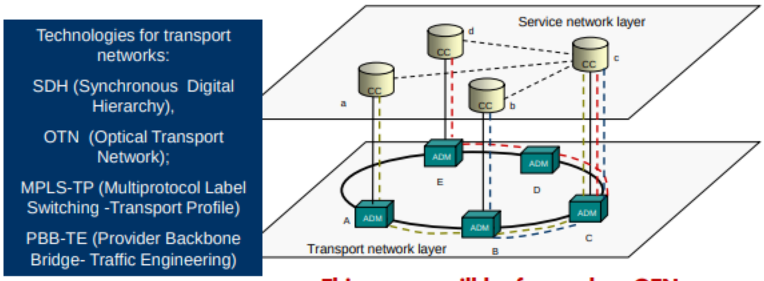
Typically, there's a layered structure in the network. The layer above acts as a client of the layer beneath and each layer appears a black box that supplies a service to the layer above.

The service layer is the one closer to us.



Add/Drop Multiplexing(ADM) are multiplexers controlled by **Control Centres (CC)** that decide what to add and what to drop form the fibre. Note that these don't manage the network, they just use it.

Nowadays, apart from local networks, everything is connected with fibre. Therefore, is pertinent to mention 4 key technologies for transport networks:



This course will be focused on OTN

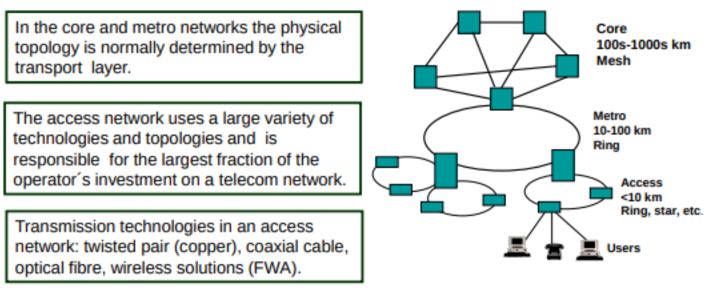
SDH is also used in Hertzian links, MPLS is Multiprotocol Label Switching and is very useful for routing through different technologies and to choose more carefully the paths. As said above, OTN will be our transport technology.

OTN has become the new standard for a while now and has a few differences compared with SDH. The most important of which is distinction between fixed frame size and fixed rates. SDH has a fixed rate while OTN can increase it's rate to match the client's and this is very important for scalability and being more future proof

A very important distinction between the transport and the service layers is that the representation in the service layer has nodes connected with **logical topologies** while the transport layer has nodes connected with **physical topologies** .

OTN	SONET/SDH
Asynchronous mapping of payloads	Synchronous mapping of payloads
Timing distribution NOT required	Requires right timing distribution across networks
Designed to operate on multiple wavelengths (DWDM)	Designed to operate on multiple wavelengths
Scales to 100Gb/s (and beyond)	Scales to a maximum of 40Gb/s
Performs single-stage multiplexing	Performs multi-stage multiplexing
Uses a fixed frame size and increases frame rate to match client rates	Uses a fixed frame rate for a given line rate and increases frame size (or uses concatenation of multiple frames) as client size increases
FEC sized for error correction to correct 16 blocks per frame	Not applicable (no standardized FEC)

The network management systems sends configurations through the **Data Communication Channel (DCC)** . Moreover, not all parts of the network are the same!



One important distinction in terms of how things are connected physically is evident comparing the access networks with the other parts of the network. Access Networks, as opposed to what happens to the rest of the network, uses twisted pair or one optic fiber. This is because of the cost versus the bandwidth a fiber offers.

One fiber can carry several times the traffic of one user, therefore can be used for multiple users. Moreover, very often the final leg is done with twisted pair due to cost reasons.

In a more abstracted way, one can identify three planes:

- **Data Plane : is concerned with the transmission of the data between the users (forwards the traffic). Assures the physical support.**
  Also called forward or switching plane
- **Control Plane: is responsible for exchanging control information (signalling) between networks elements (nodes), which is used to set up, maintain, and tear-down connections. Examples of control planes: Signalling system nº 7, GMPLS (*Generalized multiprotocol label switching*), SDN (Software-Defined Networks) etc.**
- **Management Plane : Consists of several functions like detecting (alarms) and repairing failures (fault management), network element configuration (configuration management), performance monitoring to ensure to clients quality-of-service (performance management), allowing the system administrator to control personal access (security management).**

As you already know, there are circuit switched or packet switched networks.

An important note: Packet switching can be connection oriented (MPLS) or simply connectionless (pure IP packets, best effort)

Circuit Switched require circuit establishment and tear-down at the beginning and at the end, respectively. However, a distinction is made between physically *switchable* circuits and semi-permanent circuits. The first ones, a physical circuit is easily switched to connect one end to the other, while the second type regards circuits that are more static, that are much more difficult to switch. The semi-permanent must be switched by the administrators in order to attribute these circuits to a user for a somewhat long period of time (not just one transmission).

Circuits can be switched or semi-permanent. The first ones are established by the control plane (by signalling) as the case of phone circuits. The second ones are established by the management plane as it is the case of the electrical paths in SDH networks, or optical channels in OTN networks.

The biggest difference is the amount of time each one uses de circuit for. Switched circuits is around minutes, Semi-permanent are months.

Then, of course, there are packet switched networks that allow a much better share and efficient use of resources.

The key notion to have is that all services use the transport network! This is the highway of data! Nowadays it is impossible to have dedicated physical connections for each service. And this section of *Networks - Transport Networks* is based on that!

Telephone Networks

Telephone Networks before used circuit-switching. Note that the only ones that use circuit-switching are the landlines! Nowadays, our mobile communications are able to replace this fixed circuits lines

Local exchanges (Access) - are small switching centres that serve a small area.

Transit exchanges(metro, core) occur in **Primary trunk exchanges** is used to transfer the traffic and to interconnect several circuits.

They usually have a physical connection between them, but not one of their own. It wouldn't be viable to have dedicated circuits. It's here that the transport networks comes.

Different components have different topologies. The network is a hierarchy, having different topologies in different parts of the network.

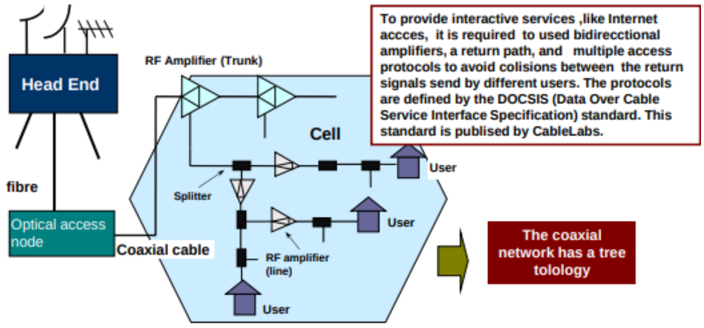
Digital Circuit-switching meaning that the data being switched is of digital nature.

Hybrid Fibre-Cable Network

There's an head end that distributes all the channels for all the users.

The network is called hybrid because it uses both fibre and coaxial cable.

Excellent example of a tree network. Like all the networks with share mediums, it required multiple access protocols to allow everyone to use the medium (FDMA and TDMA are 2 of the multiple access technologies).



The transport network is represented by the optical fibre that connects the head head to the optical access node. Like the cellular communications, one optical access node covers a certain area, with coaxial cable.

IP Networks

There are 2 ways of sending IP packets:

- Without Connection, it is necessary to have a buffer to reassemble all the IP packets. The packets are simply send to the network in a best effort way.

- With Connection: when certain QoS is required, then MPLS is used, so that certain resources can be reserved and the QoS met. MPLS establishes a virtual circuit before starting transmitting the packets. MPLS acts between layer 2 and layer 3. It places a label between the layer 2 and layer 3 labels.

There are 2 types of MPLS routers: - LER - Label Edge Router are the ones that put the labels and take them out; - LSR - Label Switching Router simply take labels out.

MPLS works by pre-establishing a path for information flow. Then, at each hop, according to the label id's available in each LSR the label arrives with one number and it leaves with other, based on the MPLS routing table in each MPLS capable router. This table entries are created based on the MPLS protocol and pre-path find procedures.

This is how a MPLS table looks like:

	Input port	Input label	Ouput port	Output label
FEC 1	1	15	2	10
FEC 2	2	25	3	20

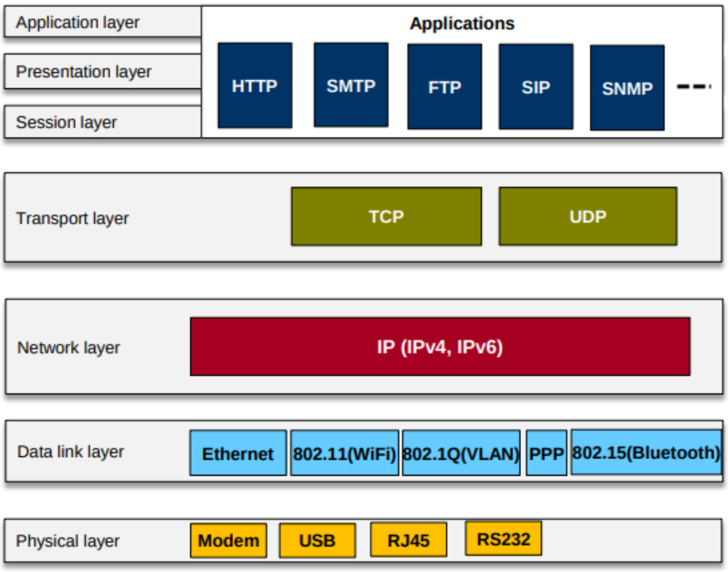
Generalized MPLS (GMPLS) is the *de facto* (practice that exists practically even though it is not formalized by law) of the control plane of the Wavelength Switched Optical Network (WSON).

Label Switching allows Traffic Engineering:

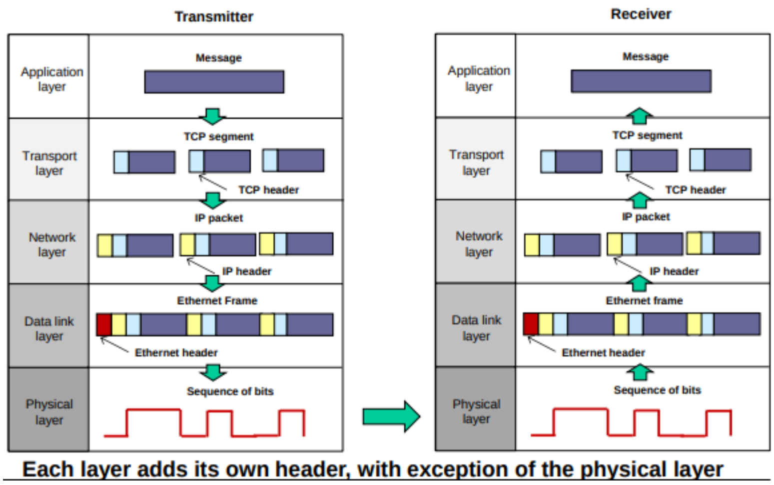
Traffic Engineering (TE) deals with a set of procedures required to optimize the performance of telecommunications networks and use network resources in an efficient way. It permits, for example, to route the traffic in order to avoid congestion and to maximize the transported traffic.

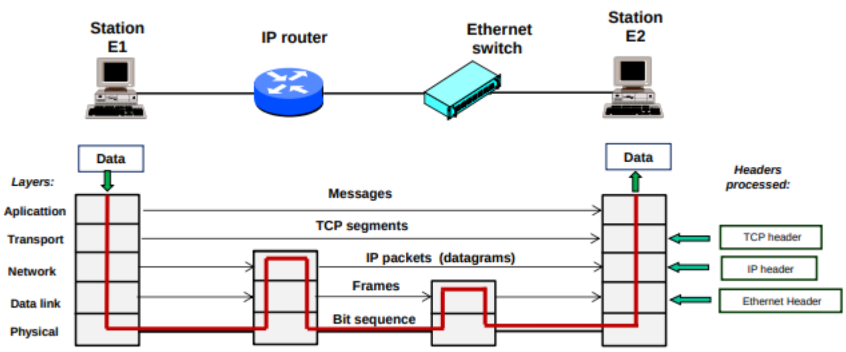
1.2.4 Layered Model Overview

Open Systems Interconnection Model:



Transport layer connects applications. Network layer connects machines with different IPs. Data link layer connects two machines with their interfaces' MAC addresses. Physical layer is what is responsible for putting the data into a cable or into the air to perform the actual transmission. For instances, spectrum, modulation and coding techniques and intervals of transmission.





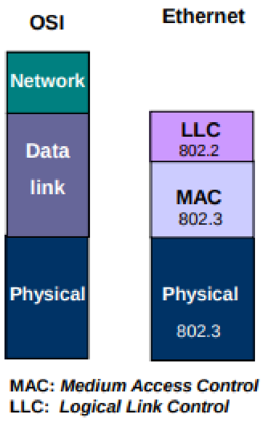
1.3 Ethernet Networks

Most of today's network traffic is generated from Ethernet interfaces. Made from twisted copper pair, through slightly more advance techniques and protocols is still possible to get quite a good throughput with a low cost cable.

Uses CSMA/CD Carrier-sense multiple access with collision detection because it can sense if there's anyone using the bus at all times. WiFi for instances CSMA/CA is used since we may be causing a collision we can't detect.

The Ethernet protocol 802.3 is used in all parts of the transport layer, not only at the access level and specially not only in LAN (Local Area Networks). Note that the Ethernet protocol was standardised by IEEE and the standard specifies not only the physical specifications but also Data Link Layer frame formats. WiFi 802.11 is another protocol that includes physical, like transmit powers and modulations, and Data Link specifications

A rule of thumb is to use optical cables for distances above 100 metres.

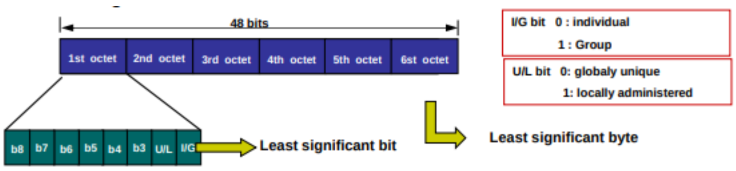


- The LLC (Logical Link Control) sub-layer is responsible for the flow and error control between the nodes.
- The MAC sub-layer is responsible for the media access control, addressing, error detection, frame delimitation, by organizing the bit sequences into frames.
- The physical layer deals with the bit transmission and reception, with the electrical, optical and mechanical properties of the interfaces, with the type of connectors used, etc.

The Data Link layer is responsible for frame processing, and error detection through the calculation of a CRC (Cyclic Redundancy Check). If a frame has errors, it is discarded. Most of error correction is done higher up in the stack. TCP guarantees error free transmissions when it says they were successful.

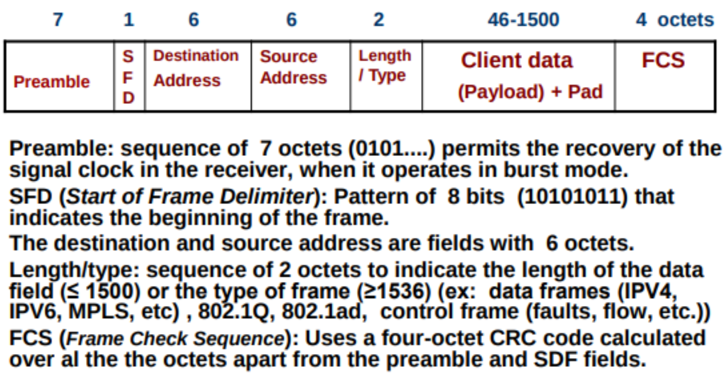
MAC stands for Media Access Control. Thus, the MAC addresses are addresses to access the physical media. CSMA protocols are placed in this layer.

There are 2 bits in the MAC address, the two least significant ones form the first octet.



U/L refers to the uniqueness of the interface. The MAC address can be changed becoming a local one. I/G refers to one interface only or a group that has that MAC address that received always the same thing.

Write *ifconfig* or *ip a* in linux (or *ipconfig* in windows) is possible to see this MAC address.



7	1	6	6	2	46-1500	4 octets
Preamble	SFD	Destination Address	Source Address	Length / Type	Client data (Payload) + Pad	FCS

Preamble: sequence of 7 octets (0101....) permits the recovery of the signal clock in the receiver, when it operates in burst mode.
SFD (*Start of Frame Delimiter*): Pattern of 8 bits (10101011) that indicates the beginning of the frame.
The destination and source address are fields with 6 octets.
Length/type: sequence of 2 octets to indicate the length of the data field (≤ 1500) or the type of frame (≥1536) (ex: data frames (IPV4, IPV6, MPLS, etc) , 802.1Q, 802.1ad, control frame (faults, flow, etc.))
FCS (*Frame Check Sequence*): Uses a four-octet CRC code calculated over al the the octets apart from the preamble and SDF fields.

