Network Infrastructures 2018/2019

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Contents

1 General Introduction

1.1 Network Components

A Network Infrastructure is composed by the following main components:

- **end systems / terminals / hosts / ...**: used by human users, but even machine users (case of IoT environments), they are input/output devices that use network in order to transmit or receive data. In some sense we can say they are the ones which generate the data that has to be transmitted
- **network elements**: take care of routing the data transmitted over the networks (routers, hubs, switches...)
- **links**: physical mediums in which data is physically transmitted

1.2 Networks classifications

All of us know that the Internet Network is a Network of Networks. These networks can be classified according to difference metrics. Here follow some classifications.

1.2.1 WAN, MAN, LAN

This first classification takes into account the *coverage distance* between network components. According to this classification, we can have:

- **WAN (Wide Area Network)**: covers a very large area (order or millions/billions components)
- **MAN (Metropolitan Area Network)**: covers a metropolitan area (order or thousands components)
- **LAN: (Local Area Network)**: networks that cover short areas, for instance a building, an house and so on (order of tons, rarely hundreds components)

1.2.2 Virtual Networks and Private Networks

This is not actually a classification, but identifies two properties a network can have. We say that a network is:
Private when users which use the network are controlled, the network relies on firewalls and other security measures and network is not directly accessible from outside.
Virtual when given a physical network with routers and links I can build on top of it multiple virtual networks, for instance one virtual network uses a subset of routers of the physical one while another one uses another subset of routers (the two virtual network can also share some routers). An instance is when a client and a server build a connection with the TCP 3 ways handshake, after which client and server are connected through a virtual network that resides on top of a physical network.

1.2.3 Client/Server vs Peer-to-Peer

The third classification is the following:
Client/Server model: some hosts are clients, other are servers, and clients make requests to servers which respond with requested data.
PeerToPeer (P2P) model: the role of each host is unbalanced, they are all at the same level, there is no client nor server here. Of course some hosts can store more informations than other ones, but it is not a semantic difference. What happens here is that each peer has some data,

and if a peer needs a specific data it asks to other peers, so even if I have a peer with more data than others, it does not mean it is a Server because it can be that this peer lacks of some data that another peer can have.

1.2.4 Communication mediums

Another classification can be done on the mediums the network uses to interconnect hosts. A first classification here can be made on whether the media is a *wired* media or a *wireless* one. Then each of the twos can be further differentiated (for example there are different kinds of wired mediums like *Twisted-pair wires*, *Coaxial cables*, *Fiber optics*, ...).

1.3 WAN, MAN, LAN, PAN, and Global Wireless Standards

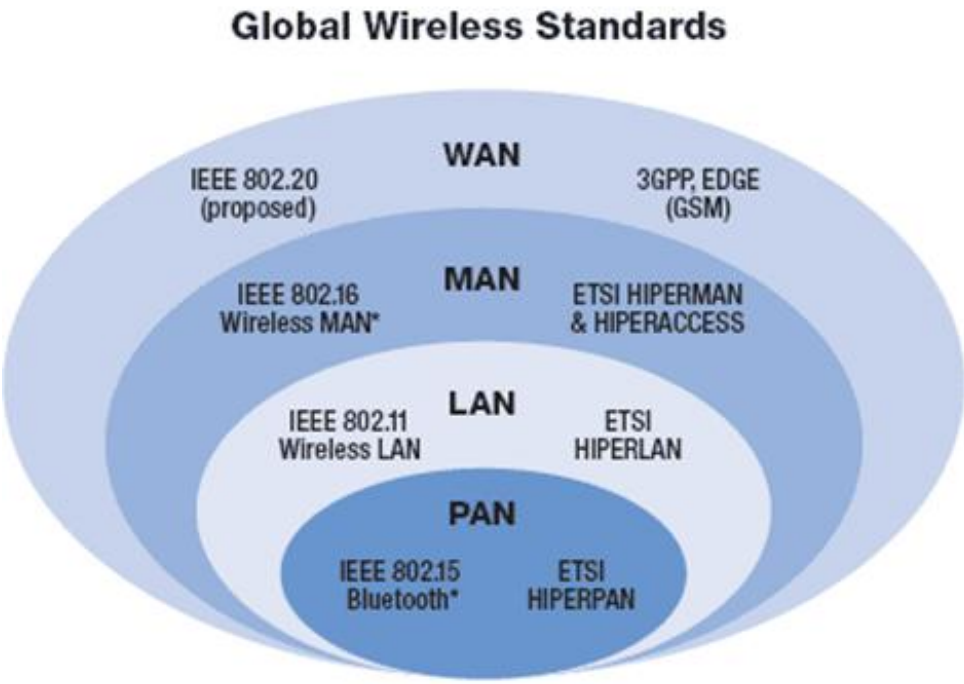


Figure 1: Wireless Standards

All these kind of networks have their own specific standard, which are *Wireless* standards belonging on the **IEEE 802** family of wireless standards, in particular:

- The PAN networks (Personal Area Network) rely on the **IEEE 802.15** standard (a.k.a. *bluetooth*), so they are the wireless networks that cover the shortest range.
- The LAN networks (Local Area Network) instead rely on the **IEEE 802.11** standard (a.k.a. Wireless Lan or *WiFi*), in particular LAN are handled by the *Ethernet* protocol and IEEE 802.11 is the implementation of Ethernet in Wireless LANs.
- The MAN networks (Metropolitan Area Network) rely on the **IEEE 802.16** standard which is able to cover a wider range w.r.t. IEEE 802.11.
- The WAN networks (Wide Area Network) rely on the **IEEE 802.20** standard, where here we can find 3GPP, EDGE, LTE and so on. These technologies are the ones implemented in the Cellular Infrastructure, and indeed the WAN networks have the widest coverage range.

1.4 Telecommunication Processors

At the beginning of this section we said that the networks are composed by three main elements. Here we are going to talk about the network elements. We have these main network processors:

- **Modem**: it interfaces a private computer network (a LAN one) to the telephone line (ADSL). Its main purpose is to convert the digital signals coming from the hosts of the LAN into analog signals which can be transmitted onto the telephone line and vice-versa. So they basically perform the **digital modulation**, meaning to modulate an analog signal so that it can carry informations brought by the digital signal. It also performs the **digital demodulation**, meaning to extract the information from an analog signal that has been modulated to carry digital informations and to build a digital signal from the extracted informations. So we can say that:
  Digital Modulation: digital→analog
  Digital Demodulation: analogl→digital

- **Hub**: the simplest ones, it is a device which is oly able to replicate the data received on a port to all other ports, it is also used to regenerate the signal. It can be used in *star* topologies for example since in such topologies hosts are connected to a central network processor.

- **Switch**: like **hub** but it is able to select the output port towards which to retransmit the data received from a port, by means of the *switching table*. They are used a lot in LAN networks (again used in star topologies but not only).

- **Router**: more than a switch, because they can do what switch do but they perform the port selection depending on the network topology (it has tables that say where to forward data and these tables are filled for example with Dijkstra algorithm).

- **Gateway**: more than routers, they can also have features like firewall, security and so on.

1.5 Network Topologies

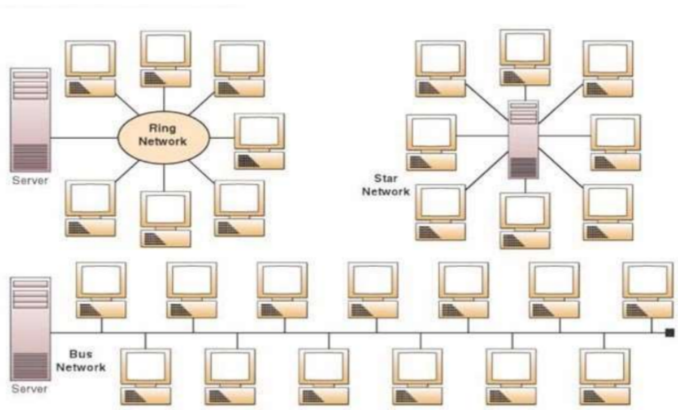


Figure 2: Bus, Ring and Star topologies

LANs were first born as **Bus** topologies, where there is a straight bus to which hosts are linked, each of which has a certain distance from the Network Processor (Server, Router, Switch, ...). After a while the **Ring** topology was introduced while the newest entry was the **Star** topology. Star topology is also used in MAN networks. For what concerns WAN networks, if we consider cellular lines again Star topology is used (we understand the Star topology is the most modern network topology). Apart the shape, all these tolopogies have different performances and offer different deatures. Which is the best? It depends! For instance the Ring topology is used in MAN networks because it is more reliable than the Bus topology. However when we will study network technologies we will see which technology uses which topology and why. However, from an user perspective, the Star topology offers the best performance (each Client is directly connected to the Server, while in the other topologies users share the same link so collisions are quite common), but is the most costly one.

1.6 Network Protocols

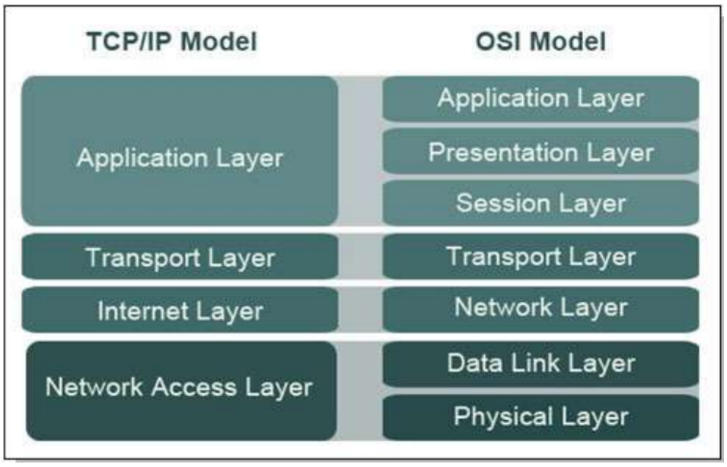


Figure 3: ISO/OSI vs TCP/IP

We know that there are two main different *Protocol Stacks* that characterize the network layers: **TCP/IP Model** and **ISO/OSI Model**.
The TCP/IP model only has 4 layers (*Application*, *Transport*, *Internet*, *Network Access*) while ISO/OSI model describes 7 layers (*Application*, *Presentation*, *Session*, *Transport*, *Network*, *Data Link*, *Physical*). We won't go into the details of all these layers, but few words have to be spent. Each layer performs a specific task, and is implemented by specific protocols. The common pattern is that lower level layers offer services to higher level layers (Network Layer allows Transport Layer to send its datagrams over the Internet, Transport Layer allows Application Layer to bundle application data into segments that will be transported over the internet, and so on and so forth).

2 Transmission Fundamentals

2.1 Analog and Digital Signals

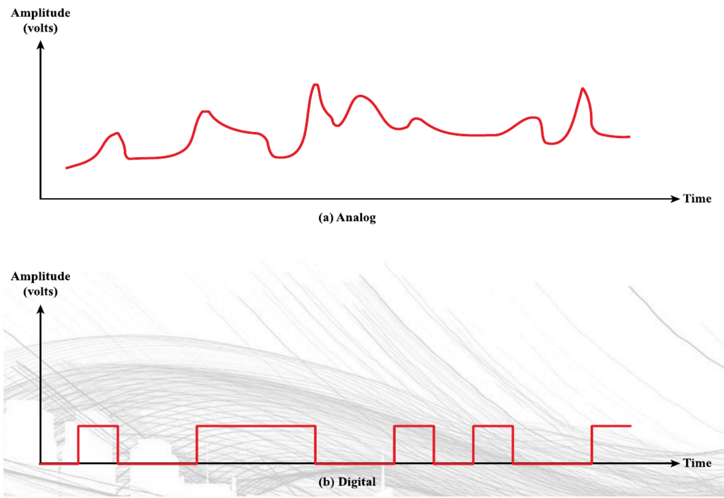


Figure 4: Analog Signal and Digital Signal

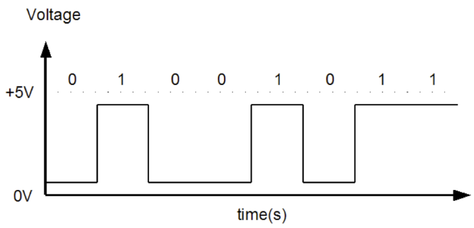
An **Analog Signal** has a continuous shape, at every instant t it has a certain amplitude. Since an analog signal is continuous, the range of values its amplitude can have is unbounde, because it is a real value.
A **Digital Signal** at each instant t has a certain amplitude belonging to a certain (tipically) finite set of amplitudes and it is not continuous, it is somewhat discrete. The possible amplitude values of a Digital Signal are the so called "levels" of a Digital Signal.

2.2 How data is transmitted: Modulation

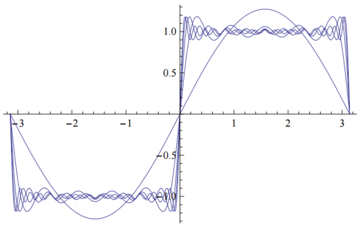
When we talk about an Analog Signal or a Digital Signal, we are referring to a **Source Signal**, the one that contains the information we want to transmit. The most common way to transmit a Source Signal, whatever it is its nature (Digital or Analog), is through an Analog Signal called **Carrier Wave**, since the transmission channels are most commonly Analog Channels (*Copper-Wires, Coaxial Cables, Optical Fibers, ...*).
So what it is done in order to transmit the Source Signal onto a channel is to put it onto the Carrier Wave through a technique called **Modulation**. Indeed, *a Carrier Wave is nothing but a periodic signal with constant frequency and amplitude*, basically a *Sinusoid*, and due to its constant nature it does not carry any information at all on its own, the information in general is given by the variation of a signal in terms of frequency or amplitude or phase, and indeed *the Modulation is the technique of making the Carrier Wave to vary in order to make it to carry the information.*
There exist different kinds of modulation: **FM** (Frequency Modulation), **AM** (Amplitude Modulation) and so on, we will not go into the details. So, the Modulation is not in general the digital→analog conversion (like one can wrongly say), but it is the technique just described.
It is commonly said that the Source Signal, which is called Modulating Signal, "modulates" the Carrier Wave.
Finally, when an analog signal has to be transmitted, a typical fashion of the present era is to first encode it into a digital signal before being transmitted, then the digital signal is modulated in order to be transmitted onto the analog channels. The reason is simple, a digital signal is much resilient to noise w.r.t. an analog one, i.e. it is simpler to reconstruct a digital signal rather than an analog one.

2.3 How Digital Signals are made

We saw that in general I have to send a Digital Signal because it is the modern standard due to its robustness w.r.t. errors. Our Digital Signal is a composition of infinite sinusoids, indeed the composition of that infinite sinusoids give to the Digital Signal that particular perfect rectangular shape.



(a) Digital Signal shape


(b) Digital Signal sinusoid components

This was just a simple digital signal, but however we know that a Digital Signal is composed of different levels rather than just two.
However, a channel has not an infinite bandwidth, so I cannot send the digital signal perfectly shaped in the channel, the channel will cut certain frequencies of the signal. So at the end the signal that passes through the channel has a shape similar to the rectangular one, but it will be somewhat distorced.

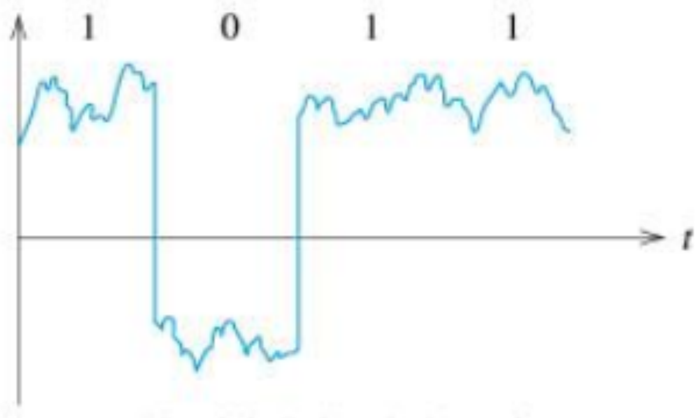


Figure 6: Signal that passes through the channel

2.4 Bandwidth and Datarate

The case just shown is the simple case where the digital signal has only two levels. In general, my digital signal has a certain number of levels, and when I transmit it if the channel has a low bandwidth it will cut too many frequencies and the signal gets distorced too much, and so the receiver will make several errors in reconstructing the original signal.
So at the end the higher is the channel bandwidth, the lower my signal gets distorced, the higher will be its precision. This results in the following: if I have an high bandwidth signal, I can shape my Digital Signal in many levels, beeing sure that the error rate in reconstructing the signal will be very low, and so the higher the bandwidth, the higher the levels my signal can have, and the higher the levels the higher the bits per symbol I can send. That's why the higher the channel bandwidth the higher is the datarate the channel can handle.

