Anatomy of a Network Connection – Web and OTT cases

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List of Acronyms and Abbreviations

3G	Third generation celular technologies
CATV	Comunity Antena Television
CDMA	Code Division Multiple Access
CDN	Content Delivery Network
CM	Cable Modem
CMTS	Cable Modem Termination System
CPE	Customer Premises equipment
DNS	Domain Name System
DOCSIS	Data Over Cable Service Interface Specification. DOCSIS specifies methods for transporting data over CATV networks
DSLAM	Digital Subscriber Line Access Multiplexer
DTT	Digital Terrestrial Television
DWDM	Dense Wavelength Division Multiplexing
E-UTRAN	Evolved UMTS Terrestrial Radio Network
FCCN	Fundação para o Cálculo Científico Nacional
FTTx	This is a generic expression to describe different fiber optics based telecommunication networks. Depending on the termination point of the fiber optics, these architectures have various designations: FTTN (Fiber To The Node), FTTCab (Fiber To The Cabinet), FTTC (Fiber to The Curb), FTTP (Fiber To The Premises), FTTB (Fiber To The Building), e FTTH (Fiber To The Home).
GE	Gigabit Ethernet
GPRS	General Packet Radio Service
GSM	Global Systems for Mobile Communications
HD / SD	High Definition / Standard Definition (video) signals
HFC	Hybrid Fiber-Coax
HLS	HTTP Live Streaming
HSDPA	High Speed Downlink Packet Access
HSPA	High-Speed Packet Access
HSUPA	High Speed Uplink Packet Access
HTML	HyperText Markup Language
HTTP	Hypertext Transfer Protocol
ICMP	Internet Control Message Protocol
IEC	International Electrotechnical Commission
IEEE	Institute of Electrical and Electronics Engineers
IP	Internet Protocol
IPTV	Internet Protocol Television
ISO	International Standards Organization
ISP	Internet Service Provider
ITU	International Telecommunications Union
LTE	Long Term Evolution
m3u	Computer file format for multimedia playlists
MIMO	Multiple-Input Multiple-Output

MPEG	Moving Picture Experts Group. This is one of ISO/IEC - International Organization for Standardization/International Electrotechnical Commission - study groups responsible for the production of standards for video compression and transmission; some of its work also applies to audio)
NFSI	NFSI - Soluções Internet, Lda (ISP company)
NGN	Next-Generation Networks
OFDM	Orthogonal Frequency Division Multiplexing
ONT	Optical Network Termination
OSI	Open System Interconnection
OTN	Optical Transport Network
OTT	Over the Top
POTS	Plain Old Telephone Service
PSTN	Public Switched Telephone Network
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keeing
RCTS	Rede Ciência, Tecnolgia e Sociedade
REFER	This neither an acronym nor an abbreviation. Is is the name of the company that is responsible for the railways infrastructure in Portugal. It is coined from "Rede Ferroviária"
RF	Radio Frequency
RNC	Radio Network Controller
RTMP	Real-Time Messaging Protocol
SDH	Synchronous Digital Hierarchy
SIP	Session Initiation Protocol
SNR	Signal to Noise Ratio
SONET	Synchronous Optical Network(ing)
TCP	Transmission Control Protocol
TTL	Time to Live
UDP	User Datagram Protocol
UMTS	Universal Mobile Telecommunications Systems
URL	Uniform Resource Locator
W-CDMA	Wide-band Code-Division Multiple Access
WiFi	This neither an acronym nor an abbreviation. It is coined from “Wireless Fidelity and refers to IEEE 802.11b wireless networking standard/technology
WiMAX	Worldwide Interoperability for Microwave Access
WLAN	Wireless local area network
WWW	World Wide Web
xDSL	There are various DSL technologies such as: - ADSL: Asymmetric Digital Subscriber Line - SDSL: Symmetric Digital Subscriber Line - VDSL (VHDSL): Very High Speed Digital Subscriber Line - HDSL: High Speed Digital Subscriber Line

1. Objectives

The objective of this document is to provide a basic understanding of the processes, technologies and agents involved when a connection is established between a user and a telecommunications network.
Two service cases are considered:

a) A web connection.
b) An OTT[1] (over the top) connection.

2. What are the Essential Steps Involved in a Connection to a Web Site?

To answer this question the following procedures will be used:

a) The connection will be made to a specific web site: www.oceanario.pt.
b) A diagnostic tool will be used, *Traceroute*[2], that identifies the route hops followed by the connection and measures transit delays of packets across IP networks,.

2.1 Tracing the connection using the "tracert" tool

Traceroute tool uses Internet Control Message Protocol – (ICMP) messages and relies on a function called TTL – (Time to Live) in the header of this Layer 3 protocol. ICMP operates between two hosts at Layer 3 (Network) level of the OSI model.
The usage of *traceroute* in the present case yields the following results:

```
C:\Users\ottserver>tracert www.oceanario.pt

Tracing route to www.oceanario.pt [81.92.221.193]
over a maximum of 30 hops:

  1    <1 ms     1 ms     1 ms  gt.det.ua.pt [193.136.82.254]
  2     1 ms    <1 ms    <1 ms  isengard.core.ua.pt [192.168.255.253]
  3     1 ms    <1 ms     1 ms  193.137.173.243
  4     5 ms     5 ms     7 ms  Router2.Campanha.fccn.pt [193.136.4.26]
  5     2 ms     1 ms     1 ms  ROUTER20.10GE.Porto.fccn.pt [193.137.4.25]
  6     2 ms     3 ms     2 ms  NFSi.AS25137.porto.gigapix.pt [193.136.251.112]

  7     6 ms     6 ms     7 ms  cr1-opo.r.nfsi.pt [94.46.143.1]
  8     7 ms     7 ms    11 ms  cr1-lis.r.nfsi.pt [81.92.223.185]
  9     7 ms     7 ms     8 ms  tge-1-1.br1.lis1.nfsi.pt [81.92.201.6]
 10     7 ms     7 ms     7 ms  ns3.waynext.com [81.92.221.193]

Trace complete.
```

Figure 1: *traceroute* results

[1] OTT stands for "over the top" and is a designation used whenever a provider delivers a service or product to users resorting to the infrastructures of third parties without their consent or active cooperation.
[2] In Microsoft Windows "tracert" general syntax is as follows:
tracert [-d] [-h MaximumHops] [-j HostList] [-w Timeout] [TargetName]
For additional explanations refer to:
http://technet.microsoft.com/en-us/library/bb491018.aspx

2.2 A possible interpretation into some detail

The results of the above *traceroute* can be interpreted[3] as shown in the following table:

	Hop	Device or Media	Local	Network/Operator/Owner	Technologies/Protocols	OSI layer
	0	Personal Computer (193.136.82.48)	GSBL UA	UA Ethernet Network / STIC / Aveiro University	HTTP	7 - Application
						6 - Presentation
					Port: XXXX	5 - Session
					TCP	4 - Transport
					IPv4	3 - Network
					Ethernet-IEEE802.3 or WiFI-IEEE802.11x	2 - Data Link
					UTP (Ethernet) or Free-Space Radio	1 - Physical
	TRANSPORT		UA	Free-Space radio (Public Domain Unlicensed) and/or UTP (Ethernet)		
	1	Router (193.136.82.254)	DETI UA	UA Ethernet Network / STIC / Aveiro University	IPv4	3 - Network
					Fast Ethernet (802.2; 802.3)	2 - Data Link
					100BASE-T (802.3)	1 - Physical
	TRANSPORT		UA	OPTICAL FIBRE Campus Backbone (Gigabit Ethernet)		
	2	Router (192.168.255.253)	STIC UA	UA Ethernet Network / STIC / Aveiro University	IPv4	3 - Network
					Gigabit Ethernet (IEEE 802.3-2008)	2 - Data Link
					Gigabit Ethernet (IEEE 802.3-2008)	1 - Physical
	TRANSPORT		AVEIRO	OPTICAL FIBRE Backhall (Gigabit Ethernet)		
	3	Router (193.136.4.26)	Estação Refer Aveiro	RCTS IP / FCCN /REFER	IPv4	3 - Network
					PPP (?)	2 - Data Link
					GE, OTN, SDH, SONET, etc	1 - Physical
	TRANSPORT		Linha do Norte	OPTICAL FIBRE FCCN Backbone (40λX40GB / DWDM)		
	4	Router Campanhã (193.136.4.26)	Estação Campanhã Porto	RCTS IP / FCCN / FCCN	IPv4	3 - Network
					Ethernet (?)	2 - Data Link
					GE, OTN, SDH, SONET, etc	1 - Physical
	TRANSPORT		Porto	UTP/Optical Fibre In-building cabling (Ethernet / GigaBit Ethernet)		
	5	Aggregator Router Porto (193.137.4.25)	Porto	RCTS IP / FCCN / FCCN	IPv4	3 - Network
					10 Gigabit Ethernet	2 - Data Link
					10GBASE (IEEE 802.3aX)	1 - Physical
	TRANSPORT		Porto	UTP/Optical Fibre In-building cabling (Ethernet / GigaBit Ethernet)		
	6	Gigapix Router (193.136.251.112)	Porto	GigaPix / FCCN / FCCN	IPv4	3 - Network
					10 Gigabit Ethernet	2 - Data Link
					10GBASE (IEEE 802.3aX)	1 - Physical

[3] The contents of column "Technologies/Protocols" is just a possible interpretation of the (scarce) information provided by the *tracert* results. Others could also be possible.

TRANSPORT		Porto	OPTICAL FIBRE Comercial Carrier Backbone (nλXmGB / DWDM)		
7	NFSI Router (94.46.143.1)	Porto	NFSI / NFSI /NFSI	IPv4	3 - Network
				Ethernet (?)	2 - Data Link
				10GBASE (IEEE 802.3aX)	1 - Physical
TRANSPORT		A1 or Linha do Norte	OPTICAL FIBRE Comercial Carrier Backbone (nλXmGB / DWDM)		
8	NFSI Router (81.92.223.185)	Lisbon	NFSI / NFSI / NFSI	IPv4	3 - Network
				10 Gigabit Ethernet	2 - Data Link
				GE, OTN, SDH, SONET, etc	1 - Physical
TRANSPORT		Lisbon	UTP/Optical Fibre In-building cabling (Ethernet / GigaBit Ethernet)		
9	NFSI Router (81.92.201.6)	Lisbon	NFSI / NFSI / NFSI	IPv4	3 - Network
				10 Gigabit Ethernet	2 - Data Link
				10GBASE (IEEE 802.3aX)	1 - Physical
TRANSPORT		Lisbon	UTP/Optical Fibre In-building cabling (Ethernet / GigaBit Ethernet)		
10	Waynext Server (81.92.221.193)	Lisbon	Waynext / NFSI / NFSI	HTTP	7 - Application
					6 - Presentation
				Port: 80	5 - Session
				TCP	4 - Transport
				IPv4	3 - Network
				Fast Ethernet	2 - Data Link
				10GBASE (IEEE 802.3aX)	1 - Physical

Table 1: Details of *traceroute* interpretation

The information from the table can also be used to construct the following scheme, revealing aspects of the possible network structure involved in the connection:

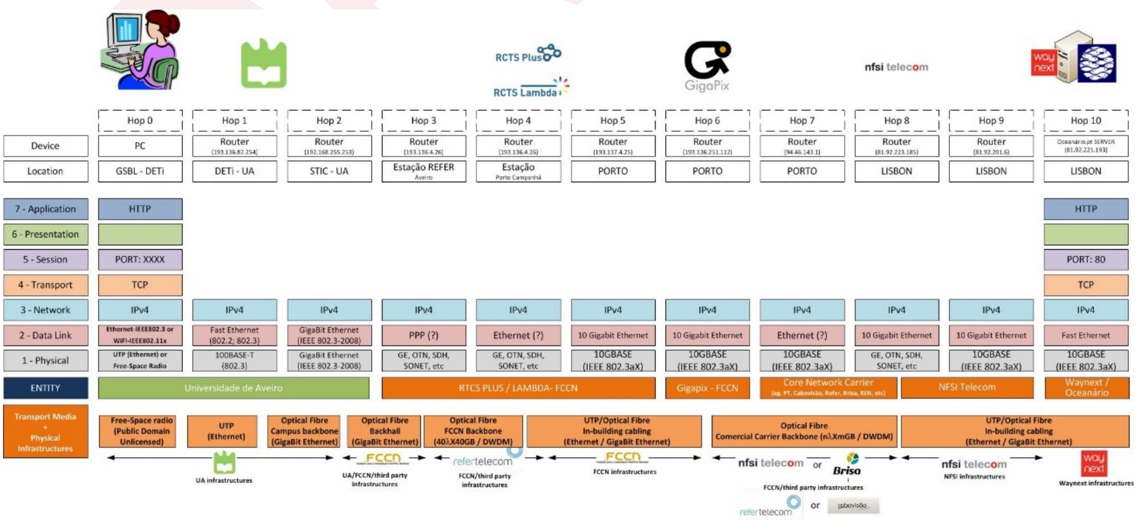

Figure 2: Possible network structure of the connection associated with in the *traceroute* results

2.3 Protocols and mechanisms involved in the connection to a WebPage

Previous section presented a possible interpretation concerning the different technologies and actors involved in the connection to a specific web site: www.oceanario.pt. Next section will look now at some of the associated tools and processes.

2.3.1 Web Browser

Opening a web page is usually the first step involved in accessing the World Wide Web (hereafter designated simply as the web) and gaining access to a large variety of information resources (documents, images, videos, etc) located in computers throughout the world.