Because there are only 1500 bytes to address (to count) with the Length/Type field, then there will be quite a few empty bits in this field. The rest of the bits are used to denote the type of the frame:

If the value in the field lies from 0 to 1500 it indicates the length of the Date field. If the value ranges from 1536 to 65 535 it indicates the nature of the Data field. Examples of field types:

0x0800: IPv4; 0x8600: IPv6; 0x8808: Ethernet flow control; 0x8847: MPLS unicast; 0x8848: MPLS multicast; 0x8100: IEEE 802.1Q; 0x88A8: IEEE 88A8.

The purpose of the Preamble is to achieve clock synchrony.

1.3.1 Multiple Access

TDMA, FDMA (WDMA), CDMA all have the same principle: divide signals in a given domain.

TDMA divides signals in the time domain: different transmissions occupy different time slots.

FDMA divides in frequencies, WDMA divides in wavelengths. The only difference is that one is more used in wireless applications and the other with fibres.

OFDMA starts combine the previous ideas: divides in the time and in the frequency domains, therefore is able to attribute resource blocks with much better efficiency. A use of OFDMA is in 4G and 5G where a physical resource block is nothing more than a set of sub-carriers used for a given time period, usually called one TTI (Transmission Time Interval).

CDMA is used in 3G and with GPS, so that different satellite can transmit all at the same time.

Then there's another category of multiple access techniques like CSMA. Namely CD and CA, collision detection and collision avoidance.

Collision Detection is when a collision is possible to be detected and after being detected certain steps are taken to minimize the risk of happening again.

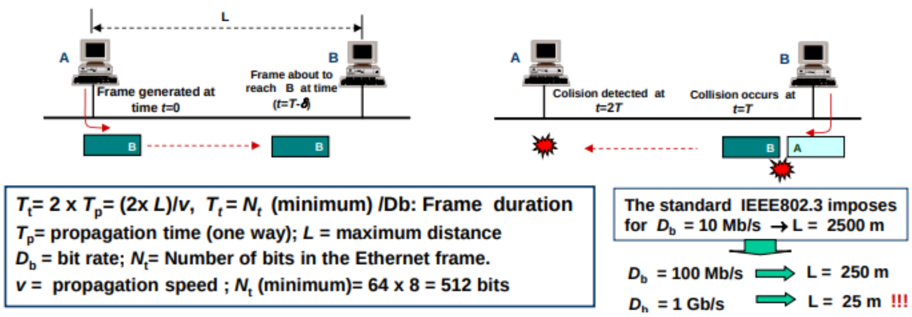
Collision Avoidance is when the collision can't be detected and should at all costs be avoided.

A curiosity: he speed of light inside an optical fibre, considering an average glass refractive index of 1.5, is:

$$v = \frac{c}{n} = 2 \times 10^8 m/s$$

In cabled Ethernet, CSMA/CD is used. Disregarding all other phenomena that limit the debit - dispersion that leads to distortion, attenuation, etc...- the use of CSMA/CD is limited by the distance that the information can travel before one discovers that a collisions as occurred. In essence, if the maximum debit is 10 Mbit/s and the maximum ethernet frame is around 1500 bytes long, it will take $\frac{1500*8bits}{10\times10^6 bits/s} = 0.0012s$.

The electrical signals propagate at speed of light - don't confuse with the speed of light in the fibre, in the fiber the light is propagating in glass, here there's a wave propagating in TEM mode along a twisted pair or coaxial cable. Therefore, in order to notice a collision, the signal must be able to go all the way to the other end and comeback BEFORE the sender is ready to send another frame. Mathematically: $T_{max-frame} \geq 2T_{propagation}$. This is because, if there is a collision at the other end, the sending machine must receive that collision before finishing the transmission, otherwise it would keep sending frames without knowing about the collision.



Since $T_{propagation}$ is proportional to the length of the connection, and the time of transmission is inversely proportional to the bit rate, then the bigger the rate, the smaller the maximum distance between machines can be.

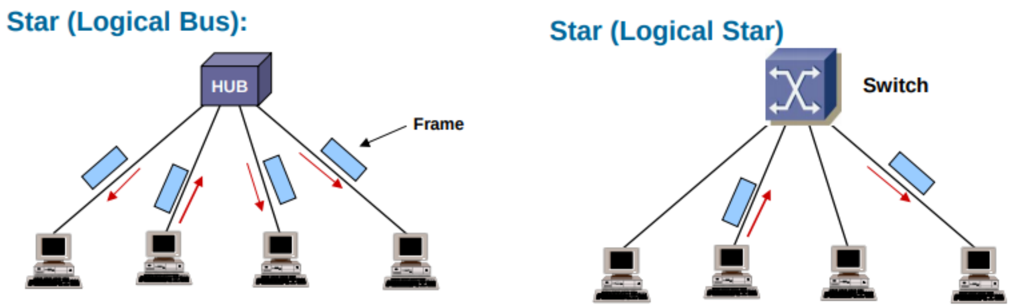
That is also why CSMA/CD is not used when the rates get too high!

Type	Bit Rate	Mode	Topology	CSMA/CD	Medium
Ethernet	10 Mb/s	Half-duplex	Bus	Yes	Coaxial cable
Fast-Ethernet	100 Mb/s	Half e full duplex	Star	Yes	Copper + Fiber
Gigabit-Ethernet	1 Gb/s	Half e full duplex	Star	Yes	Copper+ Fibre
10 Gigabit Ethernet	10 Gb/s	Full duplex	Star	No	Copper+ Fibre
100 Gigabit Ethernet	100 Gbit/s	Full duplex	Star	No	Fibre

Half-duplex → CSMA/CD (Carrier Sense Multiple Access/Collision Detection)
Full-duplex → Switched Ethernet

Note that this is only necessary if a medium is shared! If it is dedicated, it is not necessary.

A switch is a layer 2 - the Link Layer - equipment that commutes Ethernet frames.



A Hub only has physical layer, propagates to all interfaces everything. A Switch has layer 2 and Mac Address Tables. A Router has layer 3 capabilities and is able to route IP packets.

Switches with higher capacities are higher up in the grid, closer to the core, because they have to handle many more traffic flows.

Some functions of switches:

- Forwarding - Simply check the destination, it knows to which port that destination is connected to and send the frame in that port destination is
- Broadcasting - Sending to all destinations, this happens when it doesn't know where to send it and needs to find out.
- Filtering -

In order to avoid problems such as exponential spreading of frames, a **Spanning Tree Protocol** at a logical level is used. Closed loops go away with this protocol since a tree is created and all notes still are connected since it is a spanning tree. BPDU - Bridge Protocol Data Units are the frames sent to establishing this tree.

Ethernet frames don't have time to live. But in BPDU's have TTL

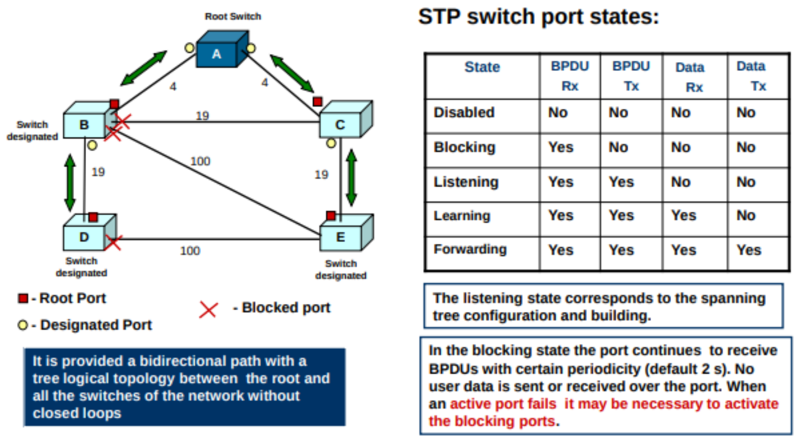
This Spanning Tree Protocol can be implemented in the following way:

1. Root Switch election - typically the one that has the smallest MAC in the network because computing the one that has the smallest distance to every other node is too troublesome. Therefore, after receiving a BPDU with an ID lower than the current root, it replaces his belief of root and propagates that information.
2. Convergence to Spanning tree starting at the Root - The root propagates BPDU's with a cost to root (if the root is sending, the cost is 0). The nodes that receive do the same: **(root id, my id, cost to root)** . - If the received packet tells a cost to root that is smaller than before, that port is designated as the root port.

   - If the received packet tells a cost that is bigger, than that port will be blocked: not packets should be sent that way!

   - The ports that are not blocked and that will connect the lower nodes to the root will be called **designated ports** . These are the ports that node can send things to.

This is an example of the status of each



How is the switch able to build the Source Address Table (SAT)

It can be filled by hand, by the network administrator. Or it can learn the MAC that communication to each port.

How packets Travel in the network We write a message in the phone or in the laptop and press send in the application (layer). First, there is some compression and encryption done by the application layer right away. Then in the application code, there is a section that is defining TCP or UDP sockets - very likely TCP -

How to build minimum spanning trees? They are needed for the spanning tree algorithms like the Link Layer LAN minimum spanning tree algorithm.

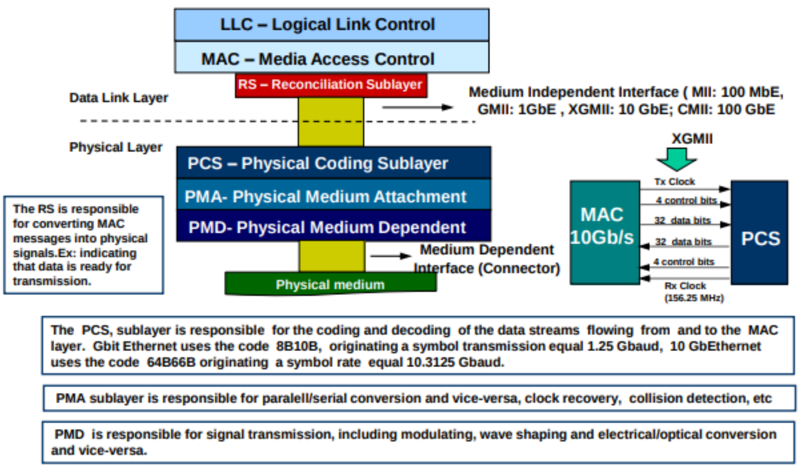
With the Kruskal's Algorithm:

1. order the edges by their cost, minimum first.
2. keep adding edges to a the spanning tree, as long as the edge doesn't create a loop (connects two nodes already connected in another way).
3. stop when the tree has **n - 1** edges.

1.3.2 Physical Layer of the Ethernet

The Ethernet protocol includes physical specifications.

Between the link layer and the physical layer (the layer right before the physical medium) there's a **medium independent interface (MII)** . In the picture below, depending on the debits being transmitted, this interface has several names. It has different names because it is a parallel interface that depends on the debit.



The first sublayer of the physical layer is coding. It will had signals to the bits. Each bit will

The second sublayer is responsible for the parallel/series conversion.

The last sublayer must generate the signal into the medium being optic fibre, twister pair, etc...

In essence:

10GMII (10-*Gigabit Media Independent Interface*) : standardized interface between the MAC layer and the physical layer. Permits that the MAC layer can work with different implementations of the physical layer.

PCS (*Physical Coding Sublayer*): It is responsive for coding/decoding the information originated/destined to the MAC layer. There are many types of codes that can be implemented : 64B66B, 8B/10B, PAM-5, etc. O PAM-5 is a multilevel code used in the 1000-Base T standard.

PMA (*Physical Medium Attachment*): It is responsible for the serial/parallel conversion and vice-versa. The clock synchronization is also carried out by this sub-layer

PMD (*Physical Medium Dependent*): It is responsible for the signal transmission. There are different PMD devices according with the medium.

MDI (*Medium Dependent Interface*): Indicates the type of connector used.

Code - mBnB

Firstly, the code. Required for robustness of transmission and to send symbols at lower frequencies than the actual bitrate.

m input, n output bits

By switching between Mode 1 and Mode 2, the number of 0's is identical to the number of 1's, which is important to keep the stochastic balance between 1's and 0's - important for detection.

3B4B Code

Input bits	Mode 1	Mode 2
000	0010	1101
001	0011	0011
010	0101	0101
011	0110	0110
100	1001	1001
101	1010	1010
110	1100	1100
111	1011	0100

After the coding, a 1Gbit/s bit rate is converted into a $\frac{n}{m} * Rate$. For 3B4B, it will be necessary to transmit 4 bits in the time it took to transmit 3. Before, the rate per bit would be $\frac{1Gbit/s}{3}$. And this needs to be multiplied by 4 to have the 4 bits going at the same time. Therefore, 1.(3) Gbaud (symbols per second) is the new rate.

The ethernet transmission is done in baseband, no modulation.

Format used in the interfaces: [Value1] Base [Value 2]

Value1 = Bit rate for the transmission : Ex: 100 $\Longrightarrow$ 100 Mb/s

Base = *Baseband Mode*: Base band transmission (no modulation)

Value 2: Type of cable (T: *Twisted Pair*, F, X, R: Optical Fibre)

Examples:

10 Mbit/s Ethernet

10BaseT: Uses UTP twisted pairs of category 3 ou 5; max distance = 100 m
10Base F: Uses multimodal optical fibre

100 Mbit/s Ethernet (PCS: 4B5B)

100BaseT: Uses UTP twisted pairs of category 5; max distance = 100 m
100Base FX: Uses multimode optical fibre (62.5 μm); max distance = 2000 m

Gigabit Ethernet (PCS: 8B10B)

1000BaseT: Uses UTP twisted pairs cat. 5e; max distance = 100 m
1000Base SX: Uses multimode optical fibre (62.5 μm); max distance = 275 m
1000Base Fx; Uses single mode optical fibre; max distance = 5000 m

For the correct names Ethernet phy layer .

Avoiding Crosstalk

Electromagnetic interference becomes a limiting problem in very high frequencies. To reduce these interferences, a metal sheet foil can be used. A more effective solution is really to use a shield Therefore, shielding is necessary for twisted pairs.

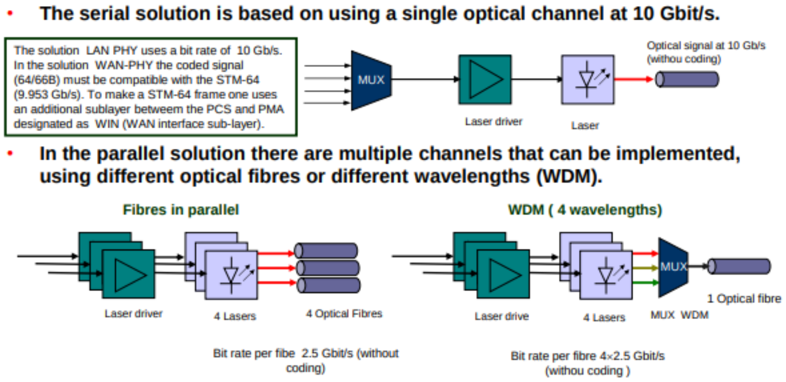
The necessary bandwidth to transmit 10Gbaud/s.

1 baud can have several bits of significance, therefore there is a strong leverage in the coding that is used.

COMPLEEEEEEEEEEEEEEEEEEEEEEEEEEEEEEEEEETE THIIIIIIIIIIIIIIIIIS

Category 7 (CAT7) : Bandwidth equal to 600 MHz((@ 100m). Supports bit rates up to 10Gbit/s (10 GbE), using a cable with 4 STP pairs.

In order to multiplex everything into the physical medium, there are two ways of doing it, normally the second one is used.



The different interfaces for 10 GbE are the following:

Interface	Medium	PCS	Distance	Source
10GBase-SR	Multimode fibre (1 pair)	64B/66B	100-300 m	VCSEL, ou FP laser (850 nm)
10GBase-LR	Monomode fibre 1 pair)	64B/66B	10 km	VCSEL, FP (1310 nm)
10GBase-ER	Monomode fibre l (1 pair)	64B/66B	40 km	Laser DFB (1510 nm)
10GBase-CX4	UTP twisted pair CAT 5 (4 pairs)	8B/10B (PAM)	15 m	Standard 10GBASE-X uses the code 8B/10B
10GBase-T	UTP CAT 6A or STP CAT 7 (4 pairs)	64B/66B (10-PAM)	100 m	4 pairs × 833Mbaud × 3 bits/Baud = 10Gb/s

Gigabit + Ethernet

When the speeds are too big, either the distances need to be smaller or fibres must be used. Single mode avoids dispersion and the pulse is able to keep is form for longer.

1.3.3 Virtual LAN

The time it would take to implement the Spanning Tree Protocol in very big networks makes it infeasible. Therefore, the same physical grid should be separated in many logical grids and the spanning tree only covers one VLAN.

VLANs also allows for traffic separation in triple play services, allow for cloud computing and storage separation. Is possible to separate departments, buildings, etc...