2.5 Relationship between DataRate and Bandwidth: Nyquist and Shannon formulas

It is not only the channel bandwidth which affects the datarate, but also the noise of the channel. The following picture shows what happens in case of noise:

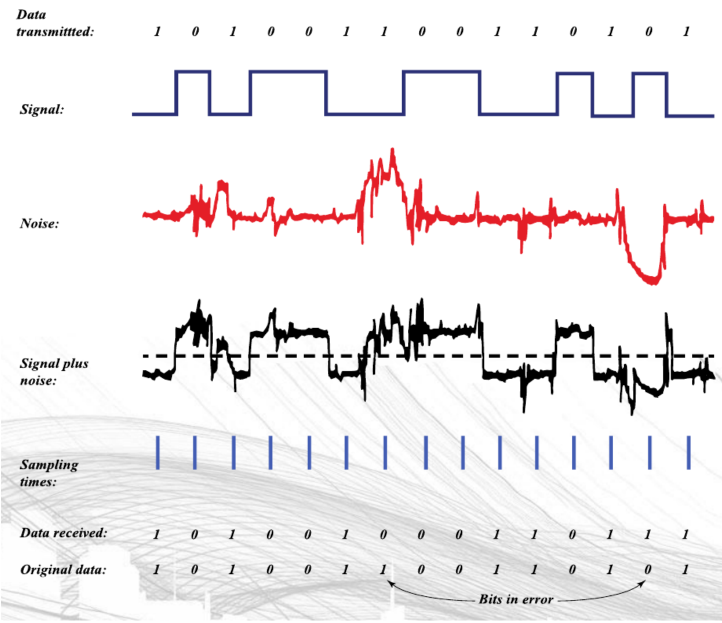


Figure 7: Signal that passes through the channel

Think of what we said about the bandwidth of the channel and how it distorces the perfectly shaped digital signal. Now add the distorsion caused by the noisy channel. What happens is that the bandwidth of the channel and the noise of the channel strongly affect the shape of my signal. At the end the datarate of my transmission is not directly limited by the channel bandwidth and/or the high noise of the channel, meaning that I can decide to send my signal at an high datarate whatever is the bandwidth of the channel and/or the noise, but what happens is that then my signal gets distorced depending on the two factors just described.

So at the end, before I transmit I have to decide the shape of my signal in a way suitable for the particular channel. *Nyquist* and *Shannon* come into play here.

Nyquist Formula Given an ideal channel, Nyquist tells us that the maximum capacity of that channel is

$$C = 2B \log_2 M$$

where B is the channel bandwidth and M is the levels of the signal to be transmitted. So it uses the levels of my signal and the bandwidth of the channel to tell what is the maximum theoretical DataRate (in $bits/s$) of the channel.

Now, one can say that even if I have a channel with a limited bandwidth B, it is sufficient to set M very high to have a very high datarate. This is not the case, indeed we saw before that the higher the levels, the harder the receiver is able to distinguish them (due to the limited bandwidth of the channel). So at the end there no exist ideal channels in real life, and so I cannot simply set M very high.

In order to understand how to shape the signal, we have to make a further step.

Shannon Formula

$$C = B \log_2(1 + SNR)$$

In this formula, B is again the bandwidth of the channel, SNR is the *signal-to noise ratio* which is a characteristic of the channel expressed in dB, and C is again the capacity of the channel in $bits/s$. Note that here there is no indication of the signal level, which means that no matter how many levels we have, we cannot achieve a data rate higher than the capacity of the channel. In other words, the formula defines a characteristic of the channel, not the method of transmission.

How can I practically use these two formulas in order to shape my signal?
Suppose I have to send data, I have to compose a digital signal suitable for the channel. With the Shannon Capacity formula I can get C, the maximum bitrate achievable by the channel, then I substitute it into the C of the Nyquist formula in order to get the number of levels M used to shape my signal.

N.B. !!

2.6 WiFi 2.4GHz channels

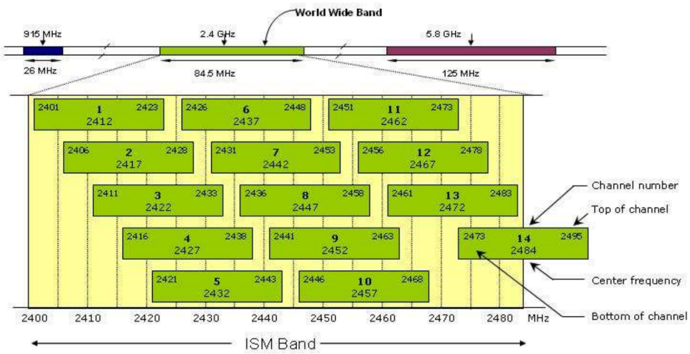


Figure 8: Signal that passes through the channel

Wifi has its *bandwidth* which is of **84.5 MHz** centered into the **2.4 GHz** *carrier frequency*, as we can see. This bandwidth is divided into 14 different channels (some of theme can overlap), each of which has its *carrier frequency* centered in the middle of the channel (for example the channel 7 has a range of frequencies from 2431 MHz to 2453 MHz and the carrier frequency is centered on 2442 MHz). Each channel has a bandwidth of 22 MHz. On each raw there are channels which do not overlap (for example 1, 6, 11).
At the end, what happens is that even if the whole available bandwidth of the WiFi standard is **84.5 MHz**, only one channel at a time can be used, so the effective bandwidth of the WiFi is **22 MHz**.
The next spectrum band is the red one, which has a bandwidth of **125 MHz** centered into the **5.8 GHz** *carrier frequency*.

3 Infrastructure Functional Areas

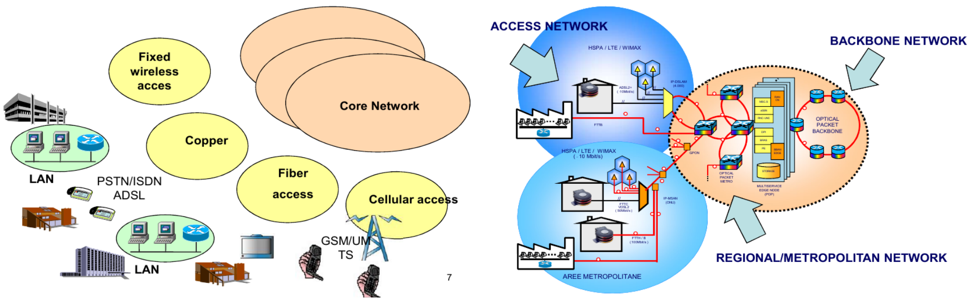
We can distinguish 3 main areas of a Network Infrastructures:

- **Access Network**: the part which connects users (in this environment we call them *subscribers*) to their service providers

- **Core Network**: it is the Backbone, which provides any-to-any connections among devices of the network (switches, routers, ...). Usually the Core Network exchanges big amounts of data.

- **Edge of the network**: it stands on the boundary between the previous two areas, providing intelligent functions which are not performed inside the Core Network. For example it classifies subscribers, specifying a behaviour depending on the class of the subscriber.

3.1 Access / Core Network



(a) Access/Core (b) TelecomItalia

Figure a: they are displayed the technologies used into the Access Network, so the technologies which allow the subscribers to reach the Core Network.
Figure b: schematic representation of a network like the one provided by Telecom Italia. We recognize here the *Core Network* (orange one) and the *Access Network* (the blue one). The border between the twos is where the *Edge Networ* stands. The red lines represent fiber optics. In the Core Network, fiber optics are multicolor fiber optics (the rainbow under the routers represent the fact that they use multicolor fibers, carrying different wave lengths). Why we need the multicolor fibers? Because we multiply data flows in the wavelength domain and we have fiber where we can do this multiplexing. So multicolor = high bitrate. Notice that the multicolor fiber is one of the most expensive technologies. On the countrary, going towards the edge and the end users we still find some fiber optics, but also some black lines and in this case they represent copper based medias. Finally the exagonal shape is used to represent a cellular system, where the yellow triangle represent the base station.

3.2 Distribution Network

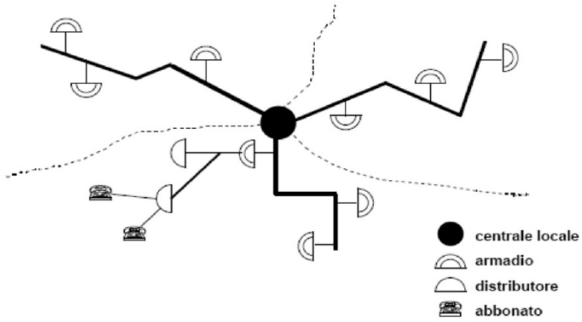


Figure 10: Distribution Network

Let's zoom into the *Access Network*. In the Access Network we find the **Distribution Network** which is the capillar part of the Access Network. The picture shows the typical scenario of the Telephone Network: the main entity is the **Central Office** (C.O., the black circle), and from that C.O. many cables come out, reaching the **Cabinets**, then the **Curbs** and finally the subscribers.
The typical scenario we can imagine is that my neighborood is attached to a curb, then multiple curbs are connected to a cab placed for example in a main street, and finally multiple cabs are connected to the C.O. This is the way the C.O. provides the network access to the subscribers.

3.3 FTTx = Fiber-to-the-x

It is a way to define how a fiber interconnects the subscriber in the Distribution Network. We can have different cases:

- FTTH = Home
- FTTC = Curb
- FTTN = Node or Neighborood
- FTTP = Premise
- FTTB = Building or Business
- FTTU = User
- FTTZ = Zone
- FTTO = Office
- FTTD = Desk

3.4 FTTE, FTTCab, FTTCurb

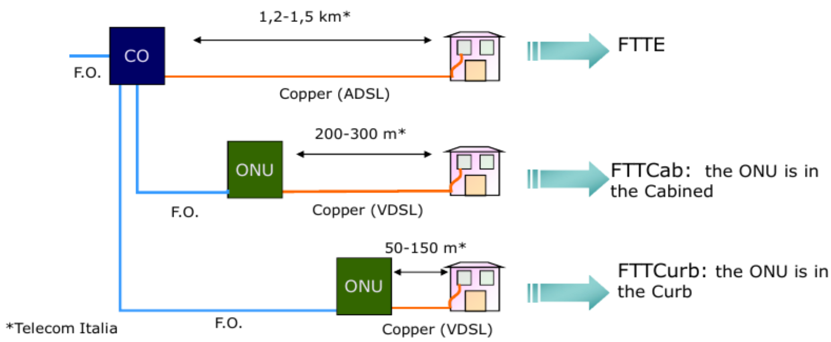


Figure 11: Fibert to the Exchange

FTTE stands for *Fiber to the Exchange* and it is basically the *Fiber to the Central Office* (meaning that Exchange and Central Office are synonimous). As the name says, for this technology the fiber cables reach the C.O. (the blue one in the picture) and then the subscribers (houses/offices) are connected to the C.O. with Copper Wires (the common ADSL we will study). Then there are two other solutions:

- **FTTCab**: in this case the **ONU** (the element which takes care of managing the fiber signals) is placed in the Cabinet, so in this case the fiber reaches the Cabinet.
- **FTTCurb**: same as FTTCab, but here the **ONU** is placed into the *Curb*, so the fiber reaches the *Curb*.

In these two latter cases, since the ONU is closer to the subscribers w.r.t. FTTE, we have an highest connection speed, indeed as the picture shows in case of FTTCab or FTTCurb we have VDSL (which is better than ADSL) while for the case of FTTE we have ADSL. In the picture they are displayed the common distances between subscribers and Central Office, subscribers and Cabs, subscribers and Curbs.

3.5 FTTP: FTTB-FTTH

Another cathegory of FTTx is **FTTP** which stands for *Fiber to the Premises*. To this cathegory the following FTTx belong:

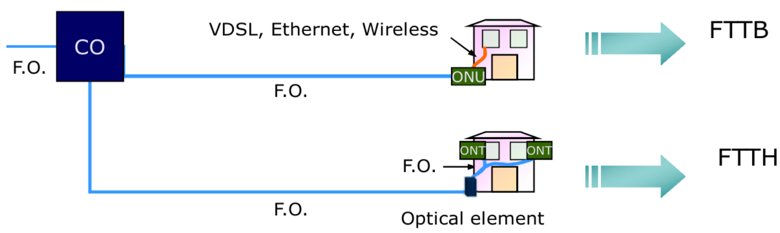


Figure 12: Fibert to the Premises

- **FTTB**: the fiber reaches the building, where building is tipically an office, a school.
- **FTTH**: the fiber reaches the home of the subscriber.

Indeed for what we see in the picture the **ONU** components are inside the *Premises*.

3.6 2G-3G Architecture

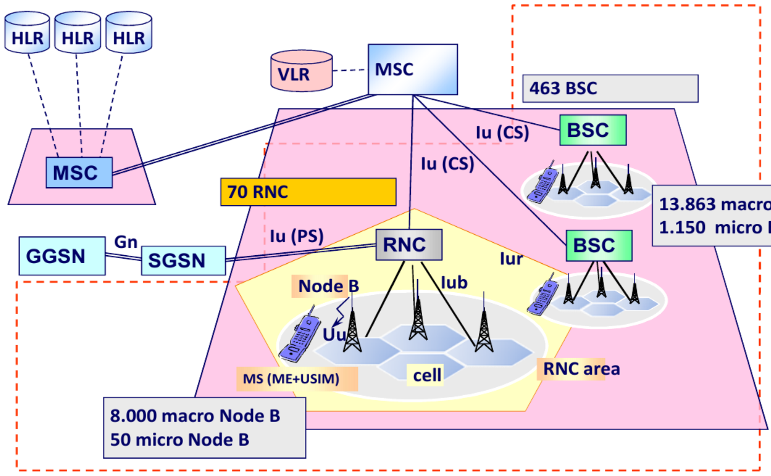


Figure 13: Fibert to the Cellular Network Infrastructure for 2G and 3G technology

As we can see, the cellular infrastructure is quite complex. In the picture, the rectangular boxes are the names of the network components of a cellular infrastructure. Each of these network components has its purpose, and we do not find many of them in other wireless infrastructures because there is no mobility in those infrastructures.
The first element we describe is the **Base Station**, which has a different name depending on the generation of the cellular infrastructure, the names of the 2G-3G infrastructure are **BTS** (*Base Transceiver Station*), in the 4G infrastructure for instance the name is *eNodeB*. These base stations have an antenna and they are interconnected to the **BSC** (*Base Station Controller*), so a BSC controls a pull of Base Stations.
Then there is the **MSC** (*Mobile Switching Center*) and this entity is fully dedicated to manage the mobility of the users, and this is the component that is not present in other wireless infrastructure that do not provide any mobility. In order to manage mobility, the MSC relies on two network components:

- **HLR**: *Home Location Register*, a database where the users of the network (in particular of a specific network provider such as Tim, Vodafone, Wind...) are registered in their official way (for example using a DNS).
- **VLR**: *Visitor Location Register*, a register where during my mobility (I change place) it stores my temporary identity (that is related to my position).

So I have two identities, the official one which is stored in an HLR and a temporary one stored in a VLR. For instance when I receive a call this call is routed to the MSC where my HLR is, and thanks to the relation between the MSC where my HLR is and the MSC where my VLR is (there are protocols to do this) it is possible to find my smartphone in order to route the call to me. Also the *Roaming* access is managed thanks to the MSCs.
Another controller is the **RNC** one, *Radio Network Controller*, which is specifically oriented to manage the resource assignment in the network (how much bandwidth to assign to users).
Finally the elements **GGSN-SGSN** have the purpose of managing the packet data network.
Consider that up to 2G the cellular infrastructure was designed to transfer only voice calls, the SMS for example were not standardized, the story is that in the network many messages are exchanged, but these are functional messages rather than user data messages, meaning that these messages have the purpose to make the infrastructure to work, now in the 2G standard some fields of the overhead were not standardized and so it was thought to use them to insert

SMS text, and this is also the reason why the SMS had a fixed length.