In order to access a web page, a user opens a web browser. A web browser is a piece of software capable of retrieving and presenting information resources originating in different locations of the web. A browser also has the ability to travel across these locations looking for the desired information as determined by appropriate addresses in the form of URLs (Uniform Resource Locators). The typical format of an URL is as follows:

- http://www.example.com/index.html:

Where:

- http: designates the protocol to be used in establishing the connection with the information resource under consideration. In the present case, it is HyperText Markup Language (http) that inserts links to other documents.
- www.example.comm: is the host name.
- index.html: is the name of the file to be accessed in the host.

In spite of browsers being mainly intended to be used with the World Wide Web, they can also be used as access mechanisms in private networks (e.g.: LANs or VPNs) and file systems.

2.3.2 DNS

To find the information associated with a certain URL, the user opens a browser and inserts that URL into the browser search engine window:

Figure 3: Using a search engine

The browser must then convert that URL (which usually is expressed as an easily intelligible name, eg: [www.oceanario.pt)](http://www.oceanario.pt) into a numerical internet protocol (IP) address that uniquely identifies and locates computer services and devices across the web. This is done by sending a query to its local name server, the Domain Name System (DNS).

The search engine returns several results and one of them seems to correspond to the target destination:

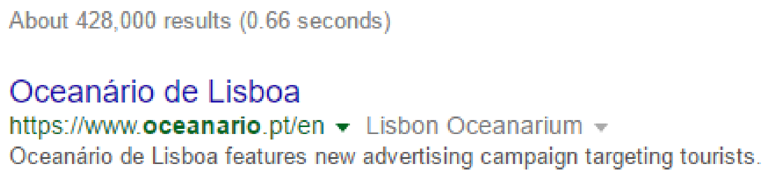

Figure 4: Using a search engine

The following diagram illustrates what has been said:

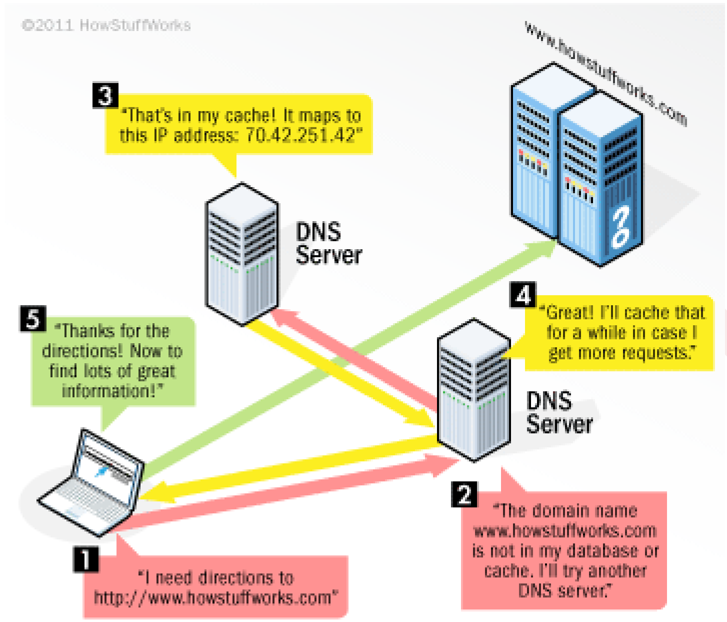

Figure 5: DNS illustration (https://s.hswstatic.com/gif/dns-rev-1.gif)

2.3.3 TCP

Transmission Control Protocol (TCP) involves some of the fundamental mechanisms of the Internet protocol suite.
Its main functionality is to ensure that all received bytes at one end of a communication system are identical to the bytes that are sent from the other end and are in the correct order. This involves following aspects:

a) Data handling and processing (streams, segments and sequence numbers)
b) Data transport, reliability and flow control
c) Management of ports, connections and connection identification

In approximate terms, it can be considered as being located at the level of the transport layer of the OSI model.
Detailed explanations of TCP can be found in [2, 3].

2.3.4 HTTP

Hypertext Transfer Protocol (HTTP) is an application layer network protocol built on top of TCP. HTTP functioning is illustrated in Figure 6.

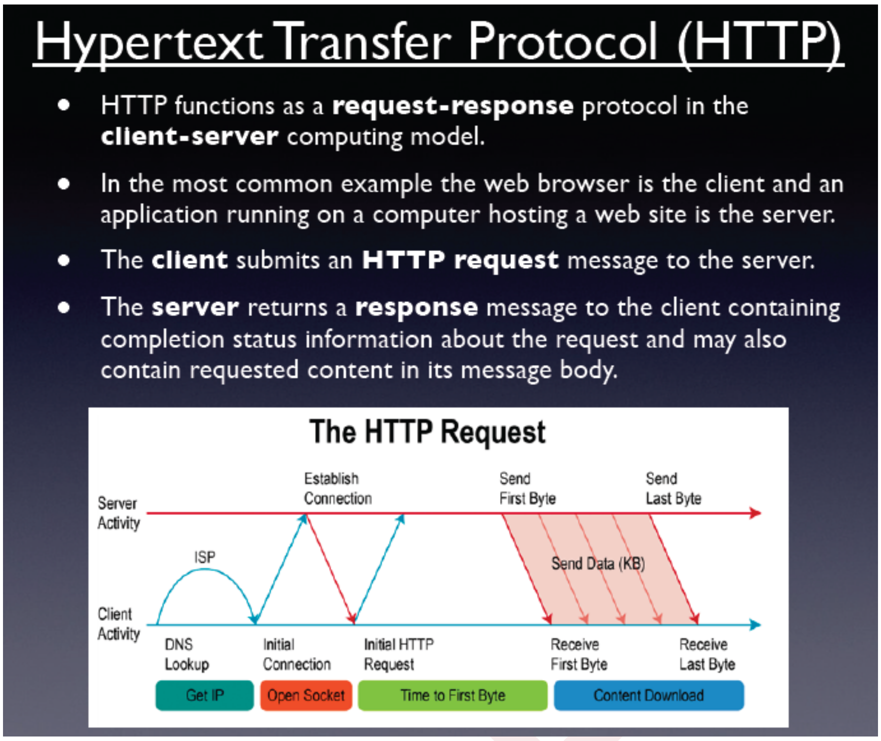

Figure 6: HTTP illustration (1)
https://www.slideshare.net/davidmichaelwallace/lecture-the-dynamic-web-2013

By default, HTTP uses TCP port 80. Other ports such as 8080 can also be used.

The three main HTTP message types are GET, POST, and HEAD.