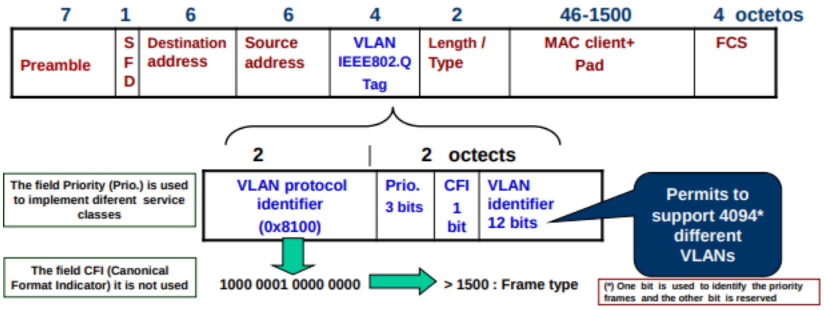
Additionally, it improves the network security and privacy since the traffic inside a VLAN doesn't need to come outside. And facilitates network management since the administrator is able to organize user groups independently of the physical topology of the network. It is the network administrator that defines the limits/boundaries of the VLAN.

The communication between VLANs is done at Layer 3.

There is 3 types of VLANs implementations:

- Layer 1 VLAN (port-based VLAN): the frames don't need to be tagged and the switch distinguish VLANs based on the ports it come from.
- Layer 2 VLAN (MAC Address-based VLAN): the frames need to have VLAN specific tag.
- Layer 3 VLAN (Network Address-based VLAN): The information of VLAN includes IP addresses, requires routers.

Here is the tag:



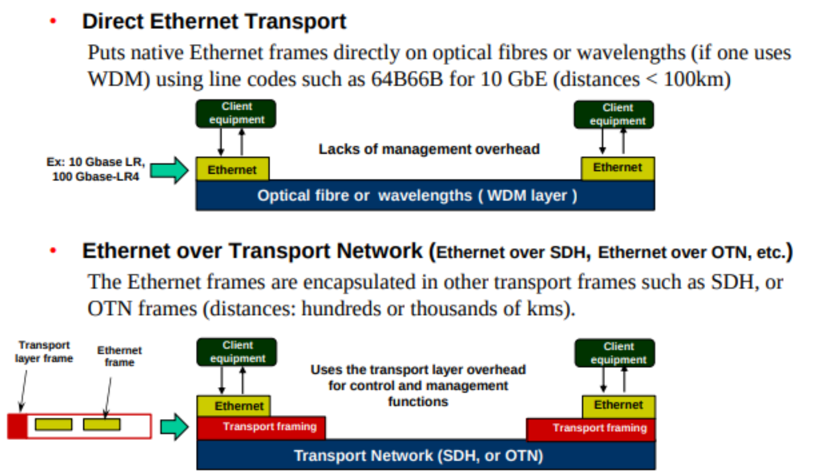
Note the first field of the tag is bigger than 1500. This happens because the switch is expecting to see a Length/Type field. This way, the switch reads it as a type that is a VLAN type and it knows how to treat the frame - that can be a BPDU or any other kind of frame - based on that identifier.

There is 2 octects to identify the VLANs (4094 because 2 are reserved). At some point, it isn't enough. What is done is multiple tagging: **simply by adding another tag, a VLAN inside a VLAN is created** . However, VLAN IEEE802.1Q is not very clear on how to treat such encapsulations, thus IEEE802.1ad Provider Bridges is used. The tag is called a Service Provider Tag and requires a Provider Bridges aware switch.

A switch can be aware to simple untagged frames, can be aware of VLANs and can be aware of Provider Bridges.

However, with these 2 tags, tables with 16 Million entrances is already too big... **There must be another way of scale the network properly and way of connecting PB switches at long distances.**

Solution 1- Ethernet doesn't have regenerators, therefore the ethernet frame can be place inside a transport network frame. Using SDH or OTN there's regeneration and there can be links with more than 40 km (the maximum length without regeneration).



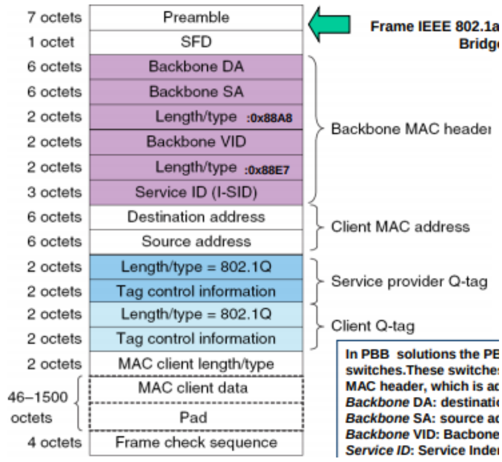
Carrier Ethernet is the idea of using Ethernet in all of the network.

PB switches are barely capable of handling Metro, can't be used in the core. PBB switches are the ones used in the core and they only process their own headers.

However, this is quite far from Ethernet Networks and requires many more functions to compete with the transport network

The standard of the transport network is recovering links after a failure in less than 50 ms. OAM functions and much more is needed to be implemented in Ethernet protocols in order to provide reliable traffic.

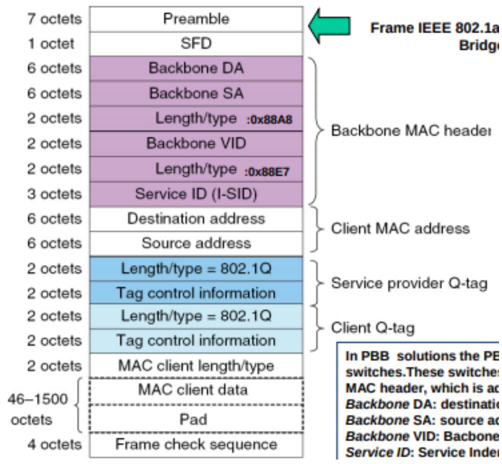
But, there's a huge Ethernet frame...

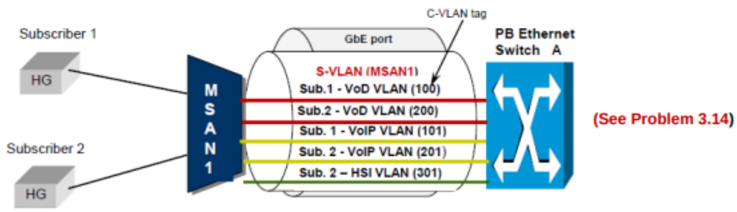


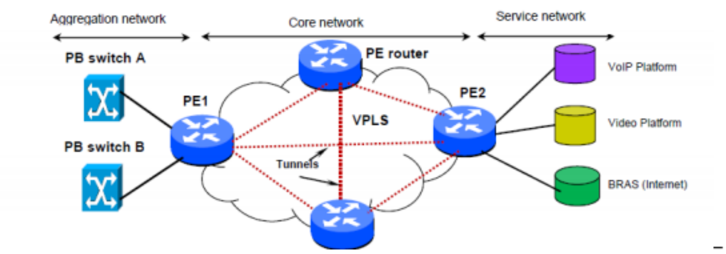
And they then it still uses spanning tree protocols? Non deterministic networks in the core? This would lead to big delays every time there's a change in a link

It is necessary to establish hard links through the management plane.

PBB-TE is a variation of PBB that doesn't use the spanning tree protocol.







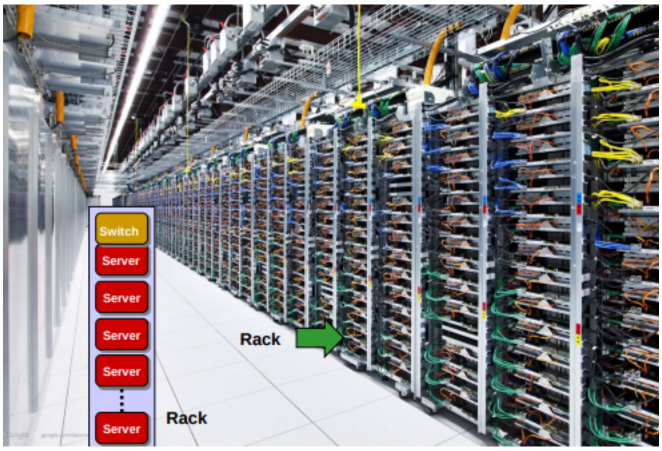
PBB-TE is a technology that can be used in the transport network. It uses only Ethernet but has a lot of overhead to

1.3.4 Data Centres

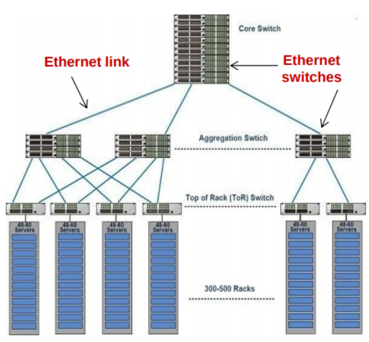
Big Ethernet Networks where an organization stores, manages and disseminates data. 10 and 100 GbE is used. VLAN aware, PB and PBB are possible to be found in Data Centres. Then there are Servers and Routers (IP and MPLS).

The connection between datacentres is called inter-datacentre connection and sometimes the data-rate needs to be so big that

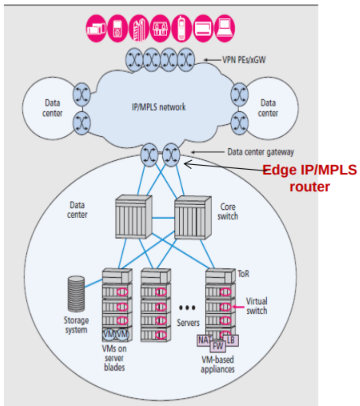
There a switch on top of each rack to connect all of the servers.



Top of Rack (ToR) switch, are connected to aggregation switches connected to just a couple of central switches and these are connected to Edge IP/MPLS routers that connect the IP/MPLS network that is on top of the global transport network.



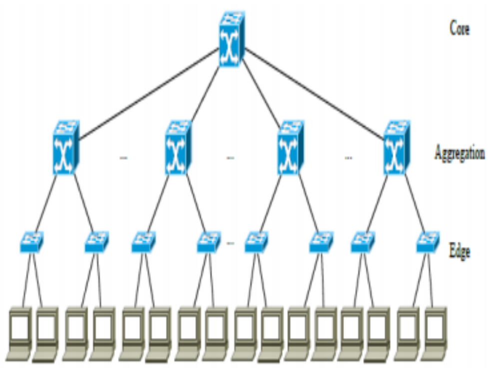
Source: https://www.multi-phy.com/solutions/data-center-intraconnect-solution/



Source:Technologies and Protocols for Data Center and Cloud Networking

For instances, to use a cloud based service a virtual machine must be created in one of those servers.

A tree physical topology despite being easy to implement provides no fault tolerance, leads to congestion (no load balancing) and doesn't scale properly since the maximum data rate the central node can handle will limit the size of the tree.



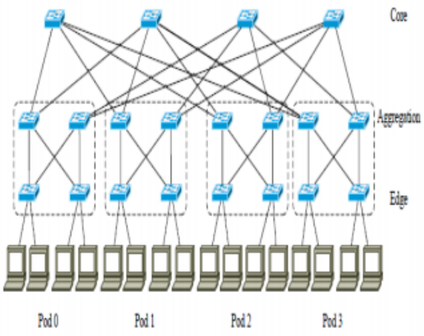
Instead, a Fat Tree is used.

Identical bandwith at any bisection

Each layer has the same agregated bandwith

Advantages:
- **Redundancy**

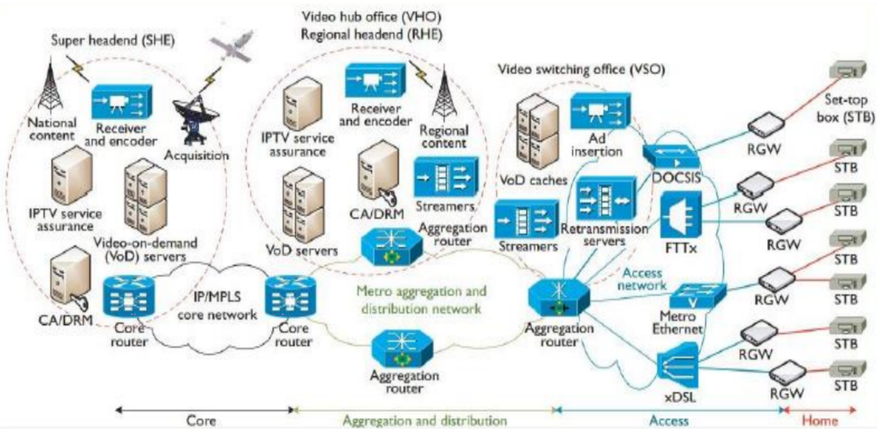
Disadvantages:
- **Wiring complexity in large networks**
- Problems of scalability



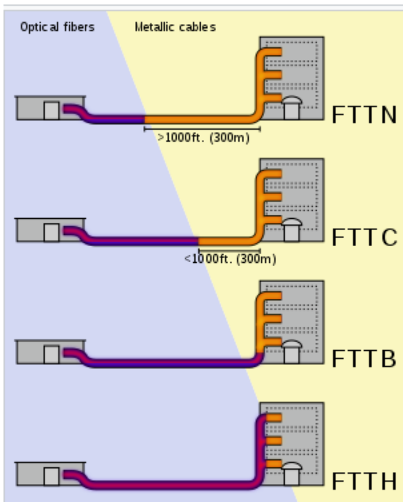
Note that:

- Each ToR switch is connected to all aggregation switches for redundancy;
- Each aggregation switch should be connected to more than one core/central switch to achieve redundancy and flexible load balancing;
- It usually comes 40Gbps from the rack but the switch only outputs 20Gbps, this is quite normal since the traffic is bursty. Overprovisioning would be to have the same output as the input.

A general overview of IPTV network:



DOCSIS Modem is used for HFC, there's FTTx (Fibre to the x, where x can Node, Curb, Building or Home, depending on how close the fibre is terminated) and xDSL (x Digital Subscriber Line).



A schematic illustrating how FTTX (**N**ode, **C**urb, **B**uilding, **H**ome) architectures vary with regard to the

The final is node in our homes in the set-top box (STB). Contains a TV-tuner input, display output for the television. Also contains, external source signal to receive the data from a cable, satellite or even other ways.

1.4 SDH - Synchronous Digital Hierarchy

SDH is not used very much, OTN is the current standard. MPLS-TP is yet another way of doing transport, using MPLS with Transport Profile.

To have deterministic links, the Management plane has to establish them.

There are multiple ways of doing multiplexing. In SDH (and in OTN), TDM is used.

To transmit our signal in SDH using TDM, it will have a slight overhead with the Start Frame Delimiter (SDM). There are many flavours of TDM. However, in a deterministic network, the timeslots are assigned in a fixed way. Therefore, even if one of the input channels doesn't have information to transmit, there will be a slot assigned to it and will be empty.

There are four accuracy levels (Stratum levels):

$$\text{Accuracy} = \frac{|f_0 - f_r|}{f_0}$$

Levels	Stratum 1	Stratum 2	Stratum 3	Stratum 4
	1×10^{-5}	1.6×10^{-2}	4.6	32

Stratum 1 clocks are atomic clocks (cesium or rubidium)

A GPS receiver can also be used as a Stratum 1 clock

Two clocks are synchronous if they operate with the same frequency and a constant phase offset ($\triangle\phi(t) = const.$). Moreover, they can be classified with other words regarding the similarities between frequencies and phases.

The asynchronous clocks can be classified as : mesochronous , plesiochronous and heterochronous.

Mesochronous clocks: have the same frequency but the phase offset is random.

Plesiochronos clocks: have the same nominal frequency, but the real one can be slightly different.

Heterochronous clocks: have different frequencies .

Regarding the accuracy of the frequency of the real clock (f_r), expressing in Parts Per Million (ppm):

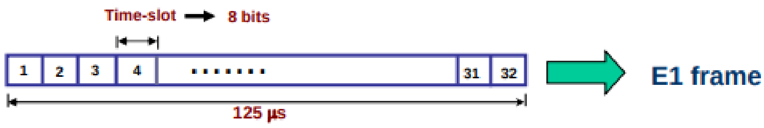
The transport network guarantees different functionalities as transmission, multiplexing, routing, protection, supervision and capacity provisioning.

A curiosity: nowadays is possible to find atomic clocks for less than 5k euros. Some as low as 1500 euros.

Because the sampling rate should be at least twice the maximum frequency of the signal to sample it properly (according to the Nyquist Theorem). Therefore:

A telephone channel uses the bandwidth between 300 and 3400 Hz. For a maximum frequency of 4000 Hz, we have a sampling frequency of 8 kHz, which gives a sampling period of 125 µs. Coding each sample with 8 bits one arrives to a bit rate of 64 kbit/s.

The frame of an E1 signal has 32 time-slots, corresponding to 32 channels, 30 of them are for data, one for the frame-alignment word (FAW) and one for signaling (control plane information).



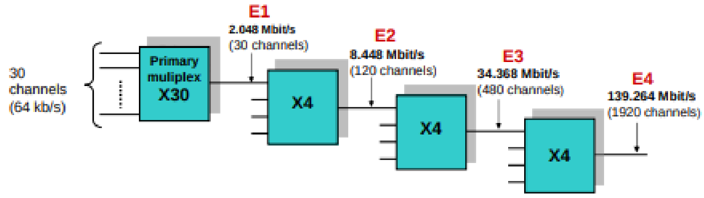
The duration of each time slot (8 bits) is equal to 125µs/32=3.9 µs, corresponding to 488.2 ns per bit, and a bit rate of 2.048 Mbit/s.