As we want to increase the datarate in a cellular infrastructure, we have to decrease the size of the cells (a cellular network is made of several cells each one managed by a Base Station). This because reducing the cell size, the number of users that the Base Station manages is lower, so each user gets more bandwidth. However decreasing cell sizes means to have more Base Stations and this means the cost is higher (for each Base Station we need to pay for the Base Station itself, for the cables to interconnect the Base Station and for the energy to be supplied to each Base Station).
Tipically in an Urban Area since the population density is high the cells are smaller so we have more cells, while in a Rural Area the cells are bigger so we have less cells.

The **Backhaul** of the network is a sort of edge between the cellular network and the rest of the network, and we have to take also it into account because if I have a lower bitrate there than it is my Network Bottleneck.

3.7 Wireless Access

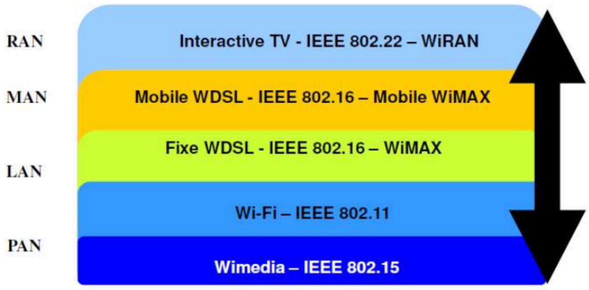
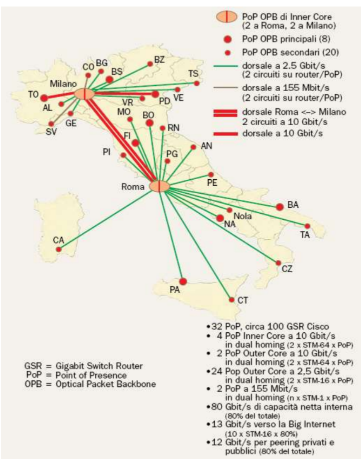


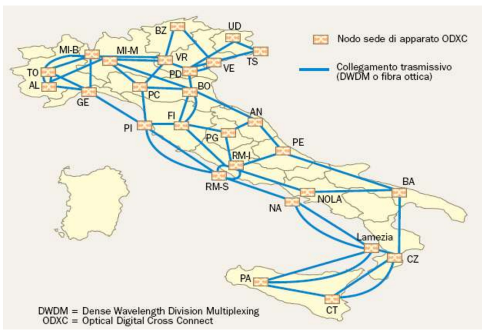
Figure 14: Standards for Wireless Networks

We have already discussed this topic, however here we see another family that is **RAN** implemented by the **IEEE 802.22** standard also called **WiRAN**. The RAN is based on the band of the TV, indeed TV uses low frequencies in order to cover wide areas / long distances and so RAN does it as well.

3.8 Telecom Italia Backbone



(a) Logical Tolopogy

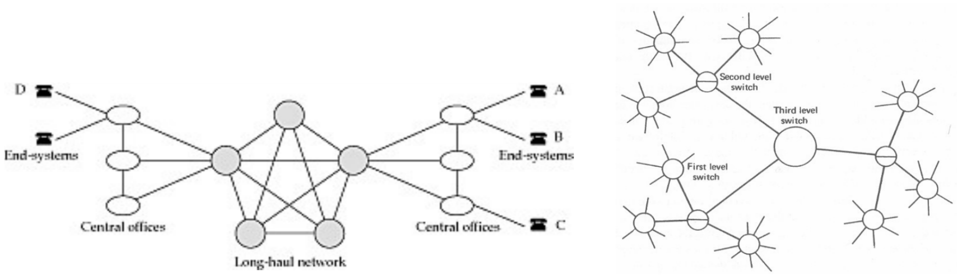


(b) Physical Topology

This is the backbone infrastructure (at an high level view) of the Network Infrastructure of Telecom Italia. We have all **PoP** (*Point of Presence*) entities here, which are of two cathegories: normal PoPs and *Inner Core* PoPs. In Italy we have 32 PoPs, two of which are Inner Core Pops, and they are one in *Milan* and one in *Rome*, and both are fully duplicated meaning that the Pop I have in *Rome* is duplicated in *Milan* and vice versa. As one can expect, an Inner Core PoP is a powerful PoP, it is a big point of presence. So watching the Logical Topology (figure a) what I want is to *logically* interconnect all the Settentrional PoPs to Milan and all Meridional PoPs to Rome, and how I do it is described in the Physical Topology, where I interconnect the various PoPs throught *fiber optics*.

4 The Telephone Network

Why we study the telephone network even if it is quite an old infrastructure? Because this infrastructure is at the base of the current ADSL infrastructure in some parts, which we will study later.
Telephone network was structured in an hierarchical structure, we have end systems, first level of Switching Centers (Central Offices), second level and so on. In particular there are three levels of Switching Centers, as represented in the following picture:



(a) Fully Connected Core

(b) Physical Topology

The reason why we have multiple Switching Levels is that the Inter Exchange Area the exchange is done by exchanging the data also with other operators, while in the very peripheral part we have to distinguish between two kinds of operators, the *Incombent* (the owner) and the *Competitive*. Basically the Incombent one rents infrastructure to the Competitive one which wants to offer that service to its users.

4.1 End Users and Central Offices

Originally in the telephone network we had a **Central Office (C.O.)** with a Star Topology towards the end users (each end user was connected individually with the C.O.). So each user did not share anything with other users due to the Star Topology, and this was good because there was no need of multiplexing (which is required when infrastructure has to be shared among users). The end users were connected to the C.O. in an Analog way, while the C.O. was implementing both Analog and Digital communication, the analog part was the one towards the users, the digital one was the one in the inner network. From this comes the fact that an important task of a C.O. was the analog/digital conversion and vice-versa.
Another aspect is that the users were originally connected to the C.O. by means of *twisted pairs*. Every user had 4 wires (two *twisted pairs*), one *twisted pair* was for transmitting and the other one for receiving. Then it was decided to use only one *twisted pair* per user both to transmit and receive.
Later what happened was that two *twisted pairs* of two users were merged into a single one called **Binder Group** (in practice two users were individually connected to a **Cab**, and the cab was connected to the C.O. by a single cable, so it is like this last cable is shared between the two users). In few words there were network portions that were shared. This was generating a **Cross-Talk** issue, because even if the users had individual cables, at some point of the network (Cab) their wires were grouped in one wire so two users were interfering one with the other.

So the **Cross-Talk** effect is generated by the interference of a cable towards another one, and what caused this effect were the two *twisted pairs* in the same *Binder Group*.
Moreover the users were using the same *twisted pair* both to transmit and receive, and this generated a "self-interference". This happened also if the user only transmitted, because there is the concept of **echo**, so the user received what transmitted. This echo can be cancelled by

having appropriate circuits in the telephone transmitter/receiver in order to cancel what I'm transmitting. The echo problem was mainly solved in the telephone network but was more present into the ADSL.
ADSL transformed the analog part of the network (the one that connected the users to the C.O.), into a digital one, and we will see this when we will be studying ADSL/XDSL.

4.2 Transmission Muxing

As we know, when I have an analog channel that transmits analog signal and I want to let him pass through a digital channel, I have to perform the analog → digital conversion. As we saw, in the telephone network we have end users analogically connected to the C.O. which was digitally connected to the rest of the network, so it was the C.O. to perform the conversion. The analog → digital conversion is simply made by the samplig method. The process is simple, at a constant frequence, so every 'tot' amount of time, I sample the analog signal and I associate the sampled value with a certain level, which is represented as a sequence of bits, meaning that each analog sample is translated to a sequence of bits. Depending on the accuracy I decide in the translation, the number of bits that represent a sample can be higher or lower.

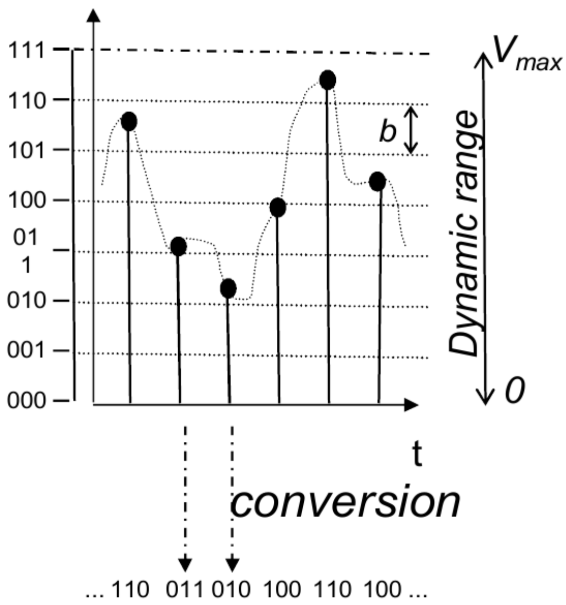


Figure 17

There is a law, the **μ law**, which states that the levels are not equally distant one with respect to its nearest ones. This law gave rise to a standard for which 256 levels were decided, and so each sample was represented by 8 bits. These levels were logarithmically spaced.
At that time, the standard was to translate the analog signal into the digital one sampling the analog signal every 125 μsec (fc=8Khz, Tc=125μsec), so since the sampling frequence was 8KHz and each sample was represented by 8 bits (256 levels), the converted digital signal had a data rate of 64 Kb/s.
Why this frequence of sampling? Because the **Nyquist Theorem** says that in order to keep the main frequency components (bandwidth) of the original signal we have to sample at a frequency that is at least two times the bandwidth of the original signal. Since the analog signal produced by the voice and sent to the C.O. had a bandwidth of 4KHz, we have to sample at a frequency of at least 8KHz.
So suming up I have 8bits per sample, and I sample every 125 microseconds, so 8000 times per second, so the sampling bitrate is 8000 * 8 bit = 64.000 bit/s = 64 Kb/s

At that period this was quite an high datarate for the digital communications, so it was decided not to transmit the voice signal but only the difference between the reconstructed voice and the syntetic one and so only 8 Kb/s.

4.3 Transmission Demuxing

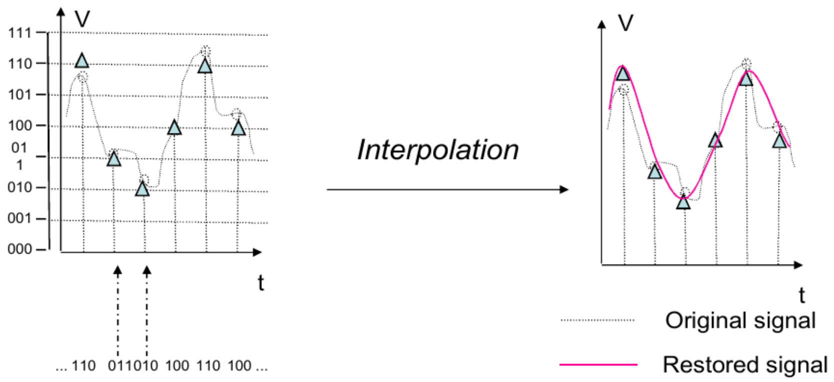


Figure 18

In the opposite way instead the C.O. received a digital signal, so bits, these bits were converted by means of the **Interpolation** into an analog signal that we call Restored Signal, which is of course not the same but similar to the original analog one.
Why similar and not the same? Because when the analog signal is converted into a digital one, some information is lost. The information is lost for some reasons. For example we have a limited number of digital levels, and when we sample we have to decide which level to associate to that analog sample, and there will be a gap (dinstance between circles and triangles in the picture) between the value of the analog signal and the level we decided, which is called *Quantization Error*, and consider that this error is higher when the number of levels are lower (less precision).

4.4 Multiplexing Techniques

Once the voice of the users has been converted to a digital form by the C.O., the digital signals are multiplexed in a unique cable. Here again we need a standard, and in the telephone network the standard decided to multiplex the digital signals in a **Time Division Multiplexing** (TDM), meaning to transmit the data of a single user in different time slots. As we know there are other Division Multiplexing Schemes, for example **Frequency Division Multiplexing** (FDM), that is mainly used in analog enviornments.

4.4.1 Time Division Multiplexing

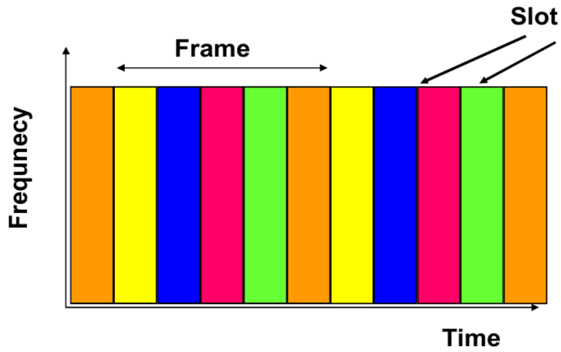


Figure 19

The Time Division Multiplexing is designed in a fashion such that there are different **time slots**, one slot per user, and so in a particular slot the information of that user are transmitted. A certain number of slots compose a **frame**, which is repeated time by time. So each user transmit its information into a slot of a frame, then in the next frame he has another slot and so on. Then all users share the same bandwidth, since they all have the same frequency since the division is made by time and not by frequency. One particular difference between TDM and FDM is that in TDM each user transmits only in a certain slice of the frame (its slot) but at the maximum bandwidth datarate, while in the FDM each user transmists contanstly but at a lower bandwidth datarate, that is the whole bandwidth datarate of the channel divided by the number of users for that channel.

4.4.2 Frame Structure in the PCM (Europe, E1)

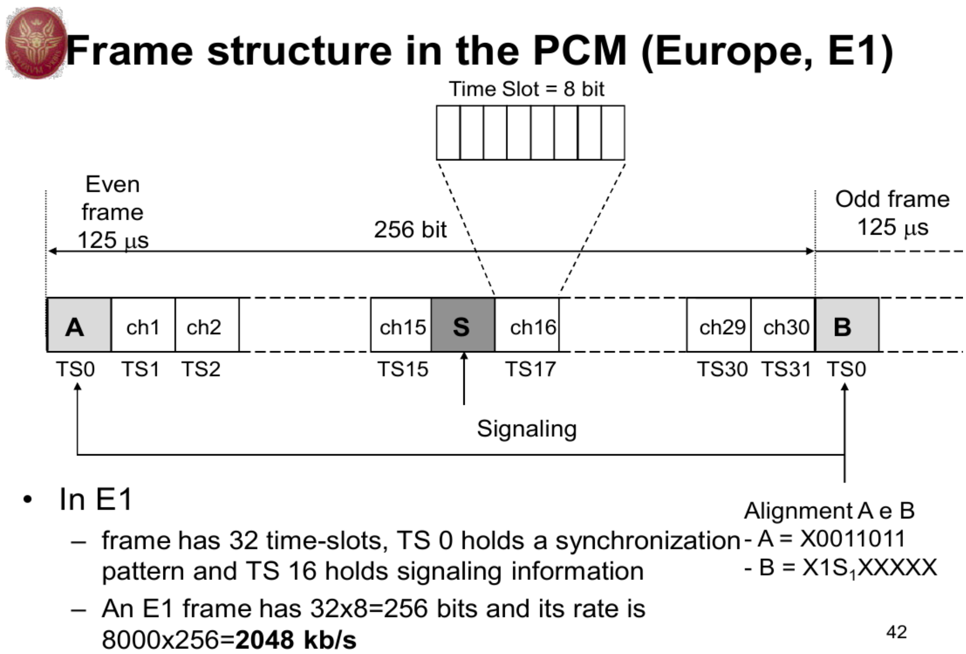
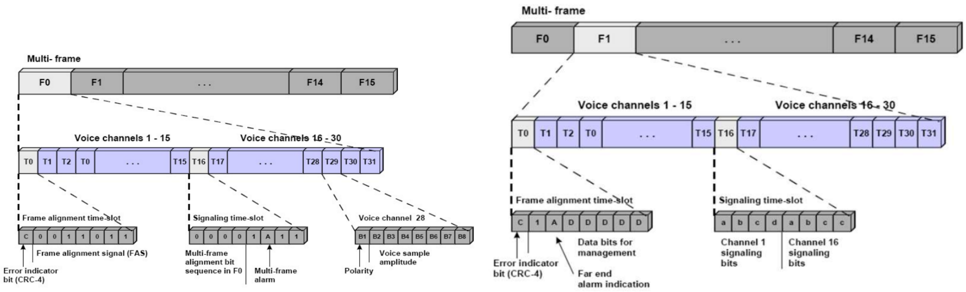


Figure 20

In the European telephone network the standard was **PCM (E1)** which was structured in the following way: the frame last 125 microseconds, and in each frame I have32 slots, each slot bringing 8 bit. Actually only 30 of 32 slots were dedicated to users (we wil see why). This means that in every frame $32 \times 8 = 256$ bits are transmitted, so the datarate is 256 bits per 125 microseconds and so 2.048 Kb/s = 2Mib/s. Making some calculation we get that every user transmits at a datarate of 64 Kb/s. This is the first stage of the multiplexing scheme, after the C.O. Also the multiplexing scheme is organized in an hierarchical manner, which means that I start with the PCM scheme, then I can multiplex togheter 4 PCMs, giving rise to the E2, so E2 brings 8Mib/s. There are further levels, up to E5, which has a datarate in the order of 500 Mib/s.