Example:

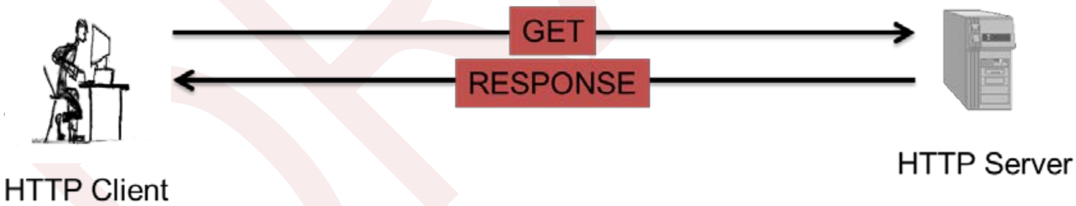

Figure 6: HTTP illustration (2)

In this example, a web browser running a client version of HTTP issues a request specifying the URL www.oceanario.pt to access a web page. On the other side, the machine that receives the request, is running a server version of HTTP and reacts to the HTTP request by sending an HTTP response which contains the desired document in the appropriate format.
When the information transaction is done, the TCP connection is closed and the user may view the document or the HTML page he asked for.

In brief:

- HTTP assumes messages can be exchanged directly between HTTP client and HTTP server.
- In fact, HTTP client and server are processes running in two different machines across the Internet.
- HTTP uses the reliable stream transfer service provided by TCP.

3. How an OTT Video connection works

When attempting to stream video in an un-managed network, routers, firewalls and ports involved in this connection are *a priori* unknown and might be configured in such as to prevent (or, at least, not facilitate) the flow of those streams. One way of overcoming these difficulties is by using the HTTP protocol for communication. HTTP uses port 80 for requests. Requests to this port are most likely allowed through any firewall or router as they are used for all web surfing.
Nowadays, several OTT streaming solutions use HTTP for signaling and data delivery.

3.1 HTTP Live Streaming Case (HLS)

Hypertext Transfer Protocol (HTTP) is a protocol used to deliver webpages and images across the Internet worldwide. HTTP is an adopted, open standard — the most ubiquitous mode of delivery online.
Concerning the delivery of OTT services, the technique commonly used consist in fragmenting a continuous stream into segments, encode these segments and make these available for download using plain HTTP methods. This is known as dynamic adaptive segment streaming over HTTP and is the enabler for deployed web infrastructure to be easily reused for live streaming.
HTTP Live Streaming (HLS) is based on the HTTP protocol and doesn't requires any streaming server because it can use an existing HTTP server, so all the switching logic resides on the player [1].
The video source is encoded into multiple files at different data rates and divided into chunks (file segments) usually with 10 seconds, but they can be between 2 to 10 seconds long, formatted as MPEG-2 transport stream. These chunks are loaded onto a HTTP server along with the text-based manifest file with the .m3u8 extension. This file directs the player to additional manifest files for each of the encoded streams. Switching bit rates on the fly is therefore not possible in the middle of an HTTP transaction, it is why the video is sliced up into "chunks"[2].

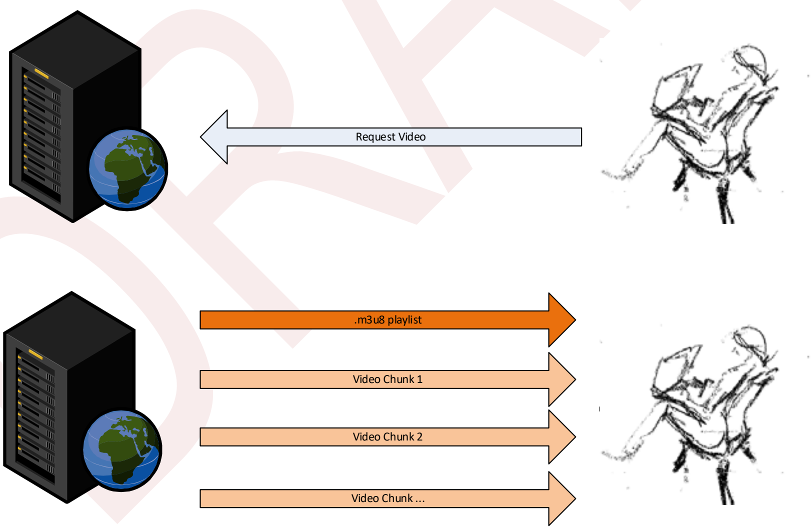

Figure 7 – Example of OTT video connection

When the client starts the player, he asks for the stream or video he wants to access in the server (Figure 7, (1)), answering this request the server sends a file with the .m3u8 extension, here called .m3u8 playlist (Figure 7, (2)). While parsing the playlist, the client learns about available bandwidths, media types, maximum media segment duration, information about if the stream is bounded or not, if there is alternative renditions of the same content or any other relevant information are also included in this file.

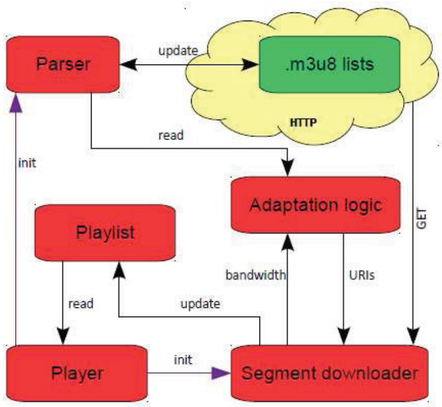

Figure 8 – HLS subsystem logic overview[1]

The chunk sizes are such that the reference IFrame at the beginning of each chunk is synchronized.
The delivery server can host several different bit rate encodings of the same video content. Each bit rate encoding has a separate play list, which is defined by a master playlist. These playlists are typically in an M3U8 format and contain the list of chunks in order. When the client detects either insufficient bandwidth or more available bandwidth, it can switch to either the lower or higher bit rate playlist and download the chunks in that list. Since each chunk is synchronized with the other bit rate streams, there is a seamless transition between them so that the video playback is not interrupted. This maintains a high quality user experience.

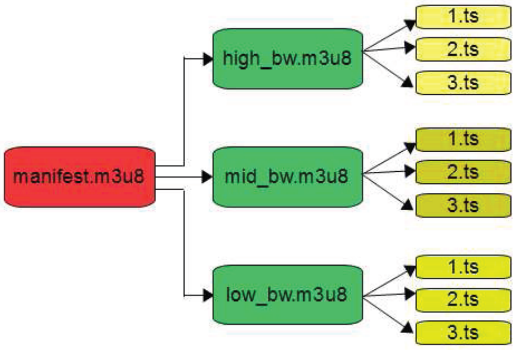

Figure 9 – Manifest Structure[1]

In HTTP segment streaming, the client has the responsibility to download the next segment before the previous segment, based on this information the client starts fetching the segments over HTTP GET request and starts streaming the content.
This solution is based on HTTP requests/responses at the level of one video fragment for each request/response, the client sends an HTTP request for a specific video fragment and receives the fragment via an HTTP response from the server. It is important to note that the server can send out a response only when an entire fragment has been published, as seen in Figure 10.