Recalling the meaning of Synchronous and Plesiochronous, now is possible to understand what are the Digital Hierarchies based on these types of synchrony. In the first (SDH), both ends are synchronized with a central clock while the second (PDH) doesn't require this synchronization.

In the PDH network, the frequencies are similar and depending on the multiplexing factor, they can drift a certain amount. If the amount of signals being multiplexed grows considerably,

The signals of the various hierarchical levels are denote with E*i* (*i*=1,2,3,4).

The first level of the hierarchy carries 30 telephone channels(64 kbit/s), while the next levels are obtained by multiplying the previous levels by four.



The clocks of the European Hierarchy require the following accuracy:

Hierarchy	E1	E2	E3	E4
Accuracy	50 ppm	30 ppm	20 ppm	15 ppm

For some reasons, the American and the Japanese networks follow slightly different standards, having different amounts of multiplexed channels.

Note also

But these PDH still use TDM? Maybe the drift in frequency is small enough to still allow it.

In SDH, the network operator has to provide the synchronization clock, the **PRC - Primary Reference Clock** .

Note that a synchronization network is required to keep all elements with the same clock. Examples of synchronous networks:

- SDH
- TDT - Terrestrial Digital Television
- GPON - Gigabit Passive Optical Networks is Synchronous as well. (OTN is not)
- Mobile Network
- PSTN

Regarding SDN then, one of the first proposals was called SONET (Synchronous Optical Network). SDN was later defined by ITU-T as an international standard compatible with SONET. TDM frames were used. Some more nomenclature:

The SDH basic signal is named *Synchronous Transport Module* (STM). The SONET basic signal is named *Synchronous Transport Signal* (STS) in the electrical domain and *Optical Carrier* (OC) in the optical domain.

SONET (Optical)	SONET (Electrical)	SDH	Bit Rate (Mb/s)
OC-1	STS-1	-----	51.840
OC-3	STS-3	STM-1	155.520
OC-12	STS-12	STM-4	622.080
OC-48	STS-48	STM-16	2488.320
OC-192	STS-192	STM-64	9953.280
OC-768	STS-768	STM-256	39813.120

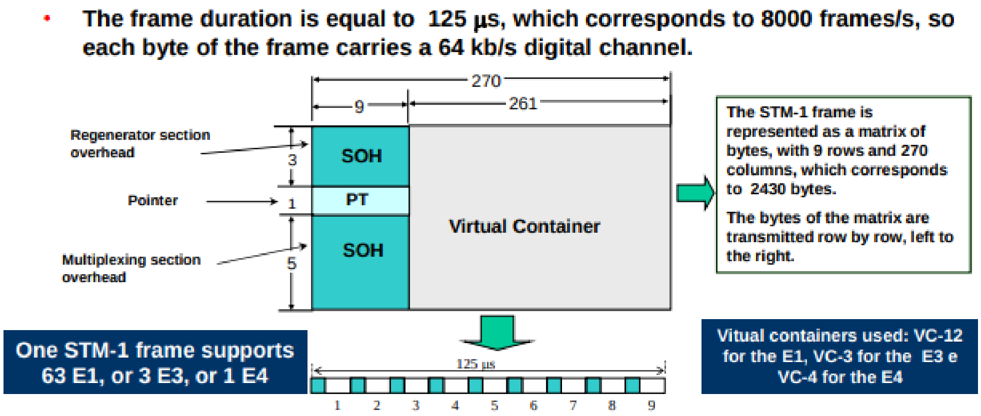
Note : SDH is compatible with PDH E-n signals. Since SDH is able to carry more information in each TDM frame, it can carry PDH hierarchies in its frames.

A SHD basic frame (STM-1) contains 3 blocks:

- SOH - Section OverHead for regenerations and multiplexing

- PT - Pointer to the Virtual Container start.
- VC - Virtual Container with the payload and the

Since this basic frame is sent every 125 micro seconds, 8000 frames per second is the frame rate. Note again that 155 MB/s is the bitrate.



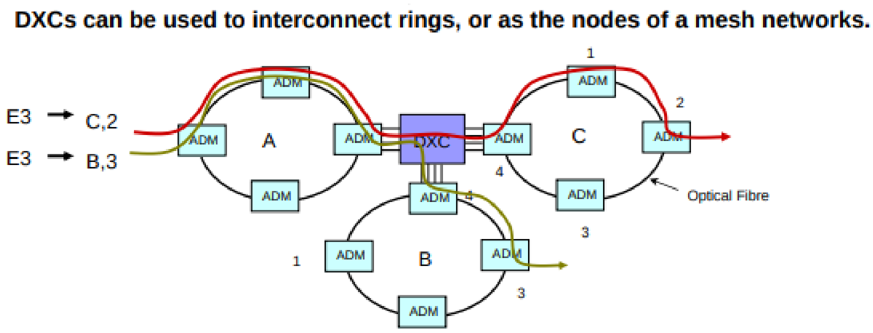
Some equipment that is necessary in this kinds of networks:

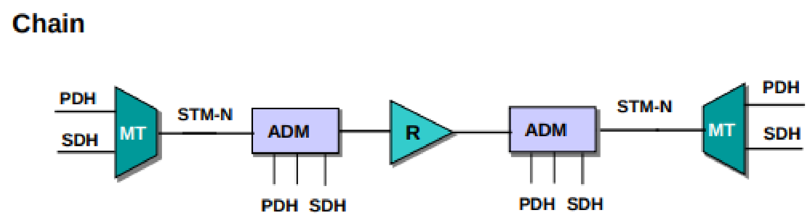
-An ADM (Add and Drop Multiplexer) selects one of the containers in the TDM frame to drop and one of its own to add, in case it is expecting containers or has containers to send, respectively;

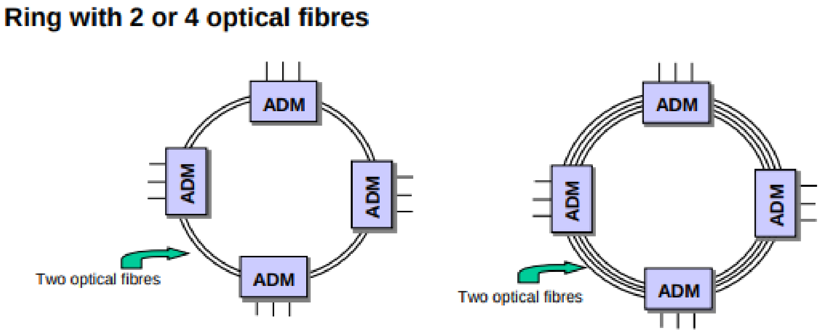
- A regenerator;

- A Terminal Multiplexer (TM) that multiplexes any amount of signals at the entrance to one signal at the output.

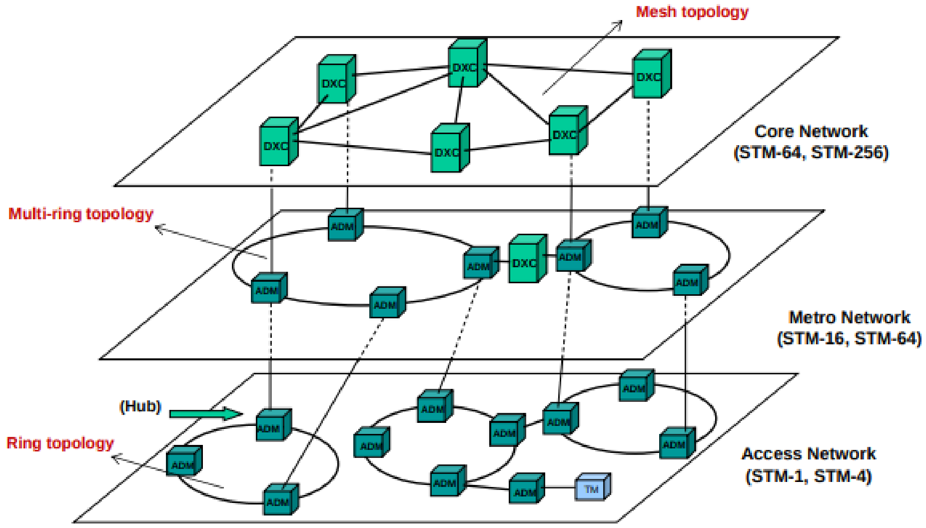
- A Digital Cross-Connect (DXC) is able to connect any input to any output, and is controlled by the management plane to establish the previously mentioned semi-permanent paths.





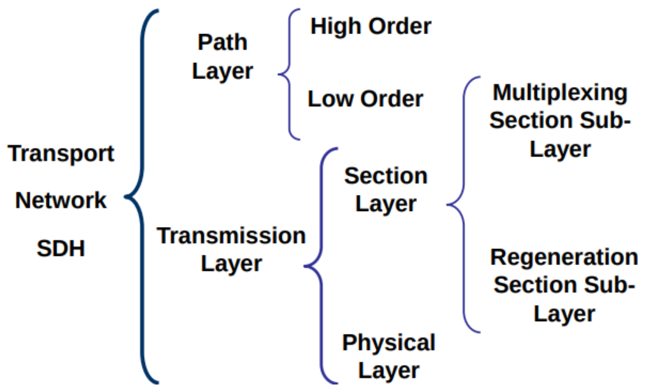


These equipments are used in accordance to the densification we have in the network. Closer to the core there are more DXCs, closer to the access there's more ADMs.



When we mention "containers" we are referring to Ethernet frames or other hierarchies of digital flows.

The transport Network SDH based can be divided in 2 parts, the Path and the Transmission.



This layers have important functionalities. Path layer functionalities are related with the connections, the paths established. High and Low order ....

These functions are why it is worth it to have extra overhead in all layers:

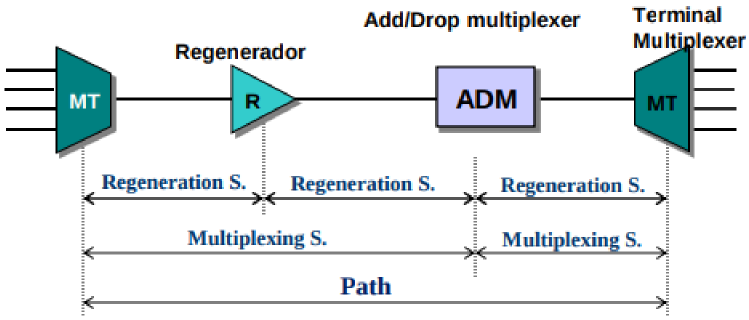
Path :
Connection integrity verification, defines the type of traffic transported in the path, error monitoring, protection switching.

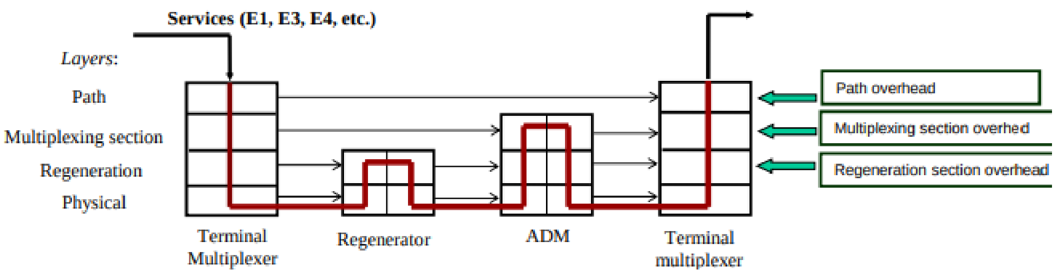
Multiplexing section:
Clock synchronization, protection switching, error monitoring, communication with the network management system.

Regeneration section:
Frame alignment signal, error monitoring, communication with the network management system.

Physical:
Optical pulses shape, power level, wavelength, receiver sensitivity, etc.

Note that these layers are similar to the OSI model layers, each of them has an overhead related to its *modus operandi* , exactly like there are IP headers and MAC headers:

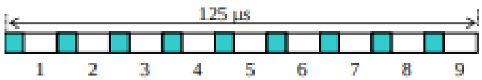




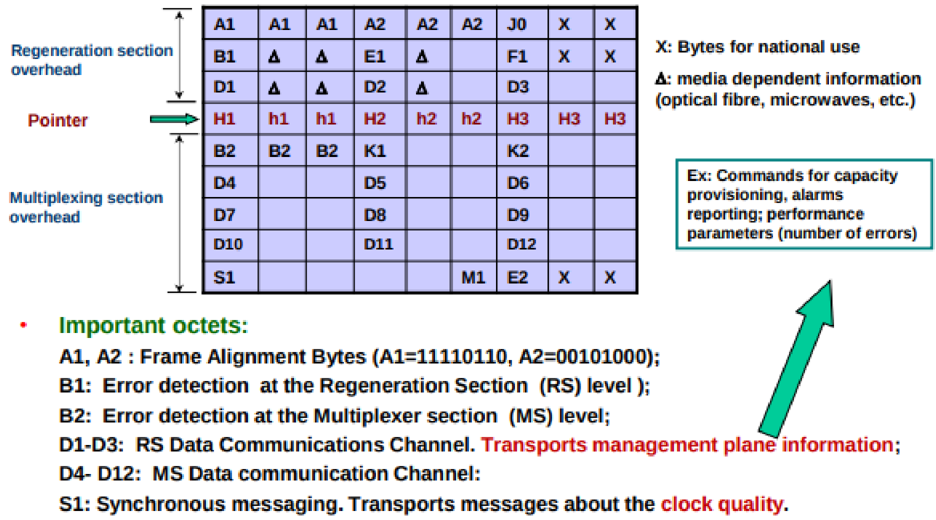
Note the scopes in a digital transmission! All of them do regeneration, but not all of them multiplex. In the multiplexing layer its where the clock synchronization comes in, note that despite being able to carry Plesiochronous DH frames, it this is a synchronous hierarchy, so it needs a PRC (primary reference clock).

To step back a bit:

1. An SDH basic frame (STM-1) contains section overhead (SOH), PT (pointer) to the beginning of the virtual container with the payload data and then the VC.
2. **One STM-1 frame supports 63 E1 frames, 3 E3 or 1 E4**
3. SOH has a regeneration section and a multiplexing section overheads.
4. A VC is a container with a path overhead: "The STM-1 frame is represented as a matrix of bytes, with 9 rows and 270 columns, which corresponds to 2430 bytes. The bytes of the matrix are transmitted row by row, left to the right." - That is why there's a 9 segment separation over the transmission interval, because the frame has 9 rows and each row is transmitted



5. In a VC there may be several containers + path-overhead
6. There it takes 125 microseconds to transmit a complete frame, therefore 8000 frames can be transmitted per second.
7. One payload byte corresponds to 64 kb/s because 8000 of those bytes will be sent per second. Therefore, to know the bitrate of a certain portion of the frame we just need to count the number of bytes and multiply it by 64kbps.
8. The pointer part of the section points to all container beginnings in the virtual container?
9. The Frame Alignment Signal sinalizes the beginning of the frame. Despite
10. The SOH format is:



Note that there are 3 bytes in a frame to transmit Management plane information regarding the Regeneration. D4-12 are the correspondent for Multiplexing. B's are for error corrections for both those planes.

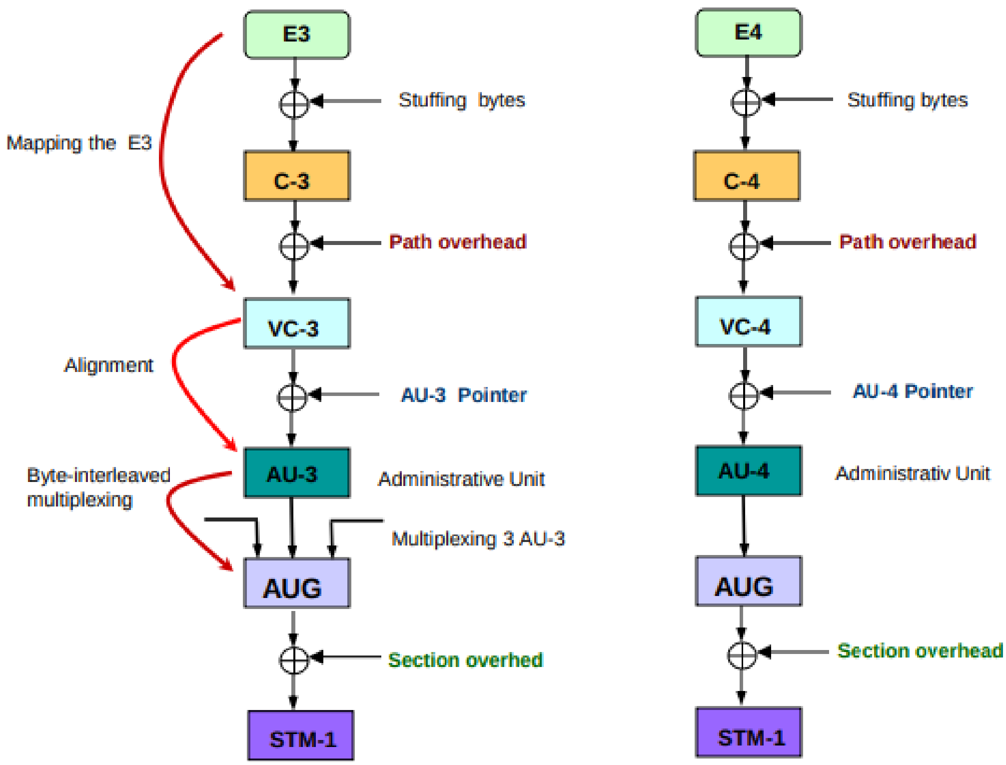
11. There are multiple Virtual containers sizes, all 9 rows long:
    - VC-4 - 261 columns long (1 path overhead, 260 C4 payload)
    - VC-3 - 86 columns long (1 path overhead, 85 C3 payload)

    Note: In a Virtual Container is possible to carry 1 VC-4 or 3 VC-3s

12. A VC-4 with a pointer is called an AU-4. A VC-3 with a pointer is called an AU-3. A frame may have 1 AU-4 or 3 AU-3.

13. The VC-4 can fluctuate inside the AU-4, it doesn't need to start in the beginning. The same of VC-3. If it doesn't start in the beginning, it can simply continue to other frames.