(a) E1 even frames (b) E1 odd frames

4.4.3 Frame Structure in the T1

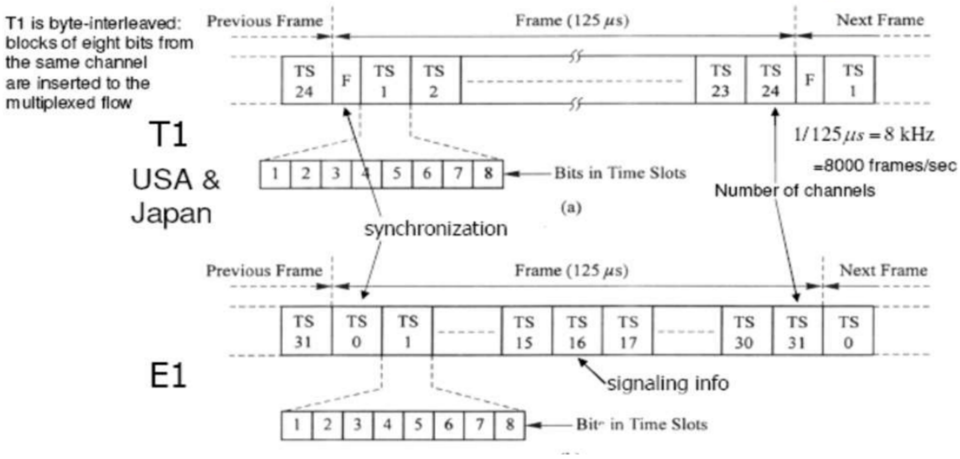


Figure 22

Other than E1, there is also T1 which was mainly used for the voice. The T1 was used in USA and Japan. Here the things are similar, each slot transmits 8 bits, and each frame lasts 125 microseconds. What changes is the number of slots per frame, which now are 25 (24 for the users and one for signaling). Moreover, the signaling slot here only sends 1 bit. This means that I have less users per frame, but I also have a per-frame datarate that is lower w.r.t. the E1 one, because here the frame lasts as the E1 one but here I have only 25 slots, of which the signaling one only sends 1 bit. So in T1 I have:
8x24 + 1 bits per frame → 193 bits per frame → 1544 Kb/s
So in T1 I have 1544 Kb/s while in E1 I have 2048 Kb/s.

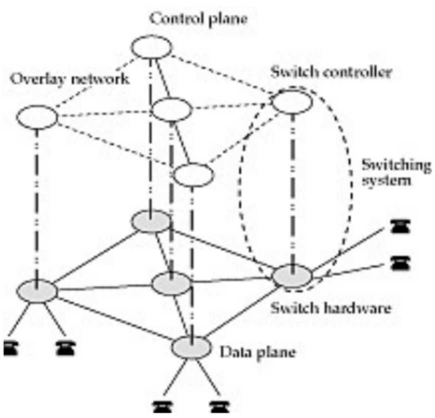
4.4.4 Reserved Slots

Last aspect is the following, we do not allocate all the slots of a frame to the users, but in the E1 standard 2 slots are reserved while in T1 1 slot is reserved. This for control purposes. It's

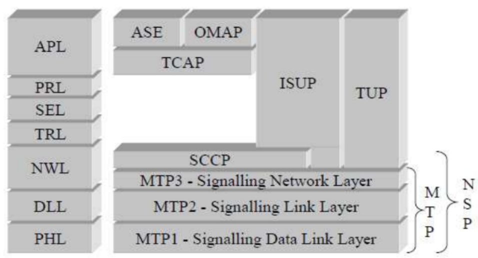
like each slot represents a channel, and we also have control channels.
Let's consider the E1 case. One of the slots is used to mark the beginning of the frame, so it has an alignment purpose. The other slot has the following purpose: at that time it was decided that slot number 16 was a time slot not used for transmit user information, but was used for signaling, in other word the network has the possibility to transmit control stuff in the 16th slot of each frame. Tipically this signaling channel was used for metadata which told how to route the 30 telephone channels.
The 16th time slot has itself a datarate of 64 Kb/s. With the passing of time it was thought to divide this datarate among the other slots, meaning 64 Kb/s divided by 32 giving as outcome 2 Kb/s. So at the end what happened was that if 2 Kb/s per-user was enough to transmit control information for that user, after a while it happened that these 2 Kb/s were too low for the users.
Today what happens is that the most of the information exchanged through the network is control information, as an example in mobile network we have to transmit the position in which we are, which reaches the VLR, and VLR has to communicate to the HLR.

4.5 Switching System Components and SS7 Network

With the passing of time the control information exchanged through the network was getting higher and higher, so it was decide to split the network into the Data Network and Signaling Network. As for today this decision is beeing taken into account also for the IP network, inedeed the IP network exchanges control information putting it in the headers of the packets.



(a) Control Network vs Data Network



(b) SS7 Stack

SS7, which stands for **Signaling System 7**, is a Control Network, physically separated by the Data Network, and SS7 controls the Data Network. This was designed by providing its protocol stack (figure b) with its Data Unit formats. So this Signaling Network is a packet-based network, and in particular it has been the first packet-based network, that controlled the telephone data network, then the IP network was invented.

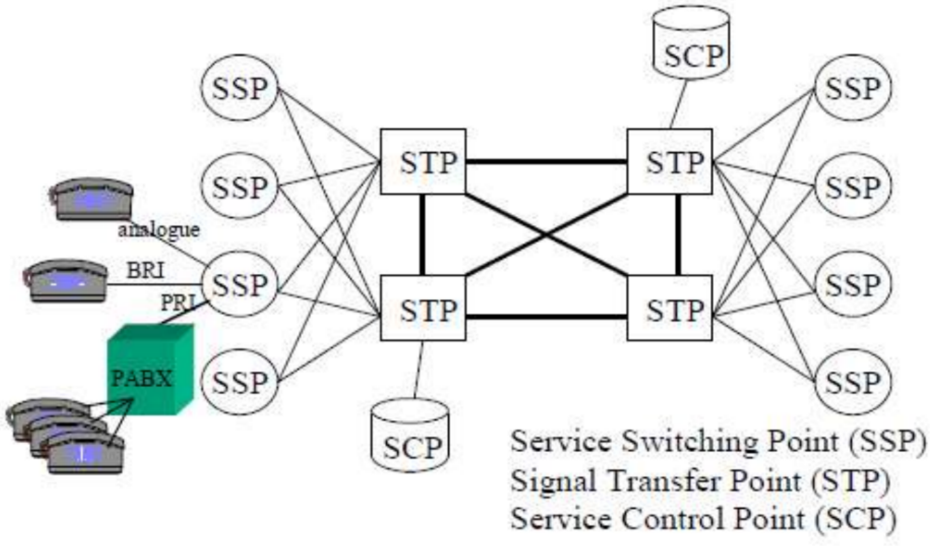


Figure 24

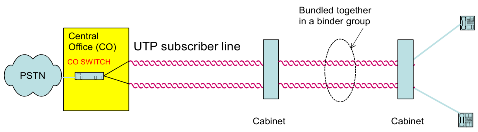
As the IP network has its network components (routers, switches, gateways...) also the SS7 network had its network components, which were Service Switching Points (SSP), Signal Transfer Point (STP), Service Control Point (SCP), so we can see these components as the routers of the old telephone network. In order to make an example, a SCP is the intelligence of the network, think about the green numbers, they are special numbers because a green number is associated to different "stations", meaning that depending on where I am the nearest station associated to that green number answers.

5 XDSL: X Digital Subscriber Line

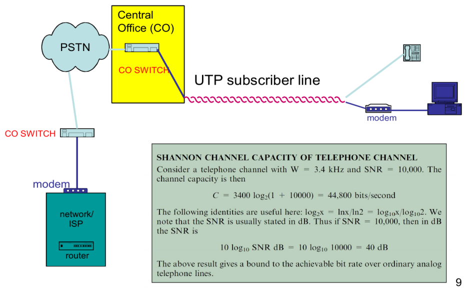
XDSL in an achronim that refers to all the different **DSL** technologies. DSL was born on top of the already present infrastructure which was the telephone network, for which the very last mile (the *twisted pair* interconnecting the *C.O.* and the *Users*) was in an **analog** form, the ADSL tried to implement in that part of the network a **digital** communication.
The first tentative of implementation was the **ISDN** (*Integrated Service Digital Network*), then there were other standards, some of which failed, other succeeded, and today we are in the era of **ADSL** and **vDSL**. The ISDN was the first attempt to use the telephone line to transmit also Internet data, and this was the so called *Integrated Service*, for which the Internet data was modulated onto an analog signal that could be transmitted on the telephone line in the 0-4KHz bandwidth. So in order to sum up, previous to the ISDN the telephone line was used only for voice, with ISDN it was used both for voice and data. Moreover, since the data was transmitted in the same bandwidth of the voice, at that time only voice or data at a time could be transmitted.

Then, with the advent of ADSL, we have instead also other bandwidths in order to keep data apart form the voice, as we are going to see.

5.1 Pre-ISDN and ISDN architectural schemes



(a) Pre-ISDN architecture



(b) ISDN architecture

(a) Every user had its own *twisted pair*, starting from him and going towards the C.O. Then the twisted pairs of different users are physically put togheter by the Cabinet in a bigger cable, the *Binder Group*.Maybe I can have more levels of cabinets, meaning that the next cabinet puts togheter the wires of the previous cabinet, but however this is a detail.

(b) How did we pass to ISDN? Put a modem at the user side and transmit the digital signal of the PC by means of analog signals. The modem transmits the digital signal building an "analog" signal with a specific shape able to transport in the correct way the digits. So the modem actually generates an analog signal with a shape that represents the bits we have to transmit.
The ISDN converted the digital signals into analog signals that had the same bandwidth of the voice, so that it could be transmitted into the already present twisted pair.
As the **Shannon formula** states, the capacity of the telephone channels was *44,800 bits/s* $\approx$ *44 Kbit/s* and this was the first capacity of the channels used to implement the digital communication.

The twisted pair, as well as the copper wire, is an unshielded twisted pair, and the fact that they are unshielded means that it is affected by interferences coming from out of the cable, so the longer the cable, the more interferences happen. Indeed the attenuation/distorsion of the signal was proportional to the distance, and so the higher is the distance, the lower the bitrate can be.

5.2 Asymmetric DSL

Remember that the voice channel was full bidirectional (one twisted pair both to transmit and receive). At that time the service that was used in the digital form was the video on demand, where an high datarate was required from *C.O.* to *Users* and a low datarate in the reverse way. So they tried to save bandwidth, meaning high bandwidth in **Downstream** and low bandwidth in **Upstream**. from that Asymmetric.
Indeed what was done was to have, thanks to the Frequency Multiplexing, three separate channels in the same twisted pair, the *Telephone*, the *Upstream* and the *Downstream*.

This new standard with 3 kinds of channels (telephone, upstream, downstream) was delivering to the users 8 Mbits/s over 2 kilometers of UTP (Unshielded Twisted Pair). Then the new standard, ADSL+, was designed to reach ip to 24 Mbit/s, depending on the distance from the C.O.

Notice that ADSL did not change the ISDN architecture, it only changed the way the twisted pair was used.

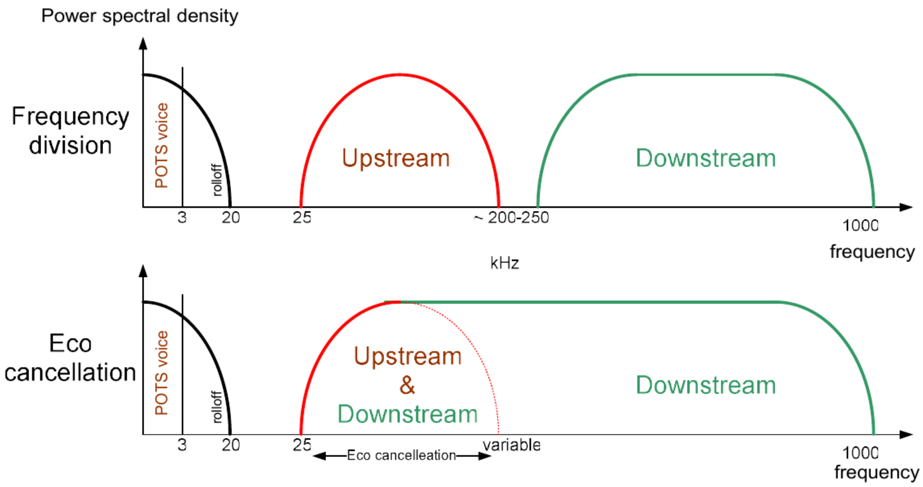


Figure 26

The 3 communication channels were implemented in the way the picture above illustrates: the telephone bandwidth was kept as it was (0 - 4KHz, actually 20 KHz due to the sounds when telephone buttons were pressed but this is a detail), then other bandwidths were used, one for Upstream and one Downstream.
These 3 communication channels sitted in the same twisted pair, so this was meaning to have a Frequency Division Multiplexing because on the same cable different services were provided, each one at its frequency range.
So the 3 channels were multiplexed in the single cable, and the standard gave [0-20] KHz range to telephone (bidirectional), about [25 - 200] KHz range to Upstream (monodirectional), about [250 - 1000] KHz range to Downstream (monodirectional).

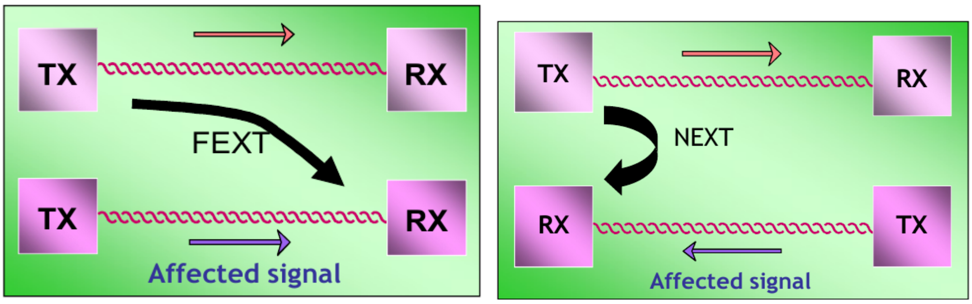
There could be also the possibility to have an higher Downstream bandwidth, and what was thought was that instead of going over 1000 KHz I go under 250 KHz so I have the sovrapposition of Upstream bandwidth and Downstream bandwidth. This means that in the 25-250 KHz range I have a bidirectional communication. This result in the possibility to transmit contemporarly the downstream and the upstream, meaning the overlap of the two signals.
How can this work? Remember that also the voice, the telephone communication, was bidirectional. Indeed the voice communication supports the bidirectionality in the same bandwidth in

the same cable. How?
Here TDM and FDM do not work!
Instead it was used a device called **Echo Canceller** which cancelled the echo that was generated by a transmitter in the communication. In order to make an example, suppose A and B talking at the same time, A receives both its echo and B transmission, but since A knows what it transmits A can cancel what transmits from the whole it receives, and the Echo Canceller performs this job.
So since the Upstream is something the user sends and knows how is its shape, I can let Downstream to overlap the Upstream bandwidth and with the Echo Canceller it is sufficient to subtract, from the Downstream data, the Upstream one.