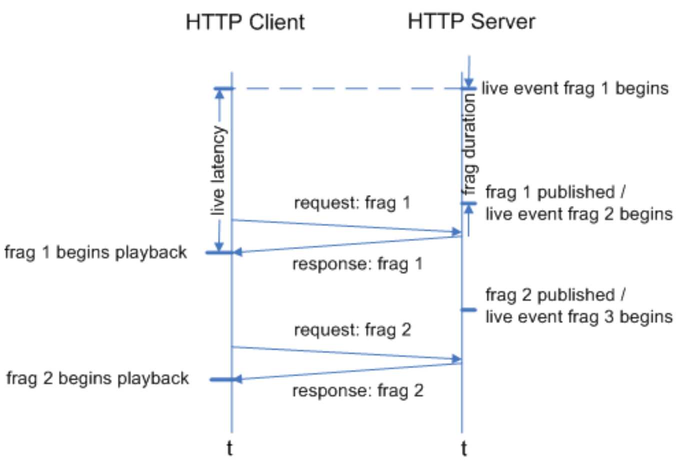

Figure 10 – Example of a fragment-based HTTP live video streaming.
The live latency is at least 1 to 2 fragment durations.[3]

Bandwidth consistency is a major issue. If a user is watching a video and someone else on the same network suddenly decides to perform a file transfer, the available bandwidth for the video can be severely impacted. In order to maintain a good Quality of Experience, content therefore needs to be encoded at different bit rates and the delivery protocol needs to be able to dynamically switch the bit rate with no interruption in playback or action by the user.
Since it is an HTTP connection, the data should pass through the same network path, and different network protocols, in the different layers, than for the webpage example described above.

3.2 RTMP Case

Real-Time Messaging Protocol (RTMP) refers to the proprietary protocol developed by Adobe Systems for streaming audio, video, and data over the Internet between a Flash player and a Flash Media Server.
RTMP belongs to the application-level protocol, and runs over TCP as transport-level protocol.
The basic unit of the RTMP to transmit information is Message. During transmission, for consideration of multiplexing and packetizing multimedia streams, each Message will be split into some Chunks.
In the process of playing a streaming media, the client can send Command Message such as “connect”, “createStream”, “play”, “pause” to control the playback of streaming media.
Message need to be split into a number of Chunks when it transmits data in the network. Chunk provides multiplexing and packetizing services for a higher-level multimedia stream protocol. RTMP Chunk Stream Protocol prescribes that the Payload of each Message is divided into fixed-size Chunks (except the last one).
Playing a RTMP-based streaming media, under normal circumstances, need to use the Flash application as client. User can use ready-made Flash web player to play streaming media.

3.2.1 Method of playing a RTMP Video

A RTMP-based video streaming need to go through the following steps: Handshake, Create Connection, Create Stream, and Play. Outlining the steps we have:

- The Handshake initiates the connection;
- Then the Create Connection step is used to establish the NetConnection between the client and server;
- The following stage is used to establish the NetStream between the client and server, called Create Stream;
- Play stage is used to transmit video and audio data.

3.2.1.1 Handshake

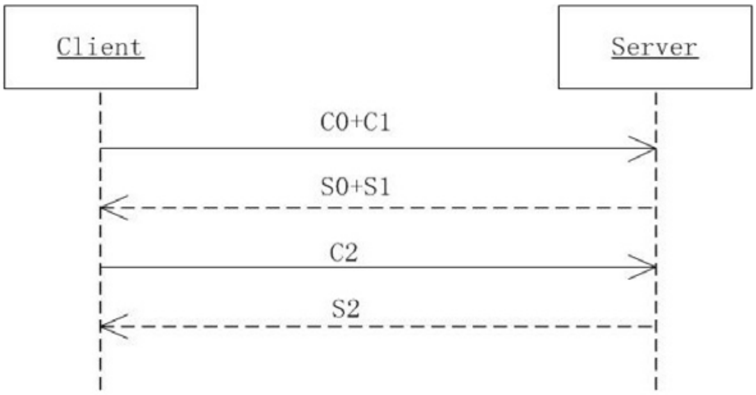

Figure 11 – Handshake [4]

1. The client sends C0, C1 block. Server receives the C0 or C1 and then sends S0 and S1.
2. When receiving all the S0 and S1, the client starts sending C2. When receiving all the C0 and C1, the server starts sending S2.
3. When the client received S2 and the server received C2, the Handshake is complete.

3.2.1.2 Create Connection

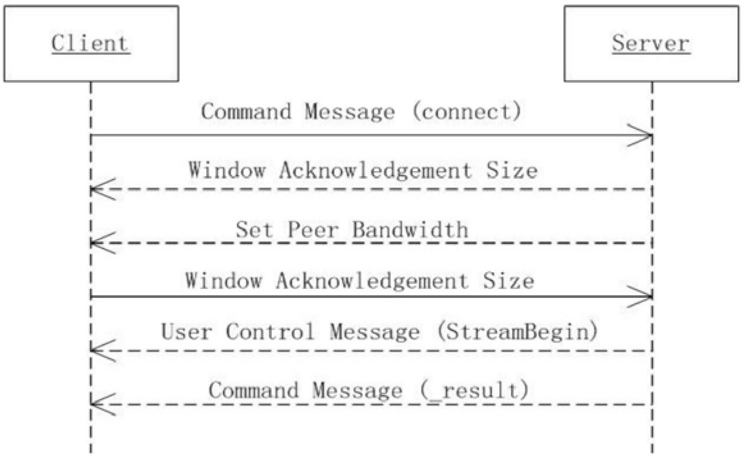

Figure 12 – Create Connection [5]

1. The client sends a Command Message "connect" to the server to establish a NetConnection with a server application instance.
2. After receiving the "connect" Command Message, the server sends the Message "Window Acknowledgement Size" to the client, and connect to the application mentioned in the Command Message.
3. The server sends the Message "Set Peer Bandwidth" to the client to update the output bandwidth.
4. After dealing with the set bandwidth Message, the client sends the Message "Window Acknowledgement Size" to the server.
5. The server sends the User Control Message "Stream Begin" to the client.
6. The server sends Command Message "_results" to notify the client the result of the Command.

3.2.1.3 Create Stream

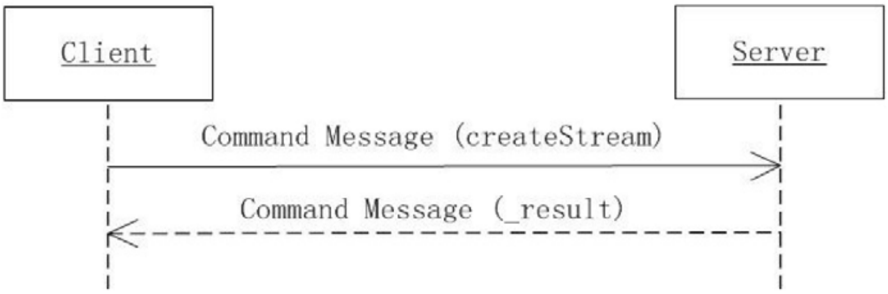

Figure 13 – Create Stream [5]

1. The client sends a Command Message “createStream” to the server to request to establish a NetStream with a server application instance.
2. The server sends Command Message "_results" to notify the client the result of the Command.