14. Overall:



AUG aggregates all AUs. If AU-4 is used, then AUG is AU-4, else, it includes more than one AU.

Management plane uses a DCC (Data Communication Channel) which has bytes in the frame overhead, more specifically, D1 to D12 bytes.

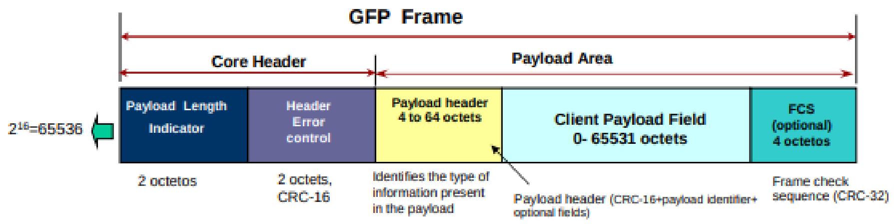
Control plane uses B types of bytes.

However, S type of bytes are of the responsibility of the data plane since synchronism is nothing more than expecting bits at certain times.

GFP

Generic Framic Procedure (GFP) is an ITU-T standard on how to map packet traffic like Ethernet and SAN (see ahead), which is of bursty nature, <u>into SDH and OTN with a constant bit rate</u> .

GFP client frames:



Other type of GFP frames are GFP control frames, where only the header is sent.

Storage Area Networks

A very redundant server and disk access network, stores information for a big quantity of data. Each server is connected to more than one switch and each switch aggregates more than 2 disk/storage arrays.

One of the most common SAN protocols are Fibre Channels that range from 1 to 128 Gbit per second.

Virtual Concatenation

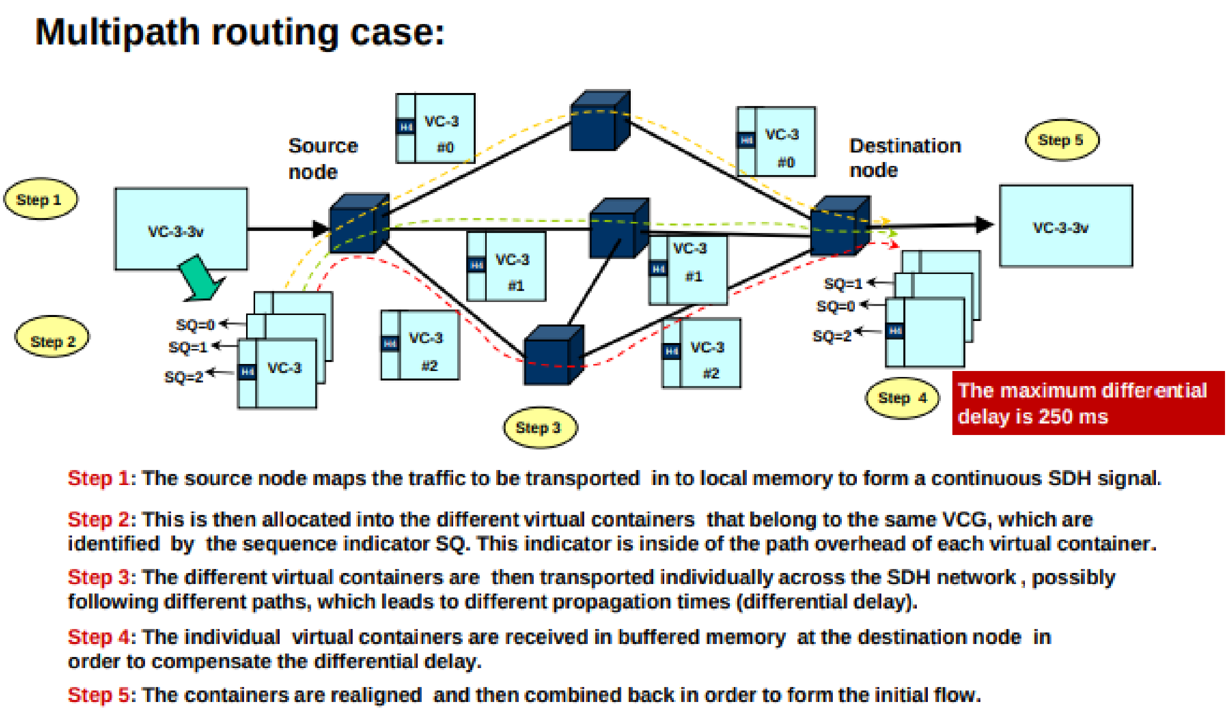
How is SDH able to cope with 1GbE? It uses virtual concatenation: creates **X** output flows with STM-1 frames. This is called inverse multiplexing.

Containers	Type	Available capacity (Mb/s)
VC-11-Xv	Low order	X x 1.600 (X=1,..,64)
VC-12-Xv	Low order	X x 2.176 (X=1,..,64)
VC-3-Xv	High order	X x 48.384 (X=1,..,256)
VC-4-Xv	High order	X x 149.76 (X=1,..,256)

Therefore, with $X = 256$, is possible to achieve $256x150Mbit/s = 38.4Gbit/s$.

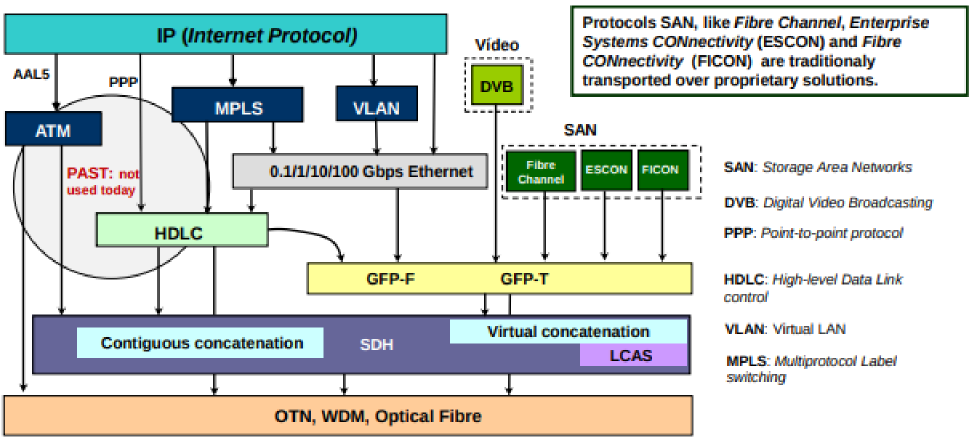
This technique can also be used to do balancing in the network since it implements multipath routing on the physical level since the X SDH flows may not do the same path.

There's a field called H4 that has the sequence number of these containers. 8 bits are able to index 256 containers, that is why X is limiter to 256.



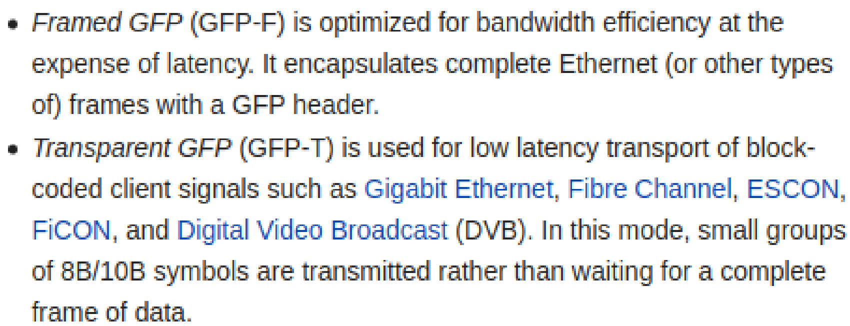
IP over SDH / OTN / WDM

Note that in order to use SDH, a synchronous MPLS is one of the important ones. But be aware that MPLS, SDH and OTN are in different planes!



In summary:

- MPLS, VLANs and directly IP use Ethernet which needs GFP to convert this non constant bit rate traffic to the constant bit rate of SDH;
- SAN and DVB are also supported by GFP (GFP-F and GFP-T) As a curiosity:



- *Framed GFP* (GFP-F) is optimized for bandwidth efficiency at the expense of latency. It encapsulates complete Ethernet (or other types of) frames with a GFP header.
- *Transparent GFP* (GFP-T) is used for low latency transport of block-coded client signals such as Gigabit Ethernet, Fibre Channel, ESCON, FiCON, and Digital Video Broadcast (DVB). In this mode, small groups of 8B/10B symbols are transmitted rather than waiting for a complete frame of data.

- DVB transmits without compression? FHD = 1920 x 1080 = 2073600 QHD = 2560 x 1440 = 3686400 UHD = 3840 x 2160 = 8294400 (4 times as much as FHD) If we have 24 bit color depth, 8 bits per color, we would uncompressed bit rates above 1Gbit/s for FHD at 30 fps. With compression we get around 10 Mbit/s.

  DVB includes Satellite, Cable, Microwave Links portabilities.

Generalization of Network Element

MultiService Provisioning Platform

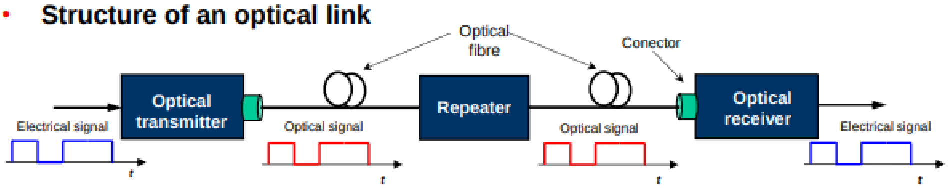
- *Framed GFP* (GFP-F) is optimized for bandwidth efficiency at the expense of latency. It encapsulates complete Ethernet (or other types of) frames with a GFP header.
- *Transparent GFP* (GFP-T) is used for low latency transport of block-coded client signals such as Gigabit Ethernet, Fibre Channel, ESCON, FiCON, and Digital Video Broadcast (DVB). In this mode, small groups of 8B/10B symbols are transmitted rather than waiting for a complete frame of data.

1.5 OTN - Optical Transport Network

These can be of 3 types: opaque, transparent or translucent.

Opaque: Electrical nodes, optical links. Transparent: Optical nodes, optical links. Translucent: Some nodes are optical, some are electric, optical links. Nodes are ODU switches. Depending on the type of network (opaque, transparent or translucent), these switches can have line cards at the entrance (equipments that transform the optical signal to the electrical domain) or work fully in the optical domain.

- **Structure of an optical link**

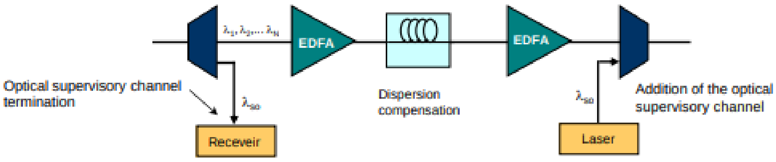


- **Optical transmitter: consists in an optical source and electrical circuits; the optical source is usually a diode laser, which converts electrical signals into optical signals) (optical modulation)**
- **Optical receiver: consists in a photodiode, which is responsible for converting the signal from the optical domain to the electrical domain (optical detection), and electronic circuits appropriate for amplifying the signal.**
- **Repeater: can be an optical amplifier, or a regenerator; the first device amplifies the optical signal and the second one gives to it the original format.**
- **Optical fibre: consists in an cylindrical guide, usually made of glass, which permits the transmission of the optical signals.**

Note the distinction between optical repeater (represented slightly below) and a regenerator. Regenerator receives the signal and sends it again, i.e the signal passes through the electrical domain. Such thing doesn't happen in transparent networks where everything is optical.

Every 80-120 kilometers an optical amplifier is needed. It belongs to the link.

An optical Line Amplifier has to compensate dispersion and amplify the signal. EDFA: Erbium-Doped Fiber Amplifier have an automatic gain control to keep power constant at their outputs.



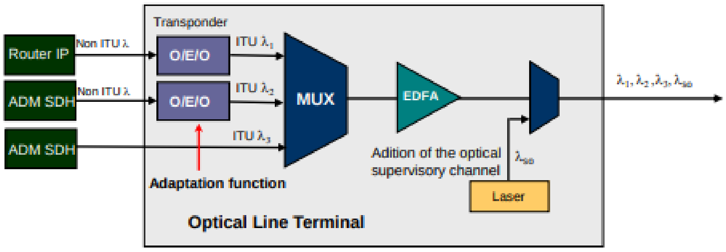
The optical supervisory channel is filtered at the input and terminated, and added back at the output. This channel is carried on a separate wavelength (1510 nm) different from the wavelenghts carrying the actual traffic, and is used to monitor the performance of amplifiers along the link, and to control them. It can also be used to carry the DCC (*Data Commumicatiom Channel*) and overhead information.

There's a signal sent in a different wavelength that has the infomation about the amplification. This signal is called Opctical Supervisory Channel and is terminated and at the input side of the amplifier and added at the output. It carries software updates and management information. One way of compensating dispersion it to use dispersion compensating fibres and make the signal propagate there. Fibres that may have higher attenuation but that make the pulses closer together.

There are 3 main network components:

- Optical Amplifier;
- Optical Terminal Multiplexer

They are used at both ends of a point-to-point link to multiplex and demultiplex wavelengths. Includes three functional elements: transponders, wavelength multiplexers and optical amplifiers.



The adaptation undertaken by the transponder corresponds to the following functions:
- Wavelength conversion in order to have at its outputs ITU-T λs;
- Addition of overhead for the purposes of network management ;
- Addition of FEC (*forward error correction*);
- BER (*bit error rate*) monitoring.

- ROADM (Reconfigurable Optical Add & Drop Multiplexer) - can have degree 2 or more (up to 8 or 9 nowadays) and must always have a conection to the client, normally connected to IP/MPLS routers, SAN switches, SDH networks, or overall GFM data;
- Optical Cross-Connect (OXC) - not very used since multidegree ROADMs are a thing.

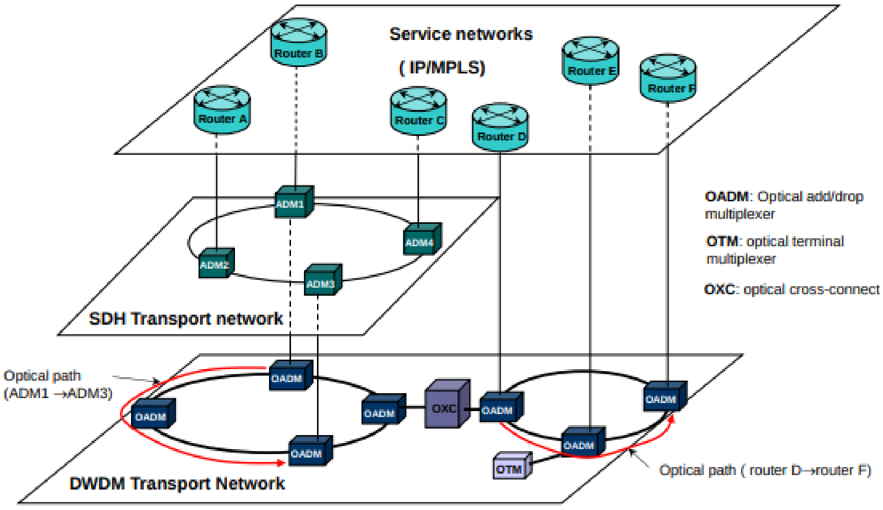
The same way SDH networks have Regenerators, Terminal multiplexers, ADMs and DXCs, OTN has Optical Amplifiers, Optical Terminal Multiplexers, ROADMs and OXCs

Optical networks operate between 800 and 1600 nm wavelengths, which are in the infrared region of the spectrum.