In order to summarize, the standard was the one of the graph on top of the picture, but in the cases where Downstream required more bandwidth, Upstream and Downstream shared a portion of bandwidth and so an Echo Canceller was required.

5.3 Cross-Talk: FEXT and NEXT

Remember that the twisted pairs of the users are merged togheter in a Binder Group by the Cabinet. It happens that in the same Binder Group some cables transmit in the same frequency at the same time, and this results in an interference named **cross-talk**. There exist two kinds of cross-talk: **FEXT** and **NEXT**.



(a) FEXT (b) NEXT

FEXT: I have two transmitters on one side, and two receivers on the other side. Who can understand there is an interference is the receiver. So the lower receiver receives the transmission of the lower transmitter (the good one) and the transmission of the upper transmitter (the bad one), because this last one crosses the cable and interference with other cables. The transmission that generates the interference is already alterated when it crosses its cable, then moreover there is the factor of distance meaning that the signal that generates the interference continues to be alterated. For these two reasons this kind of cross talk is not such an issue.

NEXT: This kind of cross-talk happens when there is a transmitter and a receiver next to him. What happens is that the receiver that is hit by this issue receives both the good transmission (what is destinated to him) and the signal of the transmitter next to him which generates the interference. This kind of cross-talk is worse than the FEXT, because first of all the good signal is attenuted due to the distance, and moreover the bad signal is very close to the receiver so it is strong and overlaps the weak alteared good signal, resulting in a mess.

There is more, the effect of a signal to cross the cable depends on the frequency, the higher the frequency is, the more the signal crosses the cable. Since in the telephone communication the frequency was not so high this effect was not much present, but the digital communication worked at higher frequencies so it suffered more for this effect.

Add here SNR discussion...

5.4 ADSL infrastructure

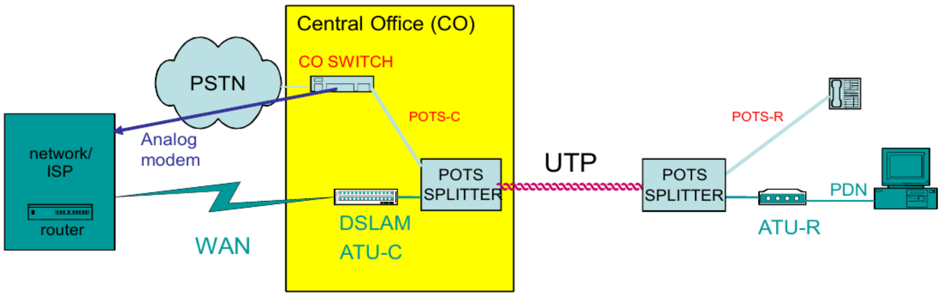


Figure 28

The ADSL architecture was the one represented here. We have a modem in the homes of the users that is the ADSL Termination Unit (ATU). This identical ATU had to be present also in the C.O. side. This results in having in the C.O. a number of modems equals to the number of users connected to that C.O.
Indeed the user as the ATU-R where R stands for Remote, while the C.O. has the ATU-C where C stands for Central.
The reason of the presence of the modem is such that the data that goes out of the PC, which is already an analog signal, does not match the Upstream and Downstream bandwidths seen before, so the modem has to perform the *modulation* in order to make the things to match with the ADSL standard according to the range of frequencies seen before.
In the picture we see another device, the **splitter**. The splitter is a filter that filters the telephone signals from the rest, meaning it acts as a low-pass filter for the telephone bandwidth and as an high-pass filter for the data bandwidth. As for the ATUs, we have a splitter at the user side and a splitter at the C.O. side. Then the splitter at the C.O. side forwards the telephone signal towards the telephone network and the digital data towards the data network.
We see also the DSLAM device. This is the DSL Access Multiplexer.

5.5 Echo Canceller

In the ADSL architecture, the **NEXT** interference happens when an user is in *Downstream* and another user is in *Upstream*, meaning that from the C.O. point of view there are respectively a transmitter and a receiver that are close to each other. So the NEXT affects, regarding the ADSL architecture, the Upstream receiver, because in the Upstream case the receiver is the C.O. and since there is also a Downstream there is a modem in the C.O. that is transmitting and its signal interferes with the one that is envolved in the Upstream.
Note: this NEXT interference can happen only in the case where the Downstream bandwidth overlaps the Upstream one, because otherwise since upstream and downstream use different bandwidths they cannot interfere.

In order to sum up, if the users use the scheme where Downstream and Upstream overlap, when there is a Downstream for an user and an Upstream of another user at the same time, in the C.O. the modem involved into the Upstream (the receiver) is affected by the **NEXT**.
This has been asked in some exam.
Another question frequently asked is the following:
How can we resolve this?
If the Upstream receiver knows what the Downstream transmitter transmits, he can recognize it and perform the echo-cancellation. In order to perform this, the Downstream transmitter and the Upstream receiver have to lay in the same device, and it is not so simple.
In order to be more precise, there are three ways to solve the NEXT problem:

- reduce the number of cables in the binder group, meaning binder group not much populated (decreases the probability of Upstream and Downstream of two different users at the same time)
- avoid frequency overlapping (this works better)
- echo cancellation, what we just discussed

5.6 ADSL modulation

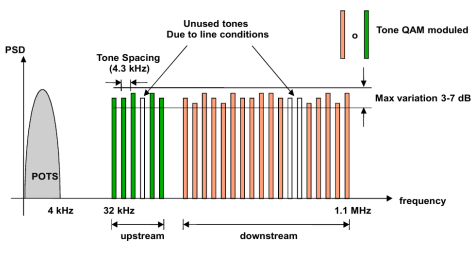
ADSL supports two types of modulations. One of the twos is the **CAP modulation**, which stands for *Carrier-less Amplitute/Phase modulation*, the other one is the *DMT modulation* which stands for *Discrete Multi-Tone modulation*.

5.6.1 CAP modulation

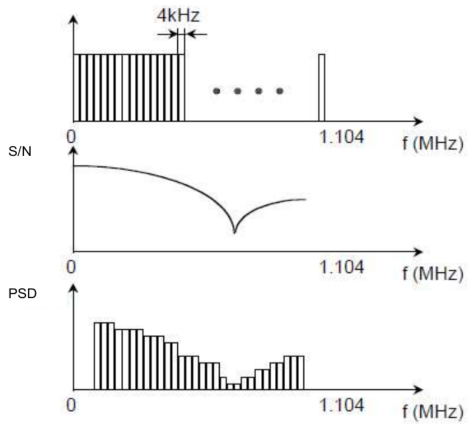
CAP modulation is Carrier-less because we expect that the carrier frequency is placed at the middle of the bandwidth but the transmitter does not transmit the carrier information, in other words does not tell the carrier frequency, meaning that it has to be known in advance.

5.6.2 DMT modulation

DMT is the most used kind of ADSL modulation. The idea is: let's consider the whole bandwidth of the channel (i.e. the downstream bandwidth of the ADSL), then instead of occupying the whole bandwidth, we divide the bandwidth in **Subcarriers**, a.k.a. **Tones**. So tones are small portions of bandwidth and are separated each one by a gap of 4.3 KHz.



(a)



(b)

Why dividing in tones? The idea is to simulate an *ideal channel* (top one of picture (a)): when we consider a bandwidth, which is basically a set of frequencies, we would like to have a "flat

behaviour of the channel", meaning that all the frequencies that the channel supports are attenuated in the same manner. But ideal channels do not exist, instead we have *real channels*. So the frequencies of a real channel are not attenuated at the same manner, but viewing the channel as a composition of several subchannels, each subchannel approximatively attenuates the frequencies it is composed of in the same manner, as we can see in the picture (b).
In few words, the DMT modulation aims at having lots of almost ideal channels.
There is a device, called *Serial to Parallel converter*, which takes the stream of bits of the ADSL communication, splits the bits in different substreams (# substreams = # subchannels) and each substream is sent on a signal that is placed on the carrier frequency of that subchannel.

The standard states that the bandwidth of the ADSL is divided into 256 subchannels of 4,3125 KHz each, resulting in a whole bandwidth of 1024 MHz.

Notice that the behaviour of each channel is different, because it is true that they all have the same bandwidth, but in each subtone I can send more or less bits depending on the specific *Signal to Noise Ratio* (SNR). The decision of how many bits to send into the specific subchannel depends on the SNR of that subchannel, and this is performed by the *Waterfilling algorithm*.

Then, at a physical level, the bits in each channel are sent by means of the **Q.A.M. modulation**.
Now, when there is noise in the channel, the noise can make the point to move into the QAM constellation, leading in an error of reading the symbol. Now, the higher is the number of points of the QAM constellation, the higher is the probability of error due to the noise, this is the reason why where there is less noise I have an high n for the n-QAM, meaning high bitrate, while where there is more noise I have to use a low n for the n-QAM. The decision of the right QAM for the channel is made by the *WaterFilling algorithm*.

In the ideal case, the standard states that if I use 25 channels in the upstream, 249 channels in the downstream, every channel having 4KHz of bandwidth, the maximum theoretical datarate I can have in the ADSL is 1.5 Mbit/s for the Upstream and 14.9 Mbit/s for the Downstream. These two numbers come from the assumption that at each Hz 15 bits are transmitted.

So, considering the upstream as example, I have 15 bits per Hz, so 60 Kbit/s per channel (15 bit per Hz, 4 KHz in total, so 60 Kbit/s). Upstream has 25 channels, so $25 \times 60 Kbit/s = 1.5 Mbit/s$.

In order to assume 15 bits per Hz, it was used the **Shannon** formula using the typical SNR of the ADSL channel.

5.7 Splitter configuration

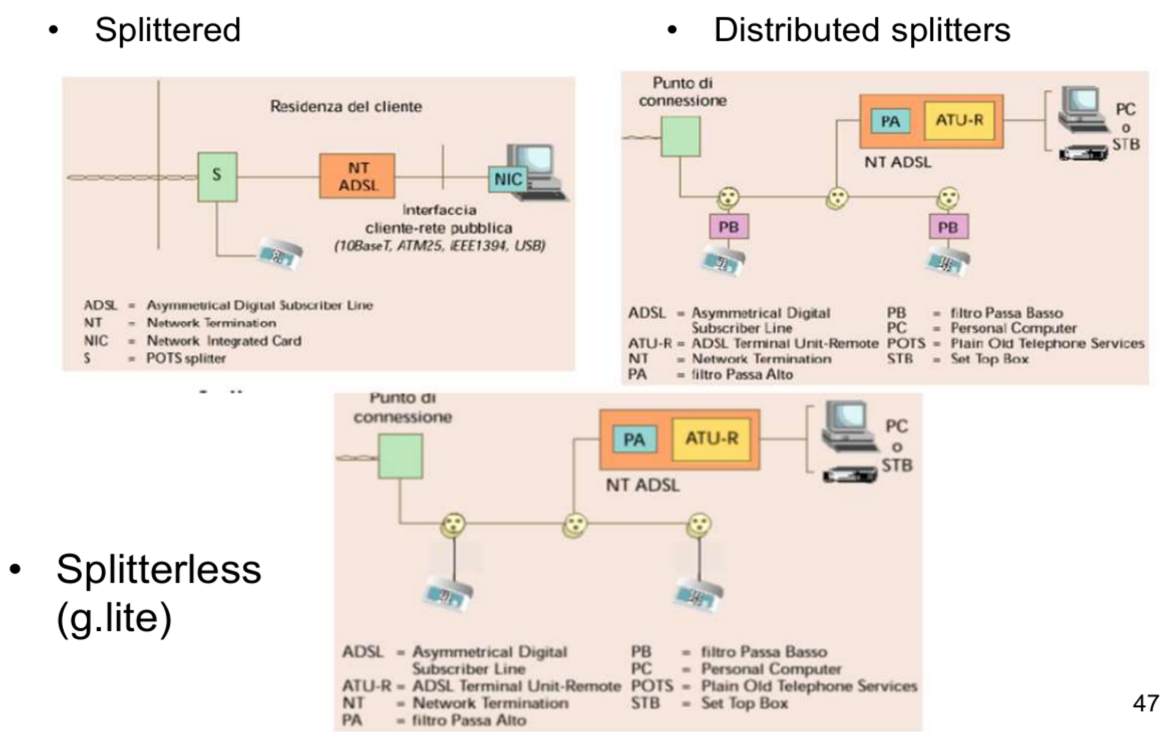


Figure 30

It is a low pass filter for the telephone part and an high pass filter for the digital data part. In the past the splitter was after the home modem (*Splittered configuration*), often near the wall, to which both modem and telephone were plugged, today we can have it embedded in the modem, indeed what happens in the modern modems is that the telephone is plugged into the modem.
In case of *Distributed splitters configuration*, there is a single Low-Pass filter per telephone (allowing to have more telephones in the home), while the High-Pass filter that lets the Digital Data of the ADSL to pass is embedded into the **NT ADSL** (the modem).
Today also the fixed telephone exchanges digital data, and this is achieved by the famous **Voice over IP (VoIP)**. In this latter case, the modem is able to convert the telephone data in digital data and this data is transmitted over the Internet, relying on the Digital Communication of the ADSL and no longer on the old telephone communication below 4KHz.

5.8 ADSL architecture and protocols

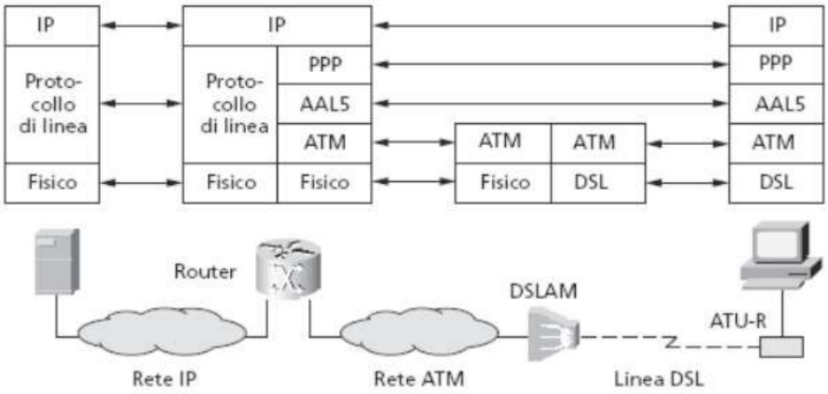


Figure 31

We have our **ATU-R** (embedded in our **NT ADSL** which is our modem), which is connected to the **DSLAM** which has multiples **ATU-C** (one per user), this DSLAM *multiplexes* the digital data of the different ATU-C in order to send it towards the rest of the network.
The DSLAM at the beginning was putted at the C.O. side, today the trend is to put it in the cabinets.
For what concerns the protocols, on top of the physical layer we find the **ATM** standard, which was a *layer2* standard to transmit data over the physical layer. So it is still used in some part of the network, in the picture we see it is used between the C.O. DSLAM and the first router of the Internet Network.
On top of ATM there is another layer, the **AAL** (Adaptation Layer), which adapts the ATM to the above protocols, the most common one is **Ethernet**. Above we find **PPP** and at the end **IP** (which is the highest protocol into the network since we do not need Application Layer there). What we have to understand about DSLAM and ATM is that DSLAM is a sort of router for the ATM standard (as routers are for IP), and the DSLAM is connected to the first router of the network by means of the ATM protocol, then that router communicates with the rest of the network by means of IP. So the final concept is that our modem at home (NT ADSL, which embeds the ATU-R) is connected to the DSLAM located at the C.O. by means of a copper wire, then the DSLAM is connected to the first router by means of ATM standard. This was the interface from our ATU-R towards the network, while the interface between the ATU-R and the homes is mainly Ethernet, by means the Ethernet Cables with Ethernet Ports (in recent times we have mainly Wi-Fi in place of Ethernet Cables).