3.2.1.4 Play

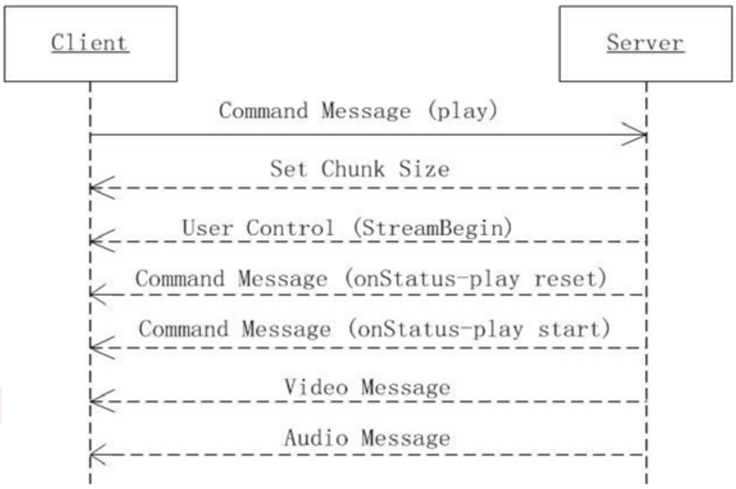

Figure 14 – Play [5]

1. The client sends the Command Message “play” to the server.
2. On receiving the “play” Command Message, the server sends “Set Chunk Size” Message to notify the client the chunk size used in the stream.
3. The server sends User control Message “StreamBegin” to inform the client that the stream has become functional.
4. The server sends Command Message “NetStream.Play. Start” and “NetStream.Play.reset” to notify the client the “play” Command is successful.
5. After this, the server sends audio and video data ,which the client plays.

3.3 Advantages and Disadvantages of HTTP vs RTMP

HTTP is less likely to be disallowed by routers, Network Address Translation (NAT), or firewall settings, this protocol uses the port 80 to communicate that is commonly open, so there’s no need to open other ports, because of this fact the content can be distributed to the client in more locations and without any special settings.

This protocol is also supported by more CDNs, a factor that can affect cost in large distribution models. In general, more available hardware and software works unmodified and as intended with HTTP than with RTSP or RTMP. Additionally, for large-scale events, HTTP natively and easily supports mirroring and edge caching, providing for massive-scale expansion when needed for the largest events. In the other hand RTMP can also be cached, but HTTP does so natively and without the need for proprietary or custom configurations.

Another advantage of HTTP is access. Some networks use firewalls to block specific content, the most popular methods to do so are protocol and port restrictions. Some firewall rules allow only HTTP content served over port 80, as said before. However, the default port for RTMP connections is 1935, a port that may not be allowed on tight firewalls. If the first attempt of the Flash Player to play video over port 1935 fails, it tries to reconnect using a few different methods. To summarize, HTTP streaming should be used to avoid dealing with firewalls and proxies.

One benefit with RTMP worth mentioning here is its ability to provide multicast support. If you manage your own network and want to deliver streams to many users without initiating a new connection for each user, RTMP is the best technology. HTTP does not provide this function, nor do CDNs.

4. Access Network Technologies

4.1 xDSL (xDigital Subscriber Lines)

xDSL technologies (where "x" stands for several possible initials as, for example, ADSL, for *Asymmetrical Digital Subscriber Line,* "V" for *Very*, etc) appeared as an attempt to extend the capabilities of existing copper infrastructure inherited from the (plain) old telephone service (usually referred by its acronym: POTS).

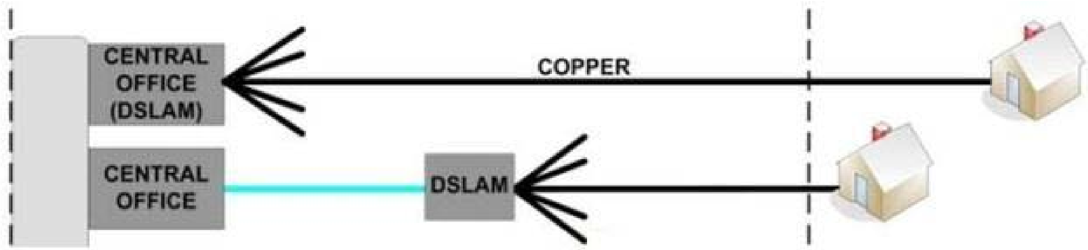

Figure 15 – xDSL Access Network [26]

The POTS network has been optimized to transmit signals in the range between 300 Hz and 3400 Hz (voice signals This bandwidth limitation (which was mainly imposed by the hybrid transformer and the coupling capacitors used to convert 2 into 4 circuits in telephone circuits, as illustrated by Figure 16) imposed great constraints on data transmission throughput.

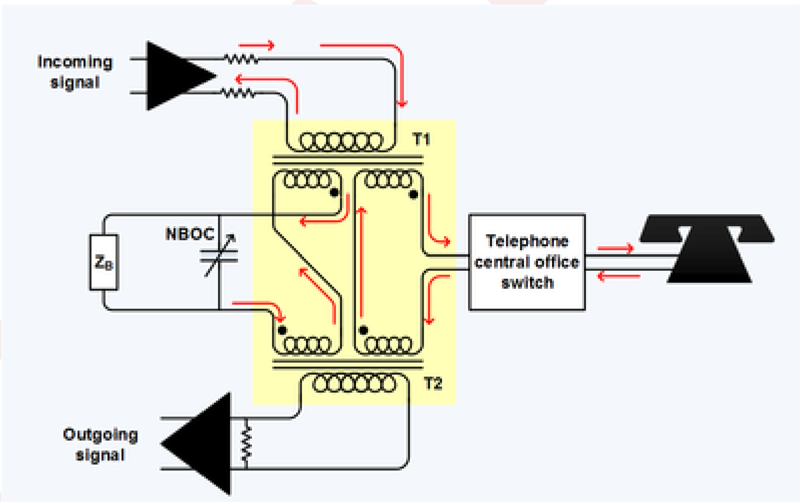

Figure 16 – Hybrid Transformer [26]

For a few decades dial-up modems were used to transform data signals into signals that occupied the same bandwidth of voice signals but the attainable data rates were only a few tens of kilobits per second (kbps).

To overcome the above problem of bandwidth in telephone networks, DSL (Digital Subscriber Line) technologies have emerged.

These technologies resort to frequency multiplexing of voice and data signals, as shown in Figure 17, taking advantage of the upper portion of the frequency spectrum that remained unused to transmit the data signals in that range.