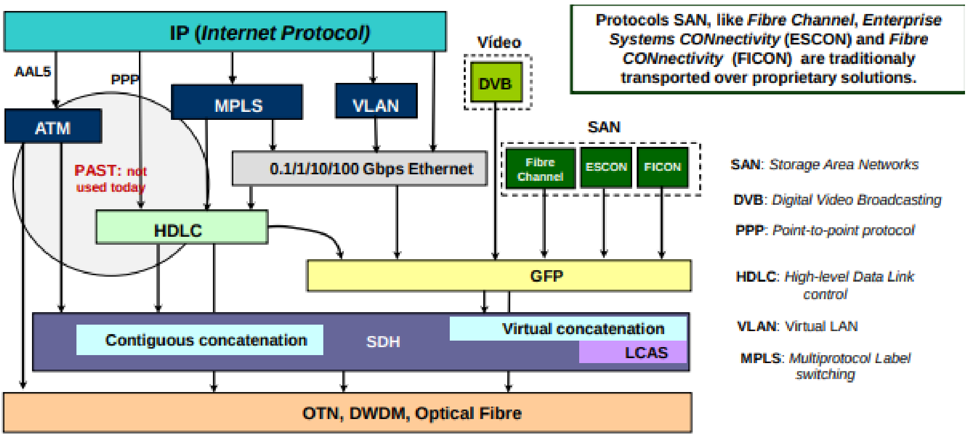
1310 to 1550 are particularly used in communications due to low attenuations.

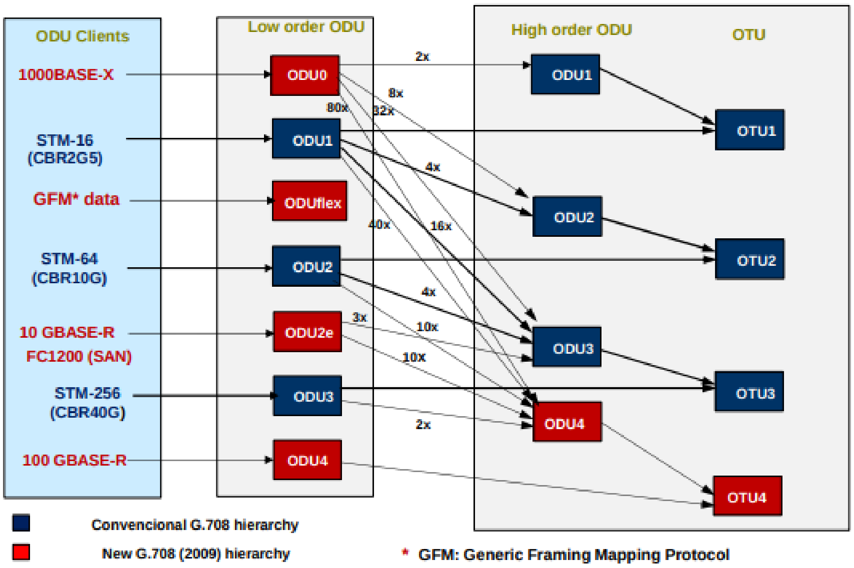
1.5.1 OTN as the transport Base - Electical Layer

MPLS is connected to the optical equipment with Ethernet, 10 or 100GbE. SDH is connected to OTN with SMT-16, STM-64 or STM-256 frames. SAN is connected with optical channels.



More generally, OTN can have all clients, it is just a compatibility matter and the standards have that considered.



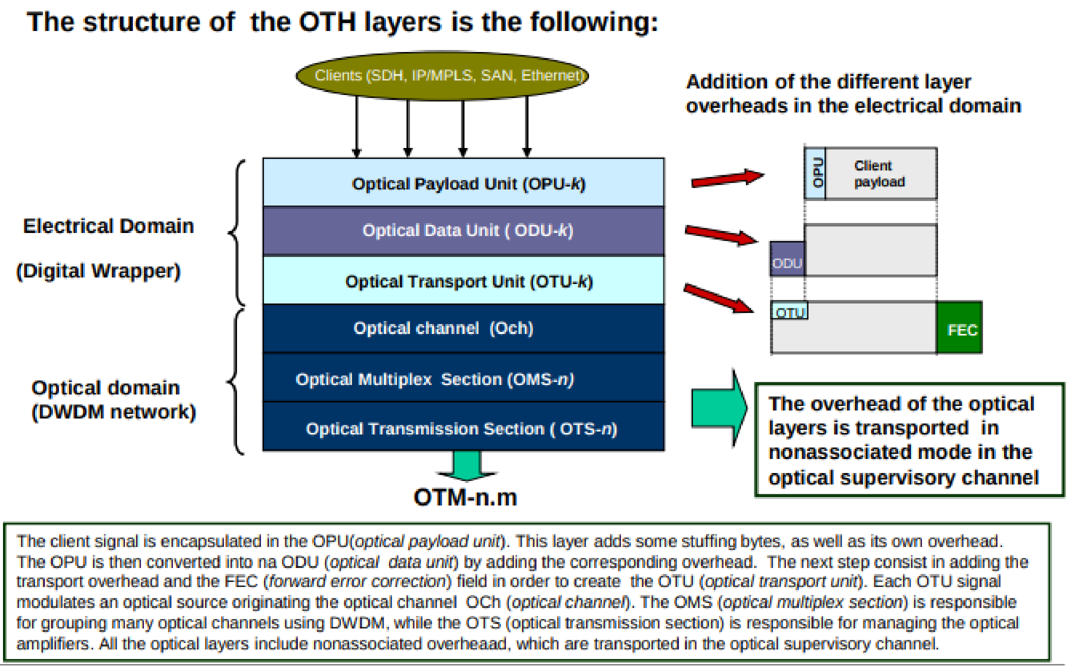


Keep in mind:

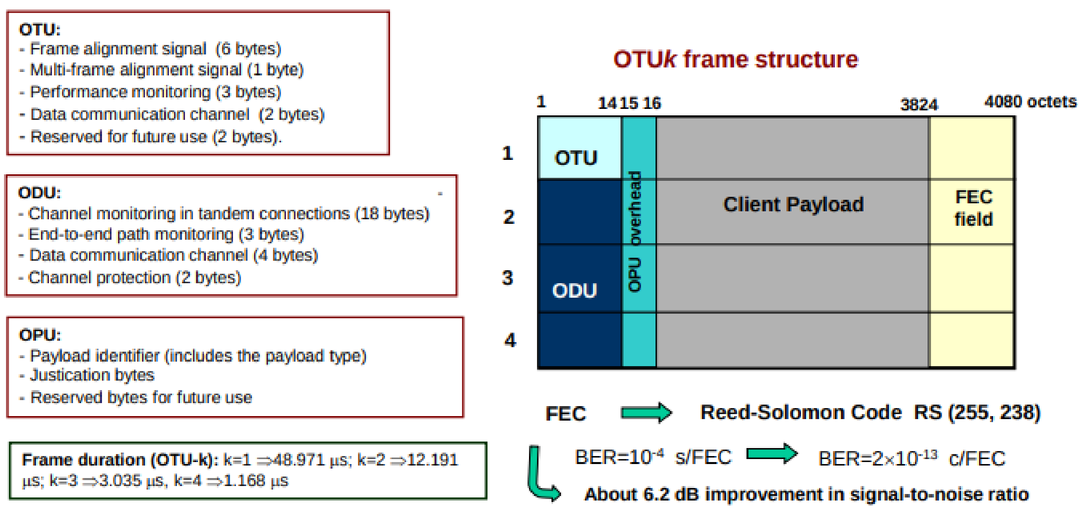
- 1000Base-SX is 1 Gbps short ranges, 8B/10B code;
- CBR means constant bit rate, expected for SDH networks;
- The same way SDH is an hierarchy, OTH is also an hierarchy;
- In OTH, the first step is in the electrical domain, the second is in the optical domain;
- The same way SDH had STM's (STM-4, STM-16, ect…), OTH has OTU (Optical Transport Unit,

OTU-1, OTU-2, etc...);

- Client data + OPU header = OPUk - Stuffing, client data type, etc... OPUK + ODU overhead = ODUK. ODUK + OTU overhead = OTUK:
  Optical Payload Unit (equivalent to C-4) + ODU overhed (equivalent to Path header) - ODUk. The purpose of ODU is to monitor end-to-end performance and do protection switching
  Optical Data Unit k + OTU overhead + FEC = OTUk - Added at the end of the OTUk is added a Forward Error Correction (FEC) field, computed over the ODUk. The OTU overhead has a Frame Alignment Signal (FAS).
- All the above mentioned operations take place in the electrical domain. Now is possible to go to the optical domain!



- Note that ODU2/OTU2 has 4 times the bitrate of ODU1/OTU1. ODU3/OTU3 has 4 times the bitrate of ODU2/OTU2 and 16 times the bitrate of ODU1/OTU1. There also is ODU0 which has half the bitrate of ODU1. Because there isn't any OTU0, 2 ODU0 must be places in ODU1 which is then place in a OTU1. For ODU3 or ODU4, the ration isn't the same and we should check the image.
- Having an ODU-k, the direct solution is to use OTU-k. But there are higher order solutions as well: to aggregate several ODU-k into OTU-k+1 or OTU-k+2, etc... We do this aggregation.
- The structure of the frame is always the same, what changes is the frame duration. For an higher k, the frames are sent faster.



OTUk bitrates can be calculated in the same way as in SDH: number of bit/bytes in a frame divided by the frame duration.

- Note that OTN is a plesiochronous network, because frame durations are not constant.
- The Error correcting code is able to correct at most 16 wrong bytes in a row. 255-238 ....???

Overall, the OTN has an hierarchy called OTH. This OTH has two layers, electrical and optical. The electrical layer has 3 layers, OPU, ODU and OTU. Each subsequent layer adds headers to the previous ones. OPU is OPU-header + client payload. ODU is ODU-header + OPU. OTU = OTU-header + ODU + FEC, where FEC is calculated over the ODU to check for errors.

OTU-k, for k from 1 to 5, has the following bitrates: OTU-1 supports 2.5 Gbit/s, OTU-2 supports 10Gb/s, OTU-3 supports 40Gb/s, OTU-4 supports 100 Gb/s and OTU-5 supports 400Gb/s.

The OTU frame is a TDM frame. OTN is a non-synchronous network, called a plesiochronous network. In a synchronous network, the clock information is propagated from a master. In plesiochronous network there's no clock information, only tools for synchronization. In optical networks is very difficult to have the complete synchrony. In the electrical part is possible and they function in an autonomous.

Another very good way of knowing if a network is synchronous or plesiochronous, is by the frame duration. In SDH every frame lasts 125 microseconds while in OTN, different frames types (k order) have different durations.

1.5.2 Optical layer

In an Ethernet network is possible to use optical fibres, however in OTN the fibres are used with much higher data rates, therefore there are different wavelengths: 1310nm and 1550nm are the most common. With these wavelengths, single mode fibres are possible.

1310 nm (0-band with 100nm of length, correspondent to 17.48 THz of bandwidth) are more used in datacentres, because the distances are not that high and 1550 nm is used in OTN. 1550 nm has 0.18 dB/km, the physical limit is 0.16 dB/km.

With the OTK-k frame, that bitrate is passed through a laser-diode that emits radiation in the visible spectrum, in a given wavelength. This signal is converter to an optical channel.

A **Transponder** will take the client data, create OPU, ODU, OTU and generate an Optical Channel.

A quick example: what would be the block diagram for sending 10 GBASE-R signals? These signals can only use ODU2e ODUs, and 10 of them only fit in one ODU4. In theory, it would also be possible to place them in 4 ODU3. Therefore, there would have to be a parallelisation until the OTU4 multiplexing and only one ODU4 would be converted to an optical channel.

Muxponder is a transponder that also multiplexes. A muxponder has several clients when a transponder has only one.

Since outputting light is the expensive part (note that a process in the electrical domain as already used in Ethernet), the price per client is smaller if they share a muxponder.

DWDM means Dense WDM, which is when many wavelengths are used. WDM term is used when the amount of wavelengths is around or smaller than 16.

To the people that have fibre to the home (FTTH), each home will have a multimode (cheap) fibre that takes the traffic to a multiplexer at the building, and another at a street level. After these, DWDM is done. Also, remember that the terminal multiplexer is always degree 1. It only matter how many of the actual connections to the network it has.

In SDH, degree 2 equipments were called ADM (Add & Drop Multiplexers). In OTN it's OADM to allow frames to be droped and added (on the side that is connected to the clients). Because it is connected to the transport fibre on 2 sides (in & out), it has degree 2.

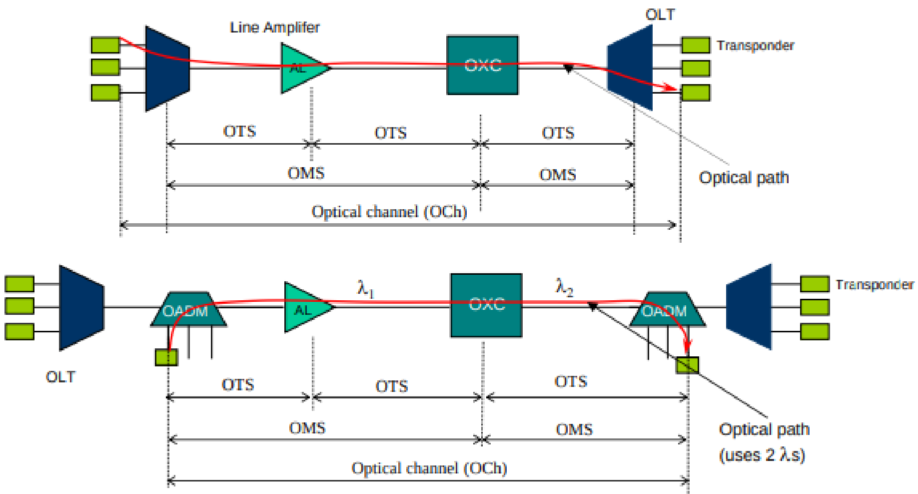
ROADM is used, the R is for Reconfigurable and the reconfigurable part is the possible wavelength of use.

There are 3 optic layers:

Optical channel layer:
Channel dispersion accumulation, channel identification, channel protection switching.

Optical multiplex section layer:
Wavelength division multiplexing, wavelength assignment, multiplex section protection switching, wavelength conversion.

Optical transmission layer:
Optical amplification, dispersion compensation through the optical line amplifiers.



At the end, an Optical Transport Module n.m is generated. n = number of optical channels m = k, for the OTU-k. If m = 0 is because there's a mix of bit rates.

1.5.3 Routing and Wavelength Assignment

Route according to the traffic demands, obtain the lightpahts and then choose the wavelengths to be used in each lightpath.

The routing problem is known from other networks. The Wavelength assignment has some rules:

- Wavelength Continuity - All the links of a lightpath must be the same wavelength;
- Distinct Wavelengths - All the lightpaths that traverse the same link must have different wavelengths.

This is common sense, but in a pool of wavelengths, how to choose?

Dynamic Wavelength assignment strategies:

- First Fit - Choose the first that fits. There are a couple of paths, check which is the lowest wavelength available (not necessarily the shortest wavelength, the wavelength with the lowest index, there's an abstraction) and assign that one.
- Most used -
- Longest First -

In Dynamic wavelength assignment, the wavelength is assigned as the demands come. In Static WA, we can simple order the demands according to shortest first, longest first, largest first or random and do the same thing as in Dynamic but considering that the new order is the order of arrival.

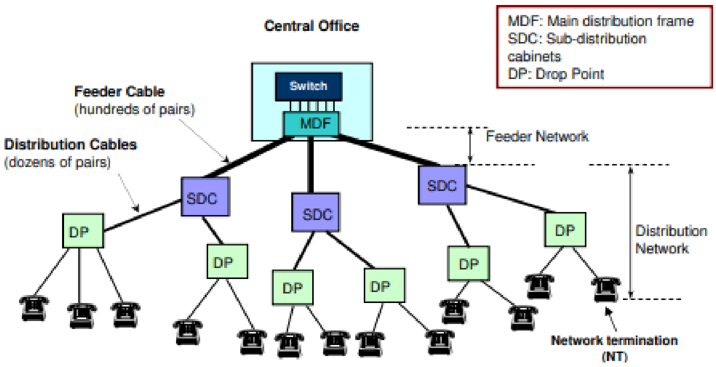
1.6 Access Network

1.6.1 Intro

The networks that allow the connection of the client with the operator.

This is the simplest yet the most expensive part of the network.

Regarding the Public Switched Telephone Network.