5.9 PPP: Point-to-Point Protocol

PPP is a layer2 protocol (below IP). Since its main purpose is to directly connect two entities, it is used to connect the user to the C.O. of its ISP.
In general we know that below IP we find Ethernet. PPP is a sort of light ethernet, because is a protocol that provides to layer3 protocols (mainly IP) a layer2 that can be easly adapted to ethernet and also to other layer2 standards.

5.9.1 PPP Data Process Unit

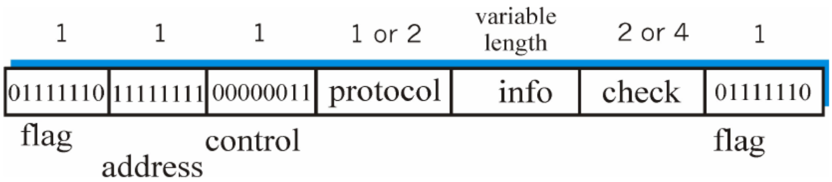


Figure 32

- **flag**: delimits one DPU from the others
- **address**: not used since useless into a Point-to-Point link
- **protocol**: identifies the datagram encapsulated into the **info** field
- **info**: data of the above protocol, the length of this field is usually up to 1500 bytes since the common protocol that exploits PPP is **Ethernet**

5.9.2 PPP sublayers

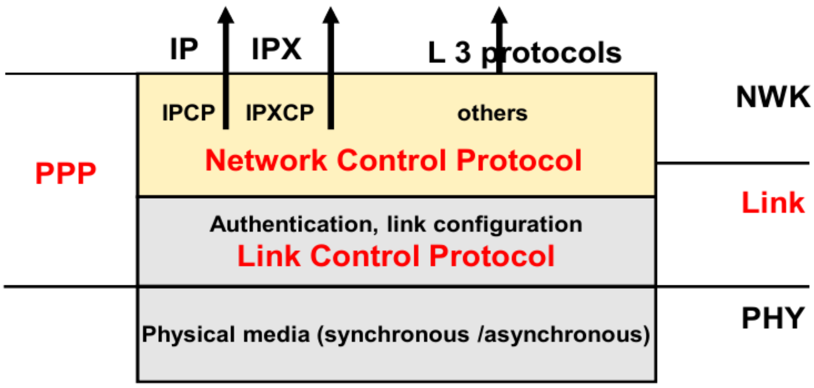


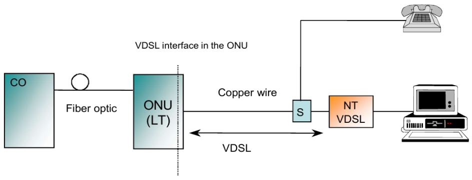
Figure 33

PPP has on its turn 2 sublayers: **Network Control Protocol** and **Link Control Protocol**. Link Control Protocol establishes the link between the two ends. In order to establish the link the two ends negotiate parameters.
Network Control Protocl is the layer of PPP that interfaces with the above protocols, e.g. IP. Practically this portion of PPP ensures the adaptation of all the possible layer3 protocols in order to let them to use PPP.

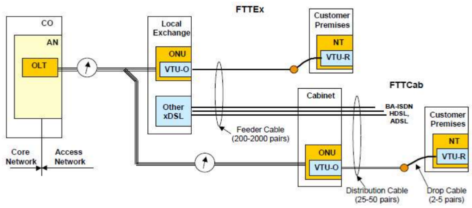
5.9.3 PPP authentication

Another important aspect of PPP is that it also implements the authentication between the modem at home and the C.O. At the beginning it was a password-based authentication, now it is a challenge-based authentication.

5.10 VDSL: Very-High-Bit-Rate Digital Subscriber Line



(a)



(b)

VDSL is the most modern tipology of **xDSL**. The concept is almost identical to the one of the ADSL, but here instead of having the **fiber** ending at the C.O. (a.k.a. **FTTE**), we have the *Fiber to the Cabinet* **FTTC** architecture, or maybe even *Fiber to the Home* **FTTH**.
Here the copper wire is shorter, since only reaches the Cabinet instead of reaching the C.O., meaning that the **SNR** is higher (better tolerance against noises) and this allows to have higher bitrates.

6 LTE/LTE-A Cellular Systems

Before starting to talks about LTE we have to recall some concepts related to the Cellular System and in general to the *Wireless Communication*.
Wireless Communication is used in several parts of the network. In order to give an idea, the very first implementation of Wireless Communication was the one used for the television: we had a base station placed in a specific position in order to reach a huge number of houses, for instance on top of a hill. In the case of satellite we have an antenna on the satellite which covers a very large region.

6.1 Wireless Technology Evolution Path

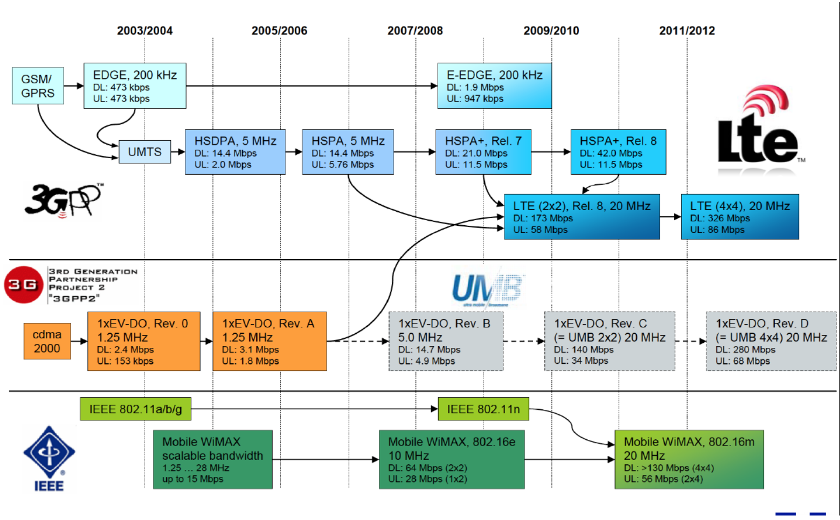


Figure 35

The wireless communication has been implemented by many standard, which are listed in the picture. The very first Digital Wireless Communication was the **3G** which was already a digital standard. Then the standards started to be separated, indeed the main ones belonged to the **GSM** family, and this in Europe, belonging to the *3GPP Consortium* (*Third Generation Partnership Project*). So starting from GSM, 3GPP started to standardize many other standards, for example **UMTS**, then **HSDPA**, **HSPA** and so on up to **LTE**.
In parallel there were other standards provided by other Consortiums, for example in U.S.A. there was the *3GPP2 Consortium* which standardized the **CDMA**, which has converged to the **LTE**. In the meanwhile, the *IEEE Consortium* focuses on **IEEE 802.x** standards. The **WiMAX** standard for example tries to emulate the Cellular System.
However, the trend today is to try to converge to a unique standard.

6.2 Introduction to Mobility

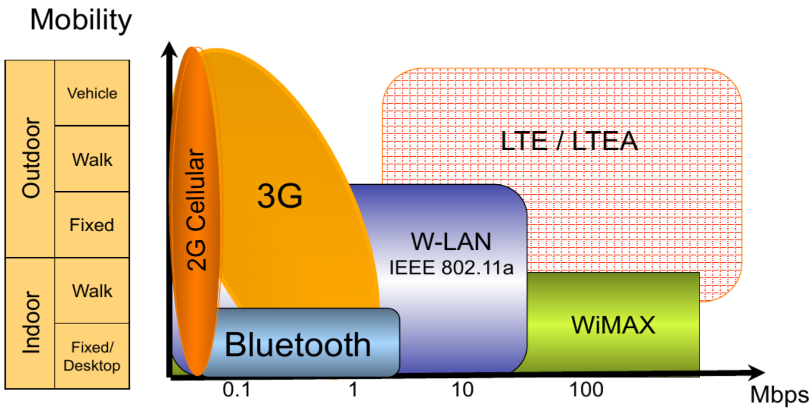


Figure 36

We can distingush technologies in two dimensions: *bitrate* and *mobility*. For instance, **2G**, which is a quite old standard, had very low bitrate but very high mobility/coverage. **3G** instead gained some higher bitrate w.r.t. 2G but it still low.
The most interesting aspect is the mobility. Mobility is a quite complex aspect, indeed determines how the network is built in terms of network components of a Cellular Network.
Mobility can be for example *fixed*, *slow* or *fast*. For example if I am in a room and I do not move my mobility is fixed, so I do not have to change my Access Point, in general we can say I keep staying in the same *cell*, the one covered by the Access Point. If on the countrary I am on top of a veichle, then my mobility is fast and this can be an issue.
At the end the challenges that have to be faced in a Wireless communication are first of all the fact that I want no wires, and then the fact that I want to be able to move during the communication. The fact of having no wires is not a topic for our discussion, we give it for free. For what concerns the mobility, we can distinguish mainly between two examples of Wireless communication: the WiFi that implements a Wireless LAN and the Cellular Communication. Indeed, the first one allows very limited mobility of the user, if you think when you are connected with your laptop to an access point and you fall out of the range of that access point, you loose your connection. The second one instead is different, because if you are connected to a base station, and you fall out of the range of that base station, you keep your connection. So both WiFi and Cellular Communication are two kinds of wireless communication, but the second one is more powerful because not only is wireless but also allows you to have the freedom to move everywhere. So keep in mind that *Wireless does not always mean Mobility*.

6.3 Wireless Network Elements

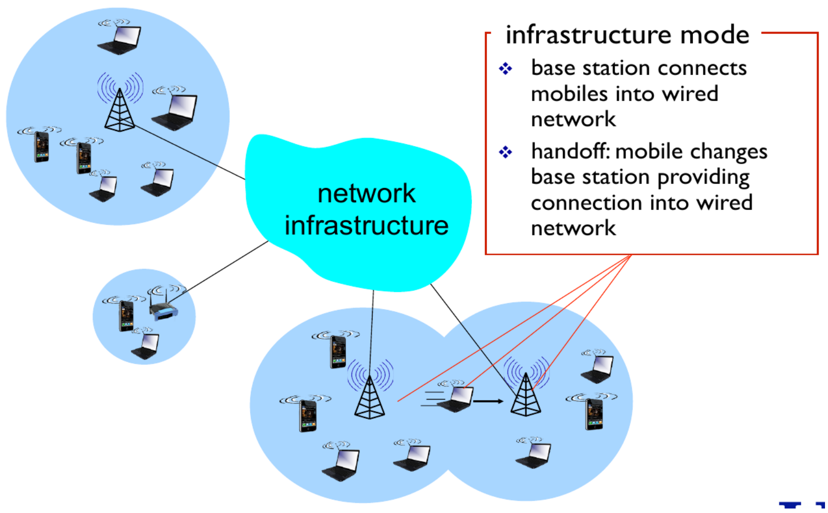


Figure 37

The general architecture of a Wireless Communication System is the one displayed in the picture. In the most general setting, I have hosts that want to be connected withouth wires and that require mobility, and there are the *Access Points* that provide to the hosts the possibility to be connected without wires. Then for example the Access Point in the WiFi case is the modem, while the Access Point in the Cellular Network case is the Base Station, and we know that each Base Station is able to cover a certain region called **cell**.
Then the Base Stations can be interconnected each one either with a wired connection or with a wireless one. Also in case two antennas are wirelessly connected it is a P2P connection, and this can be achieved by tune some parameters that allow to achieve the highest datarate on that wireless link. *However (for simplicity) we consider as Wireless part of the network only the one between antennas and hosts*, in other words we do not focus on the interconnection of the antennas.
The **Network Infrastructure** that appears in the picture is simply the already existing one, so the backbone with routers (as when we talked about the Telephone Infrastructure). Indeed, at the end we only study the Access Network part in our topics.
Notice that since a single Base Station covers a cell with several hosts, the Base Station has to resolve the Multiple Access problem (many hosts connecting to the same network element).

6.3.1 Handoff Problem (Switching cells)

The main problem the Base Stations have to solve is the problem of **Handoff**: during my communication I move, and it is possible that I go out of my current cell and I enter inside another cell, and the connection has to be kept alive. If the cells region are very small (which is the trend today, *300-600 meters*), the frequency I switch cell increases, indeed if I am in a fast train (*300 km/h*) I switch cell more or less every couple of seconds.

6.4 Ad-Hoc Networks

Another kind of topology is the **Ad Hoc** one. This is called also Infrastructureless topology, indeed we have no Base Stations, the hosts are themselves the node of the network. When or where we use this kind of infrastructure? An example can be the case where I place drones around that collect informations and the drones send informations each other and also to our smartphones, so the nodes of the network are the drones and our smartphones.

So while in the typical wireless infrastructure we can distinguish between hosts and access points, in Ad-Hoc networks there are no access points and the only elements of the network are the hosts, which represent themselves the network itself.

6.5 Wireless Links Characteristics

For what concerns the Wireless Links characteristics, obviously some of them are not present in Wire Links. We can list the main characteristics:

- **path loss**: the signal strength decreases as the distance increases. This is true also for the wired connections.
- there are more interferences w.r.t. wired links
- **multipath**: when you transmit a signal to the end of the room for example, this signal crosses the room and reaches the end but the signal also reaches the walls ,meaning that signal jumps on the walls and create paths that make interference. There are also physical object that reflect certain frequencies, so depending on the carrier frequency of the signal some object cna reflect the signal. So at the end more paths are generated, and these paths are basically replicas of my signal, which arrive late to the destination w.r.t. the original signal. Indeed late signals can be recognised and cancelled.

As for the wired links, all the wireless links tune their parameters depending of **SNR**. Again, the SNR affects the bitrate of the communication (*the higher the SNR, the higher is the points in the costellation, the higher is the bitrate*). One particular aspect is that every user has a different SNR (because each user has its distance from the Base Station), so each user has is data rate. Of course this is true also for Wired Links like ADSL, but here the aspect is more evident.

6.6 Hidden Terminal Problem

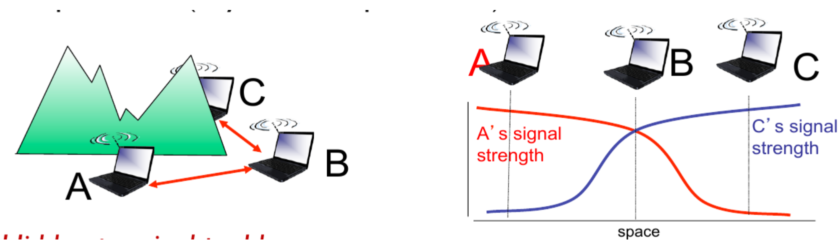


Figure 38

Hidden terminal problem: A talks with B, C talks with B, and A and C do not know the presence of each other. (Fare Domanda!!!)

6.7 Code Division Multiple Access

CDMA, which stands for **Code Division Multiple Access**, is the standard that as we saw was introduced by *3GPP2 Consortium*. The idea is not to divide resources in time, nor in frequencies, but dividing them in the *code domain*. This means that users transmit at the same time at the same frequency, and this is tipically not allowed because of interferences, apart the case of a transmission done in this way.

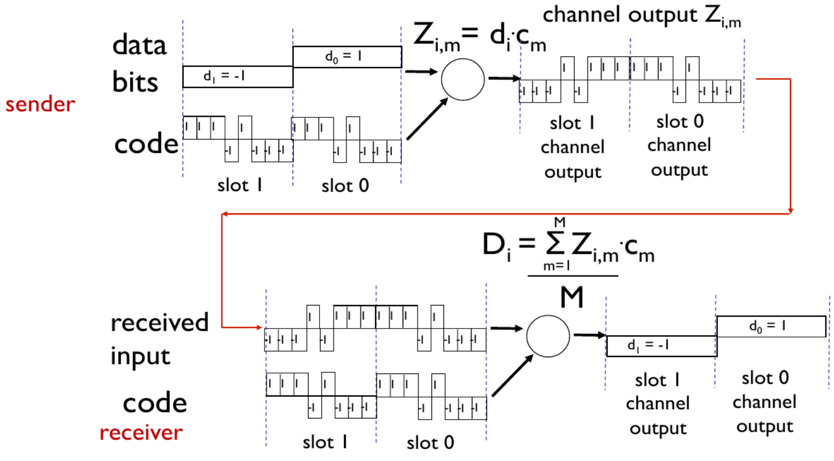


Figure 39

The transmission is done applying a code, named also "chipset", that is a sequence of bits that are multiplied by the emitter. So if I have to transmit a 1, I transmit it multiplied by the code. The receiver then multiplies by the same chipset what he has received, reconstructing the bit sent. This process does not make sense in case of a single user, because this method has a drawback, because instead of sending a bit I am sending 8 bits, so I waste a portion of datarate for doing this thing.