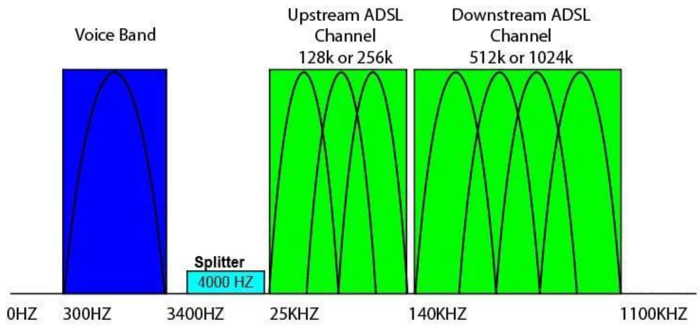

Figure 17 – ADSL Frequency Spectrum (picture in public domain)

The transmission rates attainable in this type of connections depends on the distance between the end user and the DSLAM (Digital Subscriber Line Access Multiplexer) as illustrated in Figure 18

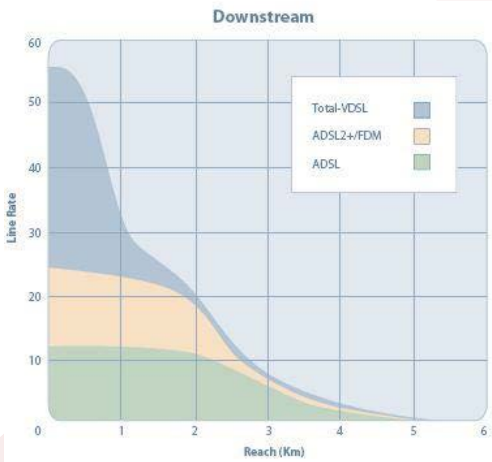

Figure 18 – Transmission rate (Mbps) versus distance (Km) of the client to the DSLAM (picture in public domain)

4.1.1 OTT over ADSL

In the Figure 19 – Transmission scheme of OTT content over ADSL, it's represented how an OTT stream based on HTTP protocols, e.g. HTTP Live Streaming, is transmitted through an ADSL Network.
The scheme represents the end-to-end transmission since the video server until the clients' device. This transmission can be done using the network of one or multiple operators, (Tier 2 operators) that will do the connection between the data center and the core network by (Tier 1 operators).
The OTT service is received by the end user through HTTP based streams. All the HTTP streams are received by the ADSL Router owned by the client and routed to the CPE where the client wants to use the service. This final step in the delivery can be done through Ethernet or Wi-Fi access network.

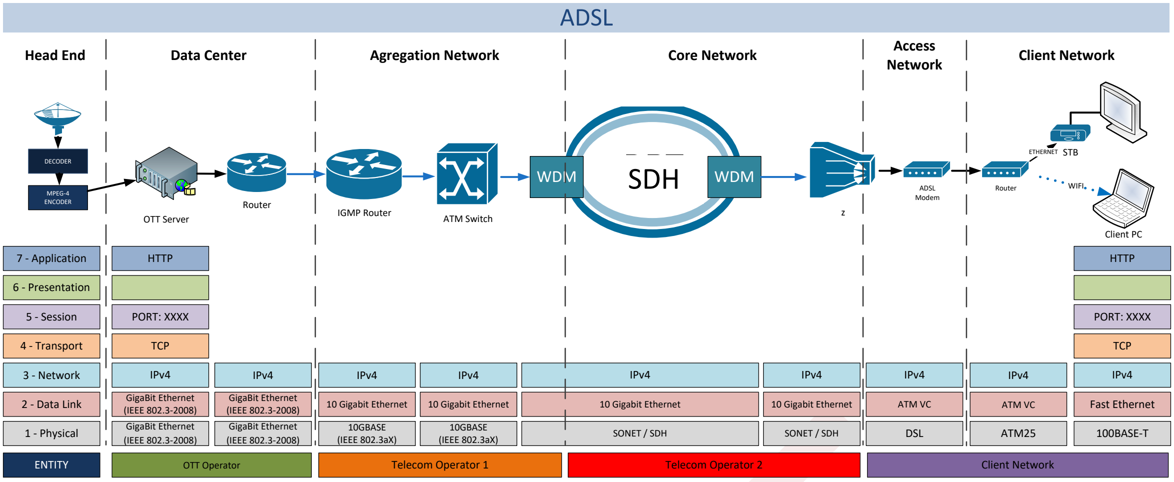

Figure 19 – Transmission scheme of OTT content over ADSL

4.2 *Fiber To The X (FTTx)*

Fiber To The X (FTTx) is a generic expression to describe different fiber optics based telecommunication networks. Depending on the termination point of the fiber optics, these architectures have various designations: FTTN (Fiber To The Node), FTTCab (Fiber To The Cabinet), FTTC (Fiber to The Curb), FTTP (Fiber To The Premises), FTTB (Fiber To The Building), e FTTH (Fiber To The Home).

- Fiber to the node (FTTN) or Fiber to the Cabinet (FTTCab), refers to a network architecture in which fiber is extended to a street-side or on-pole cabinet. These points are at a distance of approximately between 300m and 1500m from the user. From that point forward, xDSL technology or Ethernet (over copper or wireless) are used to reach the user. These architectures are suitable for small dimensions areas and low population density.
- Fiber to the curb (FTTC) is a network architecture where the optical fiber goes until a street cabinet, serving very small areas (about 300m radius) and low population density. Users connect through the existing infrastructure of copper or coaxial cables. This architecture differs from the FTTx architectures, since the cabinet street is nearer to the residence of the customer, while the FTTN architectures or FTTCab, the street cabinet is far away from the customer residence.
- Fiber to the building (FTTB), in this architecture the optical fiber reaches up to the entrance of the building, but it doesn't arrive directly to the users home. The connection to the end user is not made using optical fiber, but using other transmission means such as copper or coax.
- Fiber to the home (FTTH) refers to an architecture where commonly the optical fiber connects directly the end user. By definition, the fiber optic communication path is terminated on or in the premise for the purpose or carrying communications to a single subscriber.

4.2.1 OTT over FTTH

The main difference from the previous example of the OTT stream distribution scheme is the Access Network technology used. On FTTH, the HTTP stream is transported over a fiber access network, it means that the data leaves the Core Network through an OLT, then the signal is splitted until it arrives to the client's home and is received by an ONT. The ONT converts the optical signal in an electrical signal that is then connected to a router over an Ethernet connection. The ONT and the router may be integrated into the same hardware by some vendors.
Over the Access Network, the transmission is based on NRZ technology in the Physical Layer and then on ATM/GEM in the Data Link.