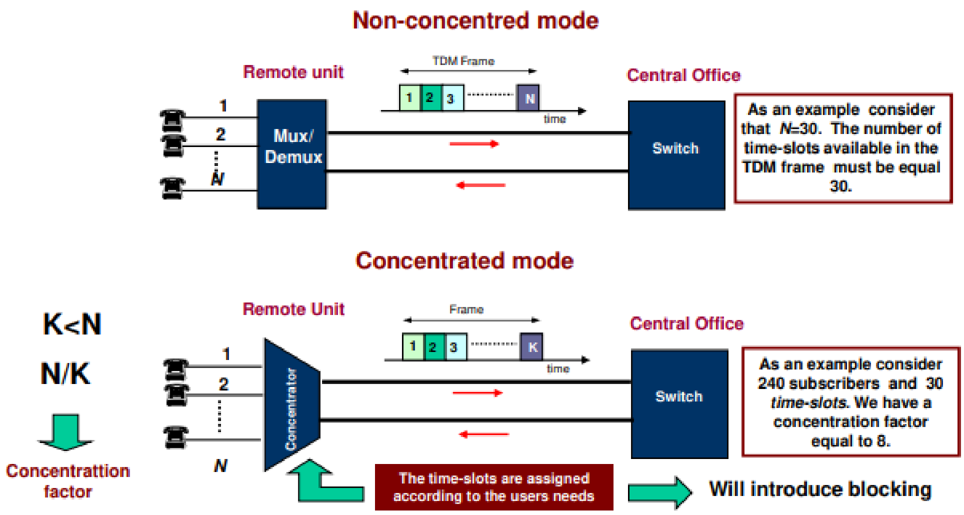
The Main distribution frame will connect the different pairs of Simetric twisted copper cables (200 or more cables), reaching the SDC it will be divided in groups of 30-50. And reaching the Drop Point it gets divided to unity, one for each client.

If we want to replace the feeder cable that has hunders of twister pairs by a fibre one has two options:

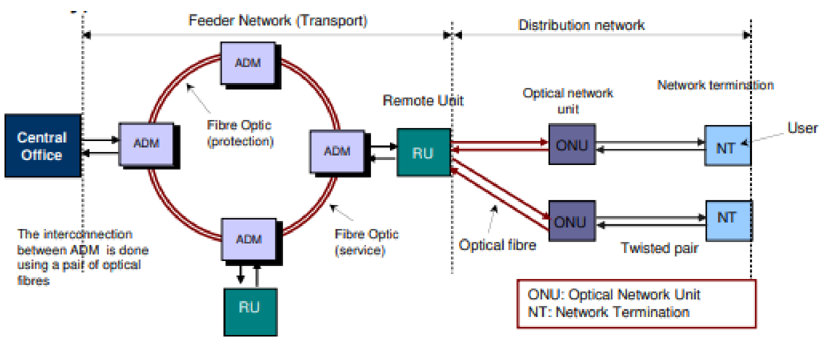
- since the fibre as much wider band that the twisted pair, one option would be to put the signal of each pair in a wavelengt. However, WDM is very expensive and one wavelength would still be too much for a copper cable.
- In order to avoid WDM, one can simply use one wavelength as multiplex in the time domain, TDM with 200 slots. Here we can't use statistical multiplexing because here is done circuit switch and the flow is given by the timeslot where it arrives. However, if we

Replacing the feeder cable by an optic fibre, is necessary to multiplex / demultiplex and convert to/from the optic domain. Note that we need one optic fibre per direction. Here the SDC are called Remote Units.

Because all the channels are not used at the same time, exceptionnaly in the new year's eve, less channels (K of them) can be used without a problem. $\frac{N}{K}$ is the concentration factor. In an image:

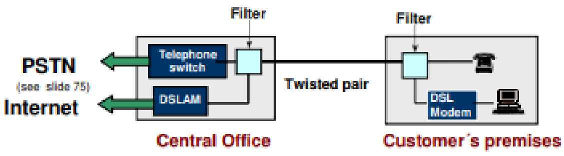


A redundant way of connecting Central Offices and RU's can be a mesh or rings.

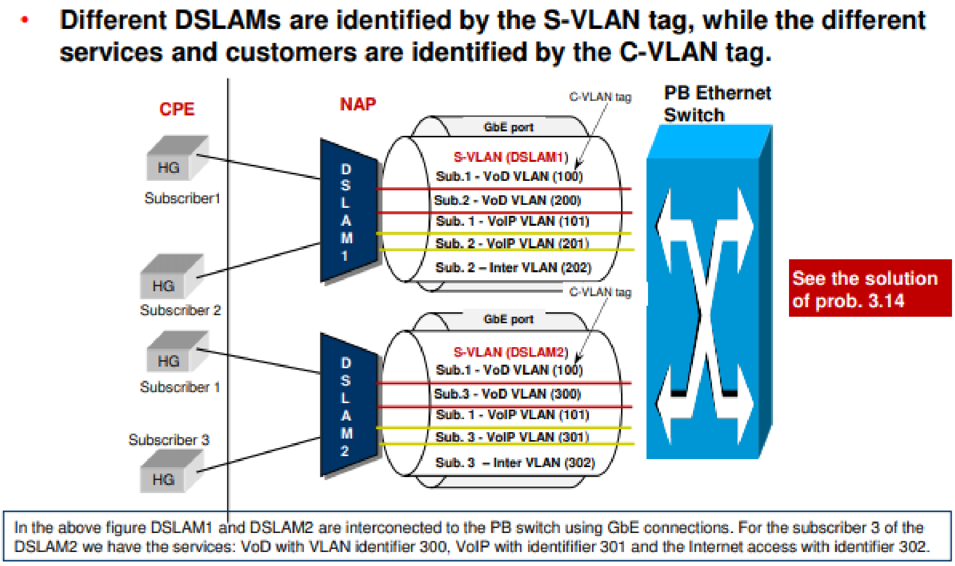


A few years ago, in order to use these copper cables to access the internet, one could modulate an internet signal in the copper cable. By using a Digital Subscribel Line (DSL) Modem (Modulator Demodulator), taking advantage of 4 kHz band, used for voice, one can get 56 kbit/s. Internet access and voice go in the same cable in different frequencies.

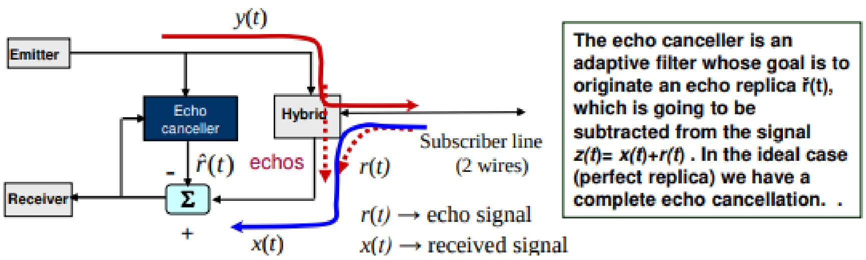
In the Central Office, the DSL Access Multiplexer (DSLAM) is connected to internet, at the time using Ethernet.



The DSLAM connects to an aggregation network, more specifically, to a PB switch. Also, VLANs are used to identify different services.



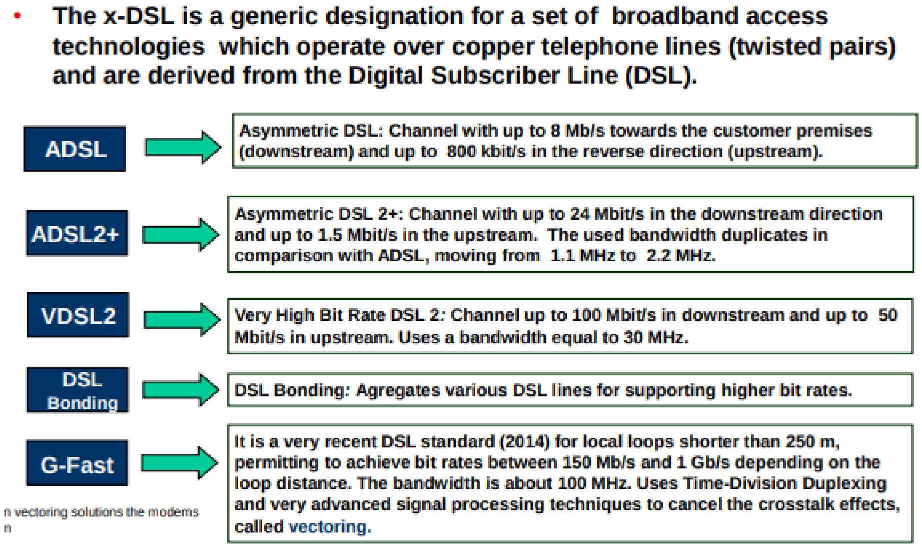
Different types of duplexing are done in the copper cable. Echo Cancelation Duplexing is based on the cancelation of echos through the generation of similar signals and doing their difference. Echos happen due to impedances mismatches. $\hat{r}(t)$ is created and will try to nullify the echo of $y(t)$.



1.6.2 Copper vs Fibre

The conventional access network is the network described so far, the one used in the PSTN. However, copper twisted pairs can still be used for internet access. When this is the case, technologies that use twisted pair for internet access use the DSL (Digital Subscriber Line). DSL is the name technologies that use the telephone line have. Obviously, nowadays it is used much more than just for telephone and it that cable dangling of 2 posts. Because it uses copper pairs, it is very cheap.

Nowadays, VDSL, ADSL2+ and ASDL are the main technologies over the DSL.

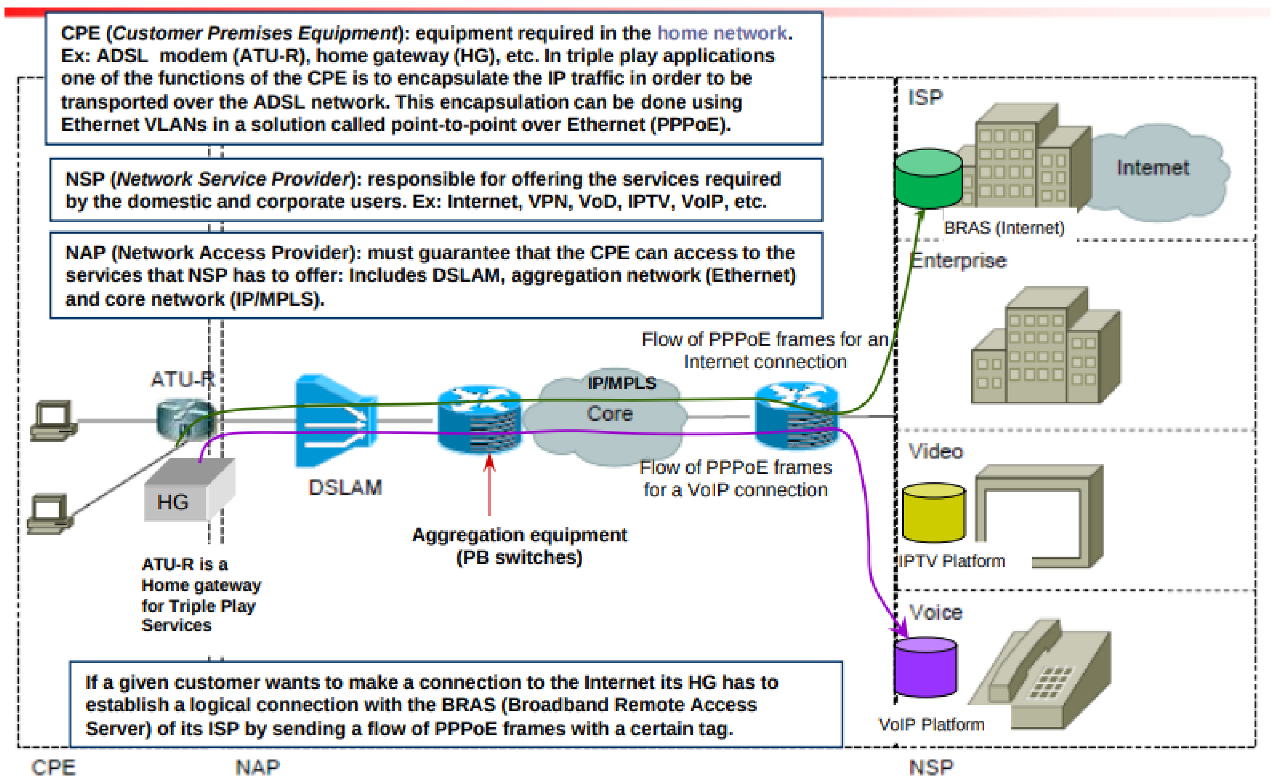


The main problem with copper cables is the range. In copper there are severe problems with attenuation therefore the code is very limited. The bitrate is a direct function of the distance.

In fibres the problem has more to do with dispersion and are capable of getting much further. That and the huge bandwidth is why they are replacing copper almost everywhere.

In ADSL and posterior copper technologies, the DSL Access Multiplexer (DSLAM) is responsible for the aggregation along with PB Switches. Between the access and the metro networks there is always an aggreagtion layer.

The image below is quite important. It shows how a triple play network is completely connected. In essence, the television is connected to a set-top box, which is used to retrive the tv signal from the modulation. Sometimes from the cable, sometimes from the over-the-air sometimes even from a satellite. Nowadays it only decodes IPTV video stream. All the remaining client devices, phones, laptops (. . .) are connected to an Home Gateway which is also a router and an wi-fi access point. The HG is connected to a Modem (in case of ADSL) that will perform the modulation and demodulation.



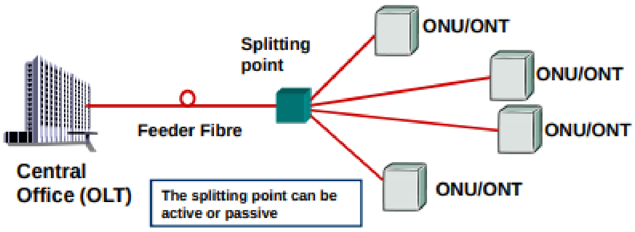
Clearly, from the picture as well, there must be a Network Access Provider that guarantees the connection between the Service Suppliers (IPTV platform, VOIP platform and Internet given by an ISP).

Something important is to know how the flows are separated at a client level, at a service level and at a DSLAM level, because the ethernet frames must reach the correct DSLAMs. For this, VLANs are used with 2 distinct tags, C-tag and S-tag. The C-tag (client) has information about the client and the type of flow/service the client is requiring. The S-tag (Service), despite the name, refers to the DSLAM where that client is connected. These tags will be read by the PB Ethernet Switch.

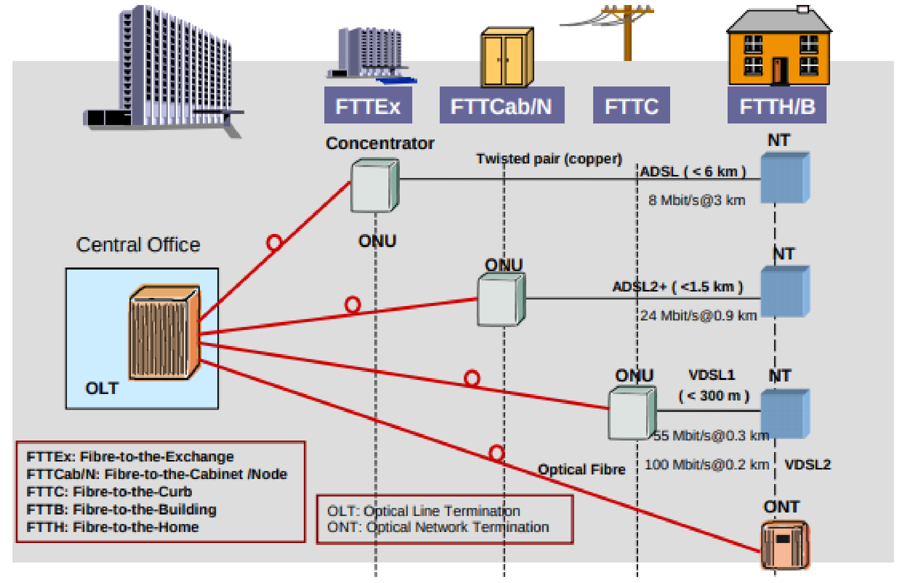
The change to fibre

We've said before that it can be done at a feeder level. But it can and is being done at distribution network! It has to use a P2MP (Point to Multipoint) connection in order to save costs in fibre.

The equipment that performs the aggregation of many fibres and that is equivalent to a DSLAM in functionality is the Optical Line Terminal (OLT).



It is connected to a ONU or a ONT, depending on how far the fibre reaches into the home of the client.



If the fibre reaches the building or even the Home, it is called an ONT. Else, if it reaches only the concentrator, cabinet or curb, the node that transforms it to the electrical domain is the ONU.

1.6.3 Fibre: Active vs Passive

Active Star (Active Ethernet)

The splitting point is an active node, like an Ethernet *switch,* which is used to aggregate traffic coming from the different ONUs/ONTs. It is a combination of switched Ethernet with P2P links.

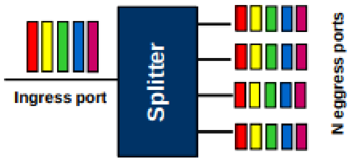
Passive Star (PON)

The splitting point is passive, being implemented with a passive *splitter/combiner*: *Passive Optical Network* (PON). The passive splitter is responsible for dividing the optical power into *N* separate paths to the users.

We work with the passive. Active would be something like normal ethernet switches. However, those are hard to power.

Therefore, Passive Optical Networks (PON) are whats most widely used.

A key element in a PON is the splitter. With directional couplers it reproduces in every exit the same as the entrance but with a fraction of the power.



Additionally, there are different multiple access PONs. In the OLT stuff needs to join to get into the same feeder.