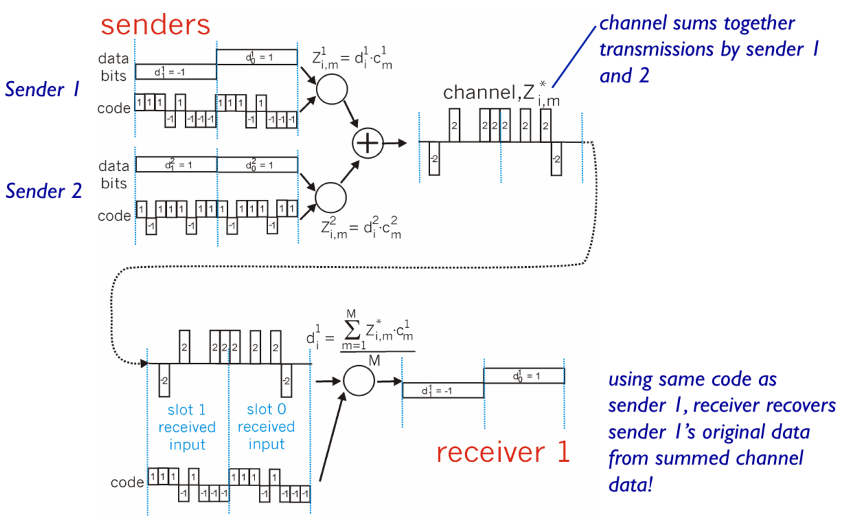
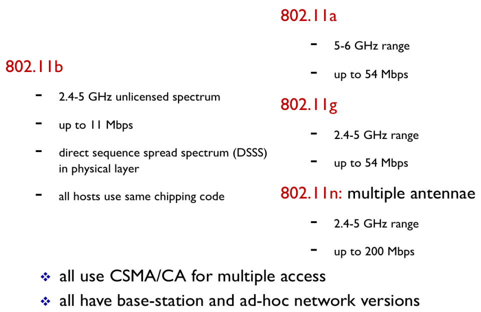


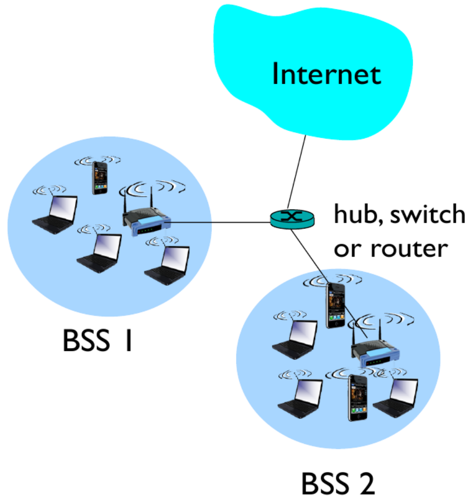
Figure 40

However, in case of multiple users this make sense because each user uses a different code, so in order to know which is the bit send by a certain user I multiply the chipset received for the chipset of that user.

6.8 IEEE 802.11 Wireless LAN



(a)



(b)

In the picture (a) we just see the characteristics of the different **802.11x** standards. For example, for the **802.11b** standard we have *85MHz* of bandwidth (from *2.4GHz* to *2.485GHz*), for which we have up to *11 Mbps* of data rate. Just for curiosity, the reason why 802.11n reaches 200 Mbps is that there are multiple antennae and they exploit the multipath. In the picture (b) instead we see the general topology. In particular, we have *Access Points* (as base stations are for cellulars), that manage cells that in the *802.11* nomenclature are called **Basic Service Set BSS**. Different access points are interconnected, tipically in a wired fashion, to a router, in order to reach the Internet.

6.8.1 IEEE 802.11 channels allocation

When you want to connect, you have to select your Access Point, and each Access Point is characterized by an id named **SSID**. This id is broadcasted by the wireless signal so a device can understand that there is an Access Point close to it.

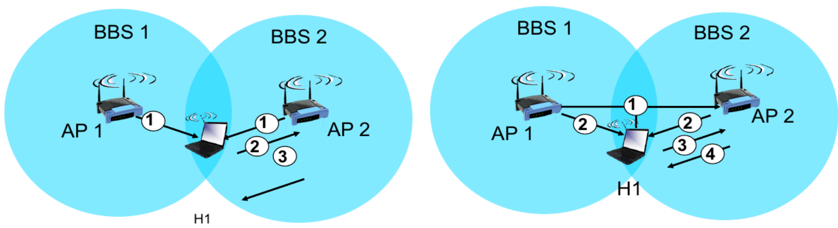


Figure 42

There are two ways for which a device is connected to an Access Point and they are:

- **passive scanning**: the device collects the SSIDs of the different available Access Points and the user selects the one with the highest signal strength.
- **active scanning**: in the device there is a list of all the WiFi networks I visited so far, the device broadcasts its list of WiFi networks, and if one of them is present then the device associates with that Access Point, if there are more than one then the user selects the one with the highest signal strength.

6.8.2 IEEE 802.11 CSMA/CA

In the Wireless communication, as for other topologies like bus topology on a LAN, there is the problem of collisions. In the Ethernet standard this problem was solved with **CSMA/CD** which is Collision Detection. In the Wireless communication however it is no possible to detect collisions, so the **CSMA/CA** is exploited, meaning Collision Avoidance. We can do this in two ways: the **DIFS-SIFS** method and the **RTS-CTS** method.

6.8.3 DIFS-SIFS CSMA/CA

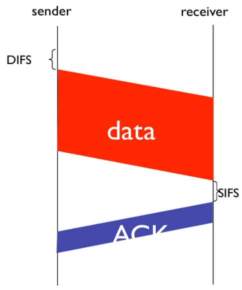


Figure 43

I listen the channel for a slice of time called **DIFS**, if the channel is free for this period of time then I am allowed to transmit and so I send my data. Who is receiving this data has to send back an **ACK**, and it does the same thing, it listens the channel for a slice of time called **SIFS**, and if the channel is free then it sends the ACK.
If I do not receive the ACK because maybe a collision happened, I have to do the BakeOff mechanism, for which I retransmit after a period of time and this period of time increases as the number of failing transmission increases, up to a certain treshold.
The period of time after which I decide to retransmitt is called *window*, and the windows have been standardized because it was discovered that some devices waited a small period of time before retransmitting, meaning to be more aggressive and so these devices stoled the channel.

6.8.4 RTS-CTS CSMA/CA

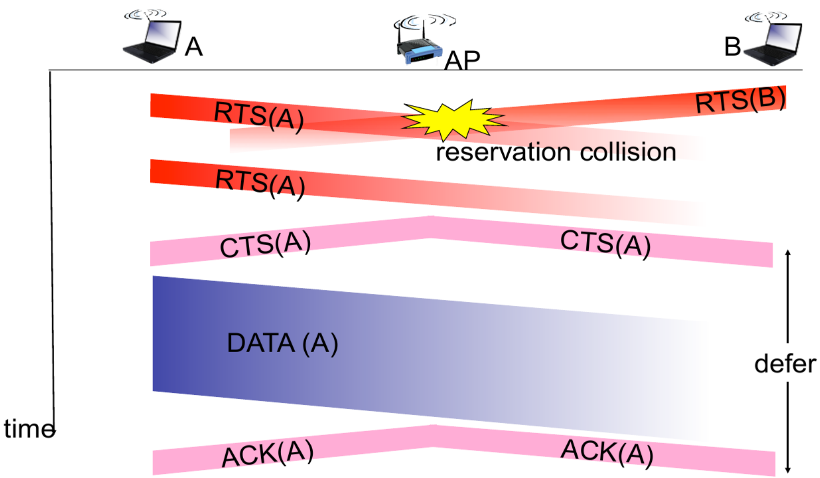


Figure 44

The other way is the RTS-CTS way, that uses the **RequestToSend** and **ClearToSend** mechanisms.
The idea is the following: A sends a **RequestToSend(A)** (that is a very short packet so the collision probability is low) to the Access Point, but there is also B that does the same thing (this is the *hidden terminal problem*), if this happens it means that a collision happens and the Access Point cannot send the **ClearToSend**. So A tries again and say that this time the **RequestToSend(A)** reaches the Access Point. So the Access Point broadcasts a **ClearToSend(A)** message and now A is allowed to send its data.

6.9 Cellular Network and Mobility

6.9.1 Introduction

The most difficult part of the Cellular Network, whether it is LTE or 3G or whatever, is how to handle mobility. We have basically two kinds of mobility: I move inside the same cell, or I move throught different cells (meaning I change my serving Base Station, this is the Hand-Off). The second one of course is the most difficult to implement, and moreover consider that the smaller the cell, the faster you switch cell, but also the faster you move, the faster you switch cell.
In case of Cellular System the mobility is handled by a special device, the **Mobile Switching Center MSC** (which we already met in the 2G-3G architecture), which is a network element dedicated for the mobility (finds out where the user is, follows the user behaviour, sees which is the Base Station that is serving the user and is ready to make the Base Station to change when the user moves to the other cell). This device understands where I am by seeing the signal strenght, so it works at a physical layer.
This metodology was already implemented in the very first technologies of Cellular Network, and since the Cellular Network started to provide Internet to the users, this methodology had to be reflected also for the Data Network, not only for the Voice Network. Indeed system also implements the dynamic streaming quality for example, so if my datarate is low the video switches to the lower level of quality, meanig *http* requests are done for the video with lower quality, while if I have an high datarate http requests are done for the higher video quality.

6.9.2 2G (voice) network architecture

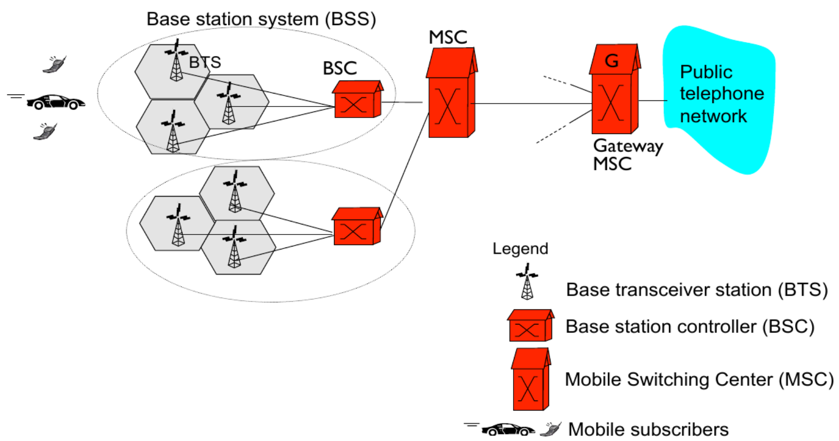


Figure 45

We have already seen this architecture, but here we focus more on the network components that we are interested in. We saw that several base stations are handled by a **Base Station**

Controller BSC, but we also see that several BSC are interconnected with a **Mobile Switching Center MSC**. A MSC indeed is responsible for a quite large area, and connects the base stations of which it is responsible to the Core of the already present Telephone Network, like a gateway connects all the computers of a LAN to the Internet.

6.9.3 3G (voice+data) network architecture

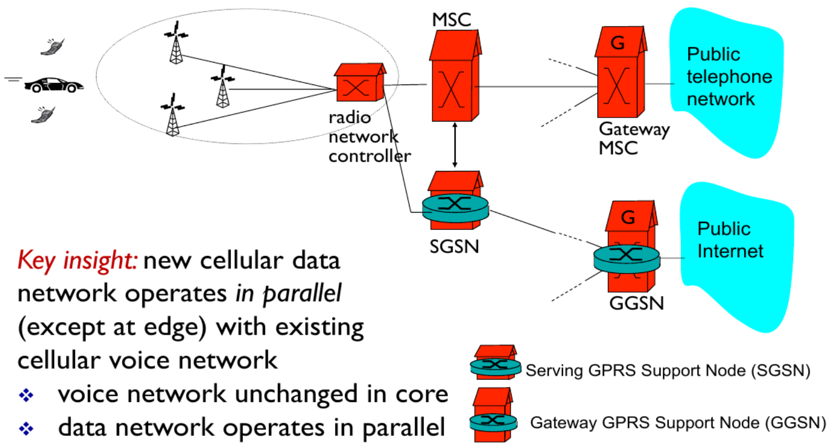
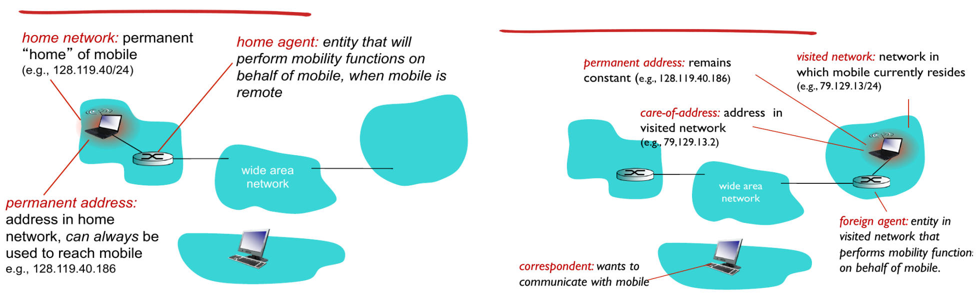


Figure 46

With the advent of 3G, the cellulars started also to exchange data other than voice. So basically the infrastructure of the 2G was kept and something else was added in order to achieve also data exchange. In particular the BSC is still connected to the MSC for the voice part, but here is also connected to some other components which handle data and that are connected to the Internet.

6.9.4 The mobility implementation: Mobile IP



(a) (b)

You know, in case of IP, when we communicate we are related to an address. In case of cellular systems each user is associated to two IP addresses, we are going to see why.
There is a standard called *Mobile IP* which manages IP addresses in mobile networks, and in more general words manages the mobility of the users. According to this standard each user has two kind of addresses and there are two kinds of network elements.
The two kind of addresses are named:

- **Permanent IP Address**: it is my constant IP, so in someway my static identity

- **Care-of IP address** (or Temporary IP Address): the IP that is given to me by the network I am currently

The two network devices are actually two databases:

- **Home Agent** (or Home Mobile Switching Center in GSM nomenclature): it is the Mobile Switching Center of the network where I tipically stay, which holds my Permanent IP Address in the *Home Location Register HLR*.

- **Foreign Agent**: it is basically the Home Agent of the network I am currently. If you think about it, from my perspective it is the Foreign Agent of the visited network,, but for another user it can be its Home Agent.

So, my Permanent IP Address is related to a network, named Home Network, where I tipically stay. If I go out of the Home Network I end up in another network, that we call *Visited Network*, where I am handled by the Foreign Agent, which is basically the Home Agent of that network. What happens is that when I am in the Visited Network I cannot use my Permanent IP Address, and the reason is simple, IP Address has the Net ID and Host ID part, so since I am in another network I cannot use my Permanet Address because in my Permanent Address the Net ID is the address of my Home Network, while I am currently in the Foreign Network. Basically everything is coherent with the typical LAN networks.
So since I am in the Foreign Network, I have the Care-of IP Address that is given to me by the Foreign Agent. From now on, for the sake of simplicity, let's say that:
Francesco=Permanent IP
Marco=Temporary IP
What happens now is that when Francesco enters into a visited network, the Foreign Agent tells to the Home Agent of Francesco "look, if someone looks for Francesco, please know that Francesco is in my network and its address is Marco".
Then the *correspondent* is simply the one that wants to contact Francesco.

Once we have seen the vocabulary of the Mobile IP, there are two ways to implement the actual mobility: *Indirect Routing* and *Direct Routing*.

6.9.5 Mobile IP: Indirect Routing

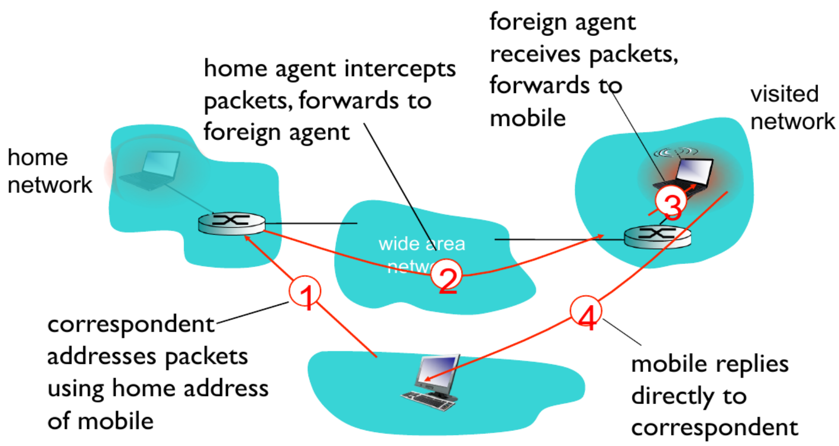


Figure 48

Imagine that the correspondent wants to reach Francesco, which is not in its Home Network. The packet is anyway routed to the Home Network of Francesco, using the permanet address,

but the Home Agent knows that Francesco is not in the Home Network and in particular knows the Visited Network where it is, so it forwards the packet towards the Visited Network, the packet is received by the Foreign Agent, which sends the packet to Marco (step 1-2-3 of the picture). When Marco has to reply, it simply directly send the response to the corresponent.
This way of routing is also called **Triangle Routing**. This solution has a drawback: the routing scheme we have just seen happens also if the correspondent is close to the receiver, even if they are in the same Visited Network, and this is quite inefficient.

6.9.6 Mobile IP: Direct Routing

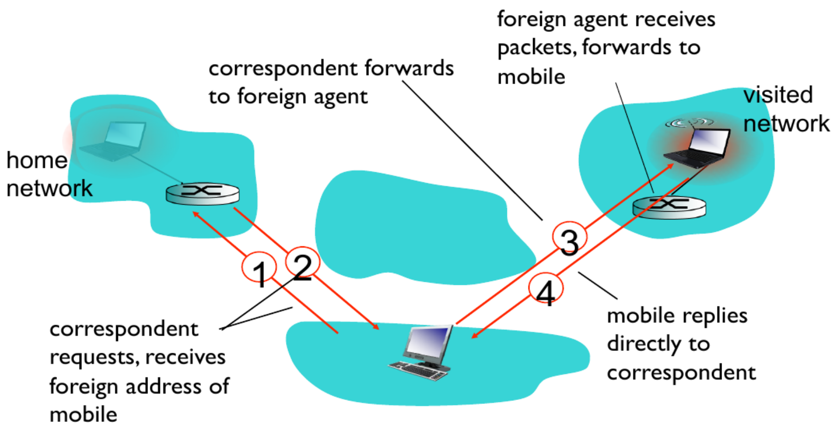
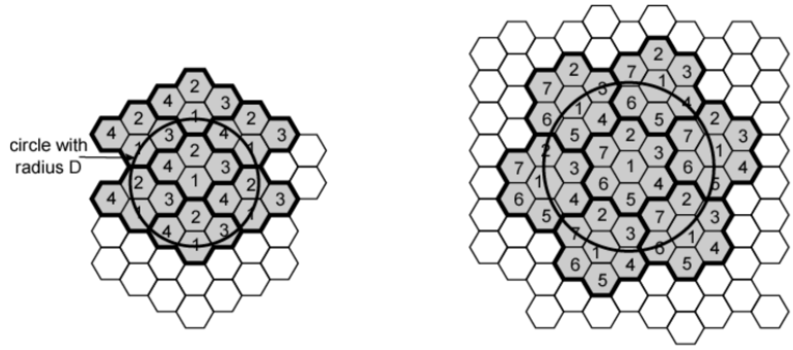


Figure 49

This is the solution to Triangle Routing inefficiency. The correspondent first sends a packet to my Home Network asking where I am, the Home Agent replies with my current care-of address (remember that it is the Foreign Agent of the Visited Network to tell to the Home Agent I am in its network and which is my care-of address), so here it is the sender to forward the packet to my Foreign Agent (before it was my Home Agent to care of doing the forward), and the Foreign Agent sends the packet to me. From now on the sender directly uses the Care-of address.
The problem here is that if I move again, the sender is no longer able to reach me, so goes again to my Home Agent to ask where I am now. Instead in the Indirect Routing it was all transparent from the sender point of view, because it sent the packet to my Home Agent anyway, no matter where I am.
What actually happens is that the Foreign Agent of the first visited network is the Anchor Agent. Now when I move to the second visited network, the agent of this latter visited network will be the new Anchor Agent, which will arrange to have data forwarded from the old Anchor Agent. So this builds a chain, meaning that the packet is always routed towards the first Anchor Agent, and then Anchor Agents follow their chain in order to reach me.

6.10 Frequency Reuse



(a) Frequency reuse pattern for N = 4 (b) Frequency reuse pattern for N = 7

Figure 50

The Frequency Reuse technique allows the Cellular System to use the same frequencies among different cells. In particular, each base station in its cell uses a certain bandwidth, and adiacent cells do not use the same bandwidth in order not to create interference, but cells that are not adiacent or close each other can use the same bandwidth. So this is the frequency reuse, and this technique forms a pattern of reuse, which can be viewed in these pictures. For example with a pattern of $N = 4$ there are 4 frequencies that are reused among cells, while for the pattern $N = 7$ the frequencies that are repeated among cells are 7 different ones. Of course, the more is the pattern, the smaller the cells can be because I can put them close each other since each of them will use its frequency and they will not interfere.
So instead of having a bigger cell with a big bandwidth and its single carrier frequency centered in that bandwidth, I replace it with multiple cells exploiting frequency reuse, and what can be done in this way is that two users in two different cells with the same carrier frequency can communicate contemporarly on the same frequency (no interference because they are in different cells). The only drawback is that now I have to put a Base Station for each cell, so the number of Base Stations increases and this has a cost (even in terms of messages exchanged by base stations).

6.11 LTE LTEA

Once discussed the main characteristics of Cellular Infrastructure in general and of Wireless Communication, let's dive into the LTE standard.

Frequency Range	UMTS FDD bands and UMTS TDD bands					
Channel bandwidth	1.4 MHz	3 MHz	5 MHz	10 MHz	15 MHz	20 MHz
1 Resource Block (RB) =180 kHz	6 RB	15 RB	25 RB	50 RB	75 RB	100 RB
Modulation Schemes	Downlink	QPSK, 16QAM, 64QAM				
	Uplink	QPSK, 16QAM, 64QAM (⇨ optional for handset)				
Multiple Access	Downlink	OFDMA (Orthogonal Frequency Division Multiple Access)				
	Uplink	SC-FDMA (Single Carrier Frequency Division Multiple Access)				
MIMO technology	Downlink	Wide choice of MIMO configuration options for transmit diversity, spatial multiplexing, and cyclic delay diversity (max. 4 antennas at base station and handset)				
	Uplink	Multi-user collaborative MIMO				
Peak Data Rate	Downlink	150 Mbps (UE category 4, 2x2 MIMO, 20 MHz) 300 Mbps (UE category 5, 4x4 MIMO, 20 MHz)				
	Uplink	75 Mbps (20 MHz)				

Figure 51

The LTE standard introduced different aspects, the main ones are:

- Several bandwidths where the system can transmit data, here we can see some channels. For example the 1.4 MHz one can be used for some services, the 3 MHz one for other services and so on. We have 6 different channels, each one with its bandwidth.
- The network supports different modulation schemes, QSPK, 16QAM, 64QAM, depending of course on the status of the channel.

Another novelty is in the nomenclature, now the base station is called **eNB** (eNodeB), while the single user is referred as **UE** (User Equipment).

6.11.1 Adaptive Modulation and Channel Quality Indication CQI

As we saw in the summary picture, the LTE supports different modulations (QPSK, 16QAM, 64QAM). The modulation is adaptive, in the sense that depending on the noise of the channel a better or worse modulation is chosen among the available ones we have listed. This is achieved by the *Channel Quality Indication*: the single UE has the task to send back to the eNB an indication, CQI, directly proportional to the Signal-to-Noise ratio of the channel, that is a number in the range [0;15] (indeed it is encoded with 4 bits) where if CQI is 15 it means that the channel where I am right now is the best one, if it is 0 it is the worst one. The CQI signal is contineously sent. So depending on the CQI received the base station can modulate the signal toward the UE in a certain way, for example for a CQI equal to 15 the best modulation is chosen, while for a CQI equal to 0 the worst modulation is chosen.
What instead happened in the past was that the signal was chosen depending on the distance of the user, so the users in a given range received the same signal, users in another range received another same signal and so on, so there was no feedback in that manner, here the CQI is a sort of feedback.

6.11.2 Hybrid Automatic Repeat Request (HARQ)

For what concerns the error corrections, there exist two main protocols:

- **FEC** (Forward Error Correction): in addition to the data I have to transmit I add more data to protect what I have to transmit
- **ARQ** (Automatic Repeat Request): in this case you transmit your data, if the data failed you retransmit the data

LTE combines the two solutions, so it uses the Hybrid Automatic Repeat Request (HARQ): I add the FEC to the data and in the meanwhile I also do the ARQ, but the FEC (in particular the amout of extra information to protect data) can be changed dinamically. This happens at physical level, so it is complex and we will not enter into the details.
However, just to give an idea, LTE uses HARQ in the following way: when I transmit a packet, I transmit it with the addition of FEC, then suppose it does not arrive correctly, so a NACK is sent back, so I have to retransmit, but *the receiver keeps the old packet*, now I retransmit, but with less FEC than before. The receiver can now combine the old packet and the new one, togheter with the two FECs, to better detect the error.
Notice that, previous to the LTE, in the wireless communication FEC and ARQ were used mutually exclusive, and in general FEC was the predominant one.

6.11.3 Resource Blocks

We already saw that in the LTE we have 6 channel bandwidths (1.4 MHz, 3 MHz, 5 MHz, 10 MHz, 15 MHz, 20 MHz) and each one is composed of different Resource Blocks, where a

Resource Block, as we will see, is the minimum amount of resource allocation that can be given to an user. In particular, for the channels we have listed the resource blocks per channel are respectively 6, 15, 25, 50, 75, 100.

6.11.4 Downlink Multiplexing: OFDMA

OFDM is a particular Frequency Division Multiplexing that exploits the "ortogonality" feature, that allows to have, in a certain bandwidth B, more sub-carriers w.r.t. classical FDM (do not care of the details). But OFDM is tipically used in such a way that to each user the whole bandwidth is assigned (each user uses the whole set of subchannels/subcarriers), and the user uses it in certain time periods (so at the end is a sort of TDM for the users).
LTE instead uses OFDMA (represented in the right part of the picture), for which each user uses a certain portion of bandwidth (a subset of subchannels) in certain time periods.

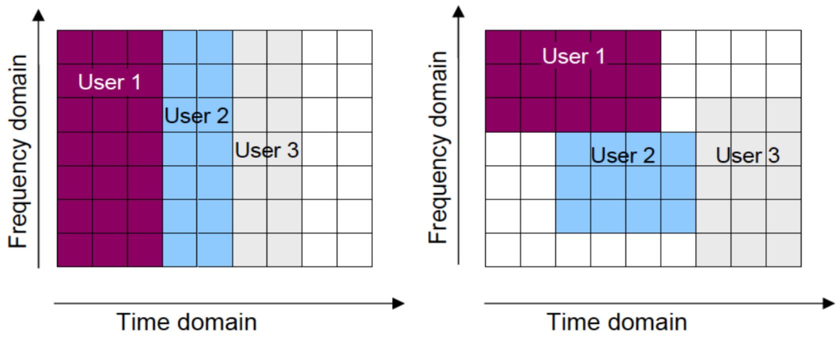


Figure 52

As we said before, there are Resource Blocks. A Resource Block is the minimum resource that can be allocated in order to transmit. In details, a single Resource Block is basically composed of 12 sub-carriers (in particular 12 subchannels 15 KHz each, so 180 KHz of bandiwdth) in the frequency-domain and 1 time-slot of 0.5 ms in the time domain.
So the more Resource Blocks a user has the higher will be its datarate.

6.11.5 FDD and TDD

In LTE, since we use the concept of Resource Blocks, we use both Time Division Multiplexing and Frequency Division Multiplexing. The frame lasts 10 ms and is composed of 20 slots, so each slots lasts for 0.5 ms (as we said for a Resource Block).
Now there are also other two concepts: **FDD and TDD**. We have to receive and transmit at the same time (bidirectional channel), so FDD (Frequency Division Diplexing) states that the Downstream is transmitted on a certain portion of bandwidth and Upstream is transmitted on another portion of bandwidth, while TDD (Time Division Diplexing) states that on a time period I transmit the Uplink and in another I receive the Downlink.
Which is better?
FDD is easier, because let's say we are using the 10 MHz channel of LTE, it can be designed for example to use 5 MHz of the whole bandwidth for the Upstream and 5 MHz of the whole bandwidth for the Downstream, having the same amount of resources in both directions. Of course I can achieve the same with the TDD. What I can do with TDD is for example, if I want to design the system such that 2/3 of time is for the Downlink and 1/3 is for the Uplink, then I can reserve 2 frames for the Downlink and the next one for the Uplink and this pattern repeats. This can be implemented by simply communicating it from the eNB to the UEs, while with FDD the whole has to be designed in advance. So for simplicity and convenience, the TDD is

mostly used.
In order to sumup, the Diplexing of Upstream and Downstream is achieved divinding the time and not the frequencies, in particular some frames are chosen for the Downlink and some other frames are chosen for the Uplink.

6.11.6 Uplink Multiplexing: SC-FDMA

For what concerns the Upstream, LTE uses **SC-FDMA** (Single Carrier -Frequency Division Multiple Access).
What happens is that the user has still its set of subcarriers, but while in the OFDMA scheme each subcarrier brings a symbol for a certain symbol time (long), in the SC-FDMA the symbols are sent sequentially by the whole subcarrier set and the symbol duration is quite smaller w.r.t. to the symbol time in the OFDMA scheme. This is why it is named "Single Carrier", because in SC-FDMA the carriers of the user form a single channel, the user uses all of them in the Uplink, so it is like to have a Single Carrier with a larger bandwidth.

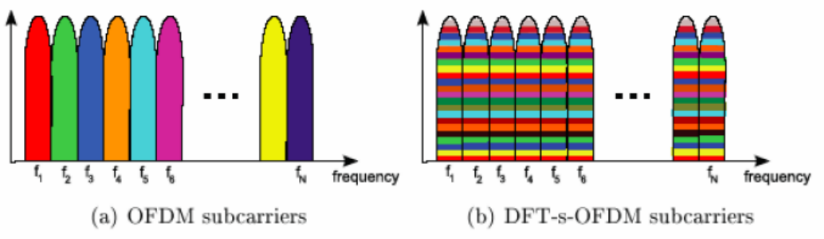


Figure 53

6.11.7 MIMO Transmission and MIMO Modes

All the systems transmit by having an antenna at the transmitter and another at the receiver. The **MIMO**, that is *Multiple Input Multiple Output*, allows to do two different things (mutually exclusive):

- **Transmit Diversity**: if I use two antennas at the sender side and two antennas at the receiver side, and the antennas transmit the same information, each receiving antenna receives two different signals bringing the same information, and this can be used to make the data more robust. The result is that I improve the SNR and so I can have an higher data rate.
- **Spatial Multiplexing**: I can use each antenna to transmit a different data stream (of course each antenna has to use its bandwidth otherwise they will interfere each other).

6.11.8 Beamforming

In the past, antennas transmitted omni-directonally, beamforming is instead a technique for which an antenna transmits a signal that can be directed to a certain direction. Imagine that each antenna covers a set of users (because it is directed towards those users), an antenna directed to another set of users can reuse the frequencies of the other antenna since the two antennas do not interfere (they are directed towards two different directions). So Beamforming can be used to achieve frequency reuse.
LTE and in particular LTEA massively use this technique. There are also base stations that have arrays of beamforming antennas, in order to provide accurate 3D beamforming to targeted users.