TDM/PON: The multiple access operates in the time domain (TDMA: Time Division Multiple Access), this is each ONU transmits information within a specified assigned time slot. Two ONUs can not transmit at the same time.

WDM/PON: The multiple access operates in the wavelength domain (WDMA: *Wavelength Division Multiple Access),* this is each ONU transmits information in its own wavelengths. Two ONUs can not transmit in the same wavelength.

Of these 2 types, TDM-PON is the most used. WDM-PON is probably just used in Korea..

The most used TDM-PON variants are the GPON (Gigabit PON) and EPON (Ethernet PON). The first operates at an aggregated bit rate 2.488/1.244 Gb/s and the second one 1.25 /1.25 Gb/s.

1.6.4 Passive: EPON vs GPON

A summary right at the beginning:

	GPON	EPON
Standard	ITU-T G984	IEEE 802.3ah
Upstream bit rate	1244 Mb/s	1250 Mb/s
Downstream bit rate	2448 Mb/s	1250 Mb/s
Maximum splitting ratio	1:64	1:32 ; 1:16 (tipical)
Maximum range	10/20 km	10/20 km
Maximum attenuation	20/25/30 dB (Classe A, B, C)	20/24 dB (10, 20 km)
Average efficiency (ε)	≈ 93%	≈ 65-70%
Traffic	Ethernet, ATM, TDM	Ethernet
Average bit rate/ONU	≈ 70 Mb/s @ 1:32 (ε=92%)	≈ 55 Mb/s @ 1:16 (ε=70%)

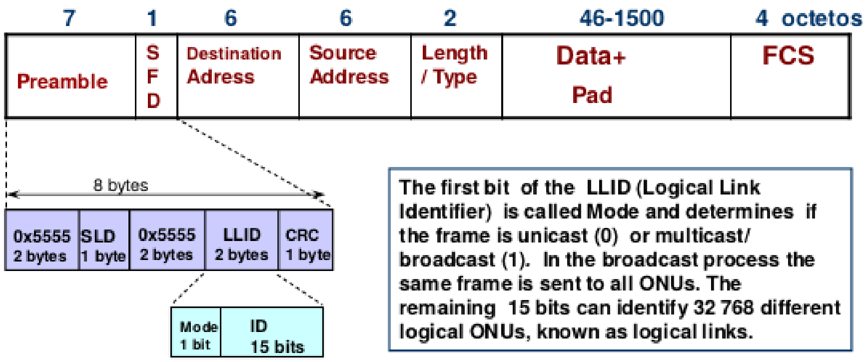
The efficiency refers to the fraction of the bit rate used for the data transport. The lower efficiency of the EPON comes from longer guard times and a larger overhead used for error detection and line coding.

EPON

Ethernet over PON, basically.

EPON frames are like:

The EPON frame has the same size and format as those of a standard Ethernet (IEEE 802.3) frame, but the preamble/SFD field is modified.



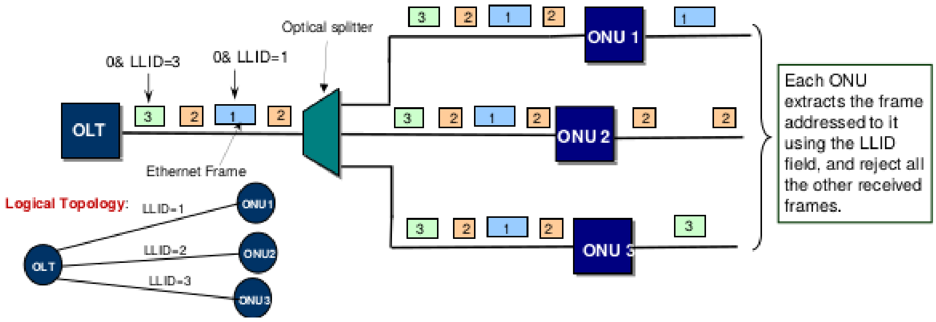
The SLD (*Start* LLID *delimiter*) is delimited in both sides with 2 bytes with the same pattern as in the 802.3 preamble frame. The next item is the LLID and the CRC field for error detection in SLD and LLID.

This way, is possible to identify to whom the frame is for. These frames are sent from the OLT to the spliter and only at the ONU the frames are filtered for the correct user.

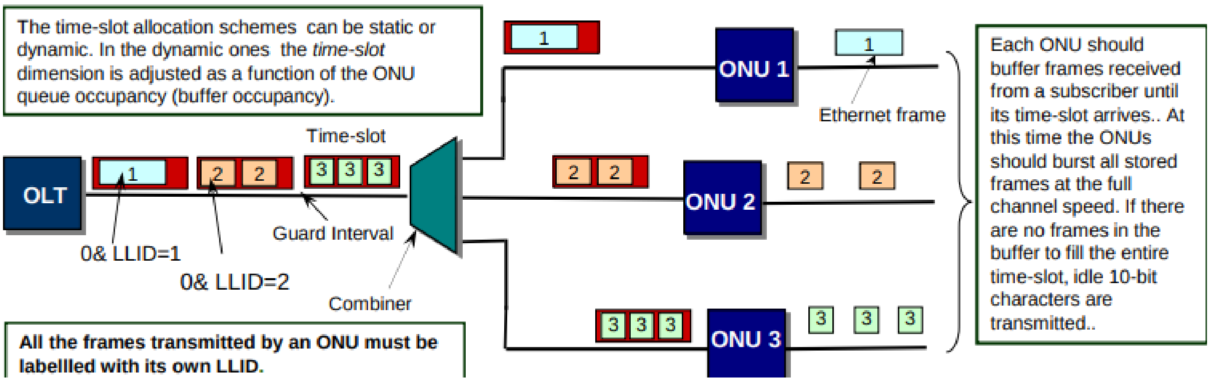
The LLID is the field responsible for this, is used for the extraction.

Downstream:

- **All the frames are broadcasted by the OLT to all ONUs, which extract the frames addressed to them using the LLID field. Note that the LLID field is only present inside the EPON. The LLID is removed by the ONU before delivering the frames do the clients.**



Upstream:



EPON has a symmetric rate (down/upstream) of 1.25 Gbaud which results in a 1Gib/s bit rate when using the 8B10B coding scheme.

To upstream, each ONU is allocated a time-slot and several ethernet frames fit in that timeslot. A complete time-slot cycle over all ONUs has 1 ms. And the guard interval has 1.5 microseconds. Therefore, it depends on how many ONUs are sharing an OLT. Typically, 16 users share a OLT. $1ms = 16\times$(ethernet-frame + guard-interval). Thus, one gets 61 microseconds of ethernet frame duration, which, at 1Gbit/s are 61 000 ethernet frames. Of course, the efficiency is not 100% due to error correction codes mainly.

A few final notes about EPON frame format:

- There are LLIDs which stand for Logical Link Identifiers. From the name is possible to understand that the links are logical. They not only identify each client but also each service. Therefore, over the same physical link going from an OLT to a certain client's ONU there can be 3 logical links connecting the service in the client's home to the service provider (tv, voice or data).
- The ONU knows what LLID's came from him and rejects all others; But, how about the VLANs? Does the ONU map between VLANs and LLIDs? **YES** . If there are no Ethernet frames, the ONU transmits only 10 meaningless bytes? **OLT Functions in EPON**

  Discovery Process: Verifies if any ONU has joined or left the network. The OLT periodically broadcast Discovery messages to all the ONUs. When powered on or reset the ONUs waits for such message, and responds to it.

  Registration Control: Controls the registration of newly joined ONUs. The registration process is carried out through a series of Discovery messages between the OLT and the ONU (handshaking). At the end of the process it is assigned a unique LLID to the ONU.

  Bandwidth assignment: Assigns to each ONU an adequate upstream transmission bandwidth based on its transmission requirements (buffers/queues occupancy).

  Ranging and synchronism: The ONUs need to be synchronized (clock) between them and to the OLT in order to transmit in a specific time-slot. The timing is done through an exchange of synchronization messages between the OLT and the ONUs and require the calculation of the transmission time delay (round trip delay) between them.

  Depending on the buffer occupancy, the frames can be longer or shorter. The ONU reports this occupancy with... (guess what) a **report** . The OLT sends a **gate** to each ONU and this is the bandwidth assignment process .

The control plane is the responsible for this and all other OLT functions described above. Tip: to have a complete answer regarding the control plane,

- The duration of each time slot is called the transmission window and this window size is what will be attributed to an ONU. The windows size is the number of bytes that can be transmitted. Therefore, to get the slot duration, one simply does $\frac{8\times \text{Windown}}{\text{Datarate}}$. Additionally, there's a guard interval.
- It is called a cycle time for the set of N time-slots. N corresponds to the number of ONUs. For EPON, this value is typically 16. **The number of time slots is equal to the number of ONUs connected to the OLT because in a cycle everyone needs the oportunity to transmit** . A cycle time is nothing more than the sum of each time slot duration + guard time:

$$T_{\max} = \sum_{i=1}^{N}\left(\tau_g + \frac{8\times W_{i,\max}}{D}\right)$$

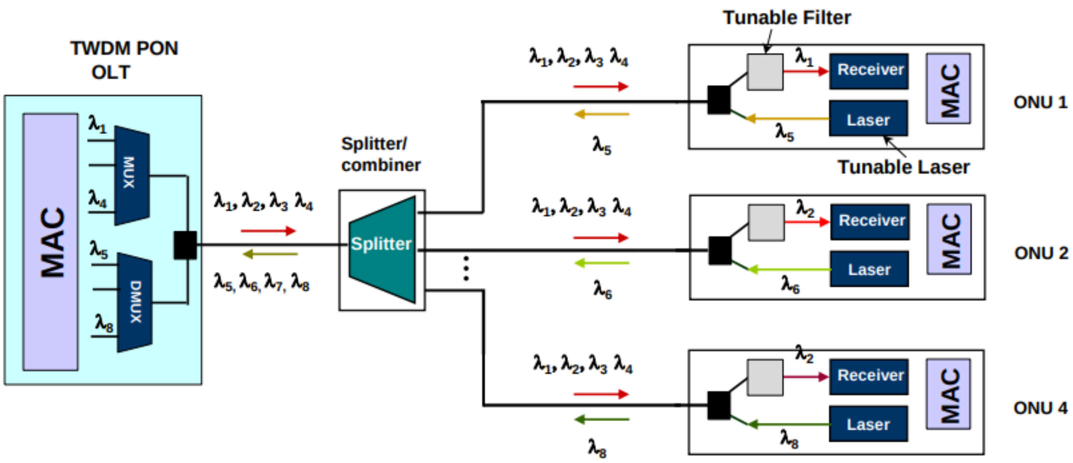
GPON

GPON has around 2.5 downstream and 1.25 upstream.

slides 297 and 298 mainly.

Frame summary: slide 300

XGPON through faster electronic is able to increase the bitrates per wavelength to 10 Gbit/s. TWDM-PON does 4×10 downstream and 4×2.5 upstream.



There are two main limitations with increasing the bit rate in optical fibre. Problems with fast electronics and dispersion. The narrower the pulse, the more bandwidth it takes and thus the higher the dispersion. Then it starts to get really expensive and this is only access..

Network operators assume only about 40 % of the users use the resources at the same time.

In GPON, the bottom two layers are GTC and PHY while in the EPON was MAC and PHY.

- **GPON stands for Gigabit Passive Optical Networks and is defined by ITU-T (Recommendations G.984.1 through G.984.6.). It supports triple play services, high bandwidth (up to 2.5 Gb/s), long reach (up to 20 km), etc.**

- **GPON can transport Ethernet, TDM and ATM traffic. All the traffic is carried directly in the payload of the GEM (GPON Encapsulation Method) frames. These frames are a slight modification of the GFP frames (see slide 187).**

- **Each GEM frame (or segment) includes a field called Port-ID (P-ID), which is used by each ONU to filter (select) the incoming frames.**

- **The GPON TC (Transmission Convergence) (GTC) layer is equivalent to the data link layer of EPON. It specifies GPON frame format, medium access control protocol, OAM function, etc.**

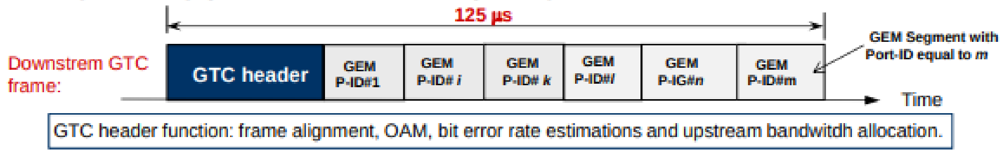
GPON Encapsulation Methods is one of the functionalities of GPON Transmission Convergence Layer in the GPON network.

GEM header has several functionalities like having the FAS (Frame Alignment Sequence)

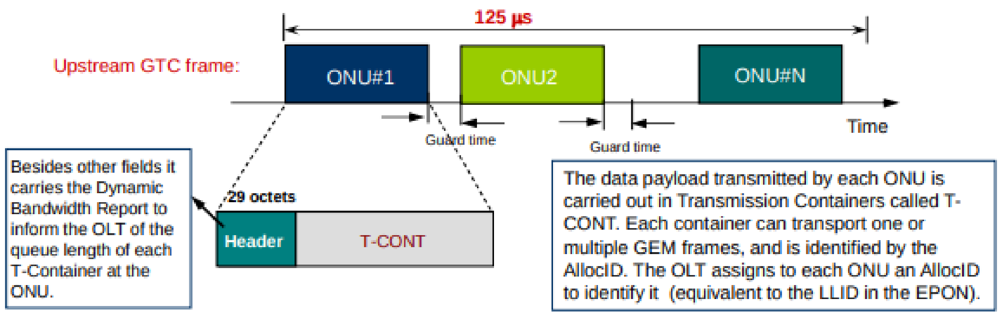
P-ID in the GEM header, payload is an ethernet frame

While in EPON the downstream ethernet frames are transmitted *raw*, in GPON networks the P-ID is

- **The downstream GTC frames have a duration of 125 μs and is 38880 bytes long (bit rate 2.48832 Gb/s). They are broadcasted to all ONUs.**



- **The upstream GTC frames have also a duration of 125 μs and is 19440 bytes long (bit rate 1.2446 Gb/s). Uses TDMA .**



Upstream, the ethernet frames are placed inside GEM frames and those go into a T-container (transmission container). Inside the GEM frames is possible to transport different types of frames : Ethernet, ATM, etc.. Depending on the required Qos of the traffic, different T-containers can be used: Different T-containers are meat to separate traffic types by the QoS (VoIP is different than IPTV). Each T-container has a **alloc-ID** . The alloc-ID tells the OLT from where the T-container came from. There can be multiple frames of that traffic in each T-container.

Only in the upstream, T-containers are used. Both is upstream and downstream, GTC frames are used. however in upstream they have many headers + t containers + guardband while in the downstream they simple are a bunch of GEMs together that each ONU will remove the one that has it's P-ID.

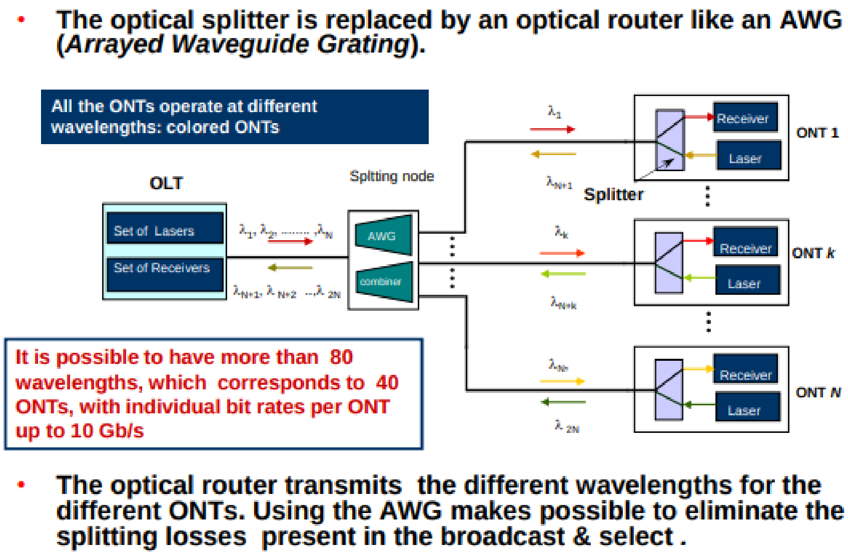
In the EPON, the dynamic bandwidth allocation Gate and Report messages are used and this reduces the efficiency of the network. In GPON these messages are included in the header GTC, there aren't specific separated messages simply for these reports.

Splitting ratio in the number of ONUs connected to a OLT.

1.6.5 XGPON, TWDM-PON and WDM-PON

TWDM-PON is able to serve much higher bitrates but to far fewer users since they need $2N$ wavelengths to serve up and downstream N users.

WDM-PONs are GOOD because are very future proof. If each user has it's wavelength (and each wavelength has a bandwidth of 50 GHz) it is possible to increase the bitrates until something like 40Gbit per user, which is still slightly out of range of today's technology. With this network structure is possible to increase everyone's bitrates by simply changing the OLT.



1.6.6 todo