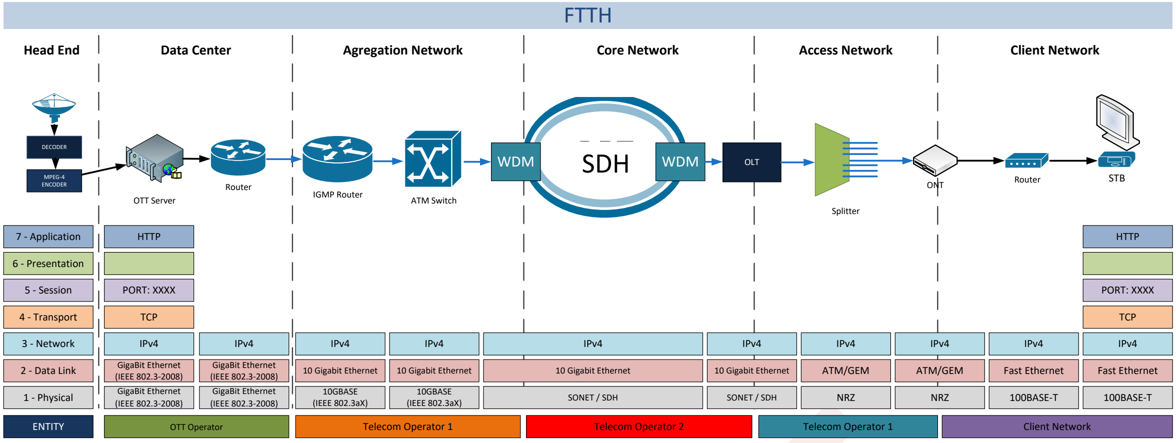

Figure 20 - Transmission scheme of OTT content over FTTH

4.3 HFC

HFC (Hybrid Fiber-Coax) networks appeared as an evolution of CATV networks. Cable networks or CATV networks were originally designed to broadcast video over coaxial cabling until the subscriber's residence. However these networks have evolved to a multi-service platform, offering not only TV broadcasts but a variety of telecom services, such as: FM radio programming, high-speed Internet, telephone, and others.
With this evolution the physical network had to evolve from a broadcast only model to a two way communication network, with separate user communication in order to ensure that user privacy is not compromised. This capacity has been achieved by the use of a new set of frequencies between 50 and 860MHz for downlink and between 5 and 65MHz on the uplink [8]. Each downstream/upstream data channel uses a 6MHz window.
The architecture of a hybrid fiber coaxial network uses fiber optic cables in the core network and coaxial cables in the distribution/access network, as seen in Figure 21.

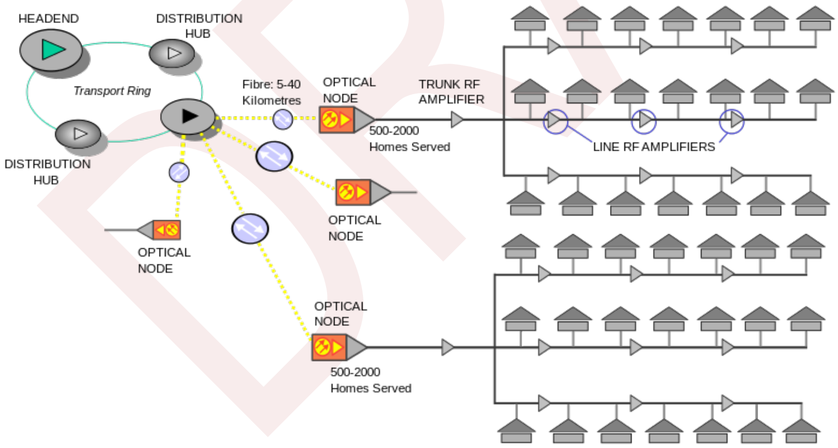

Figure 21 – HFC Network Diagram [8]

An advantage of these networks is that some of the characteristics of the fiber optic cable, like low noise and interference susceptibility (apart from the obvious high bandwidth), can be brought closer to the user without having to replace the installed coaxial cable that goes until the subscriber's home.
The signal is composed at the head-end, were the television signals are received, they are then encoded and finally injected into fiber optic cables. The broadcasted channels are received via satellite or DTT. The signal is transported via optical networks until the distribution centers, were the optical signal is converted in electrical and finally distributed via the coaxial network until the subscriber's home.

In order to adapt the HFC networks for interactive services and normalize supply, ITU-T adopted in 1998 the Data Over Cable Service Interface Specification (DOCSIS) as standard ITU-T J.112 that enables interoperability and access to data services.

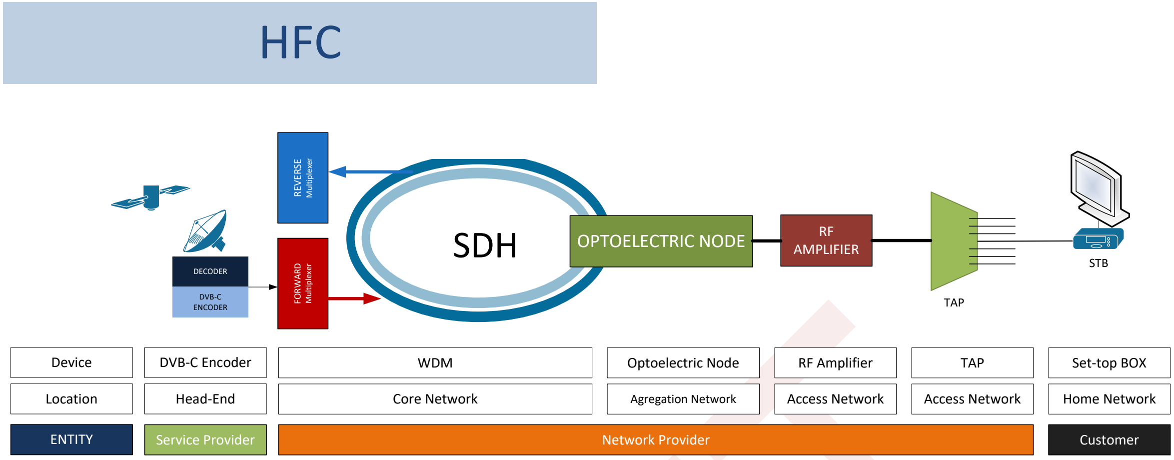

Figure 22 – Hybrid Fiber Coax Network Overview

DOCSIS specifies methods for transporting data over CATV networks using QAM and/or QPSK RF modulation techniques. A DOCSIS architecture includes two primary components: a cable modem (CM) located at the subscriber's location, and a cable modem termination system (CMTS) located at the CATV head-end. Cable systems supporting on-demand programming use a hybrid fiber-coaxial system. Fiber optic lines bring digital signals to the nodes in the system where they are converted into RF channels and modem signals on coaxial trunk lines, making it a point-multipoint communication system between the CMTS and the subscribers CMs. The CMTS is similar in function to a DSLAM used in xDSL systems. The number of users served by a node will have to take into consideration: thermal noise, ingress noise, common path distortion, etc.

According to ITU-T recommendation J.222.1, these networks are defined by:

- Symmetrical Transmission (upward and downward);
- The maximum distance between the cable modem termination system (CMTS) and the cable modem (CM) is 160km in each direction, although typical maximum separation is 15-24km.

4.3.1 OTT over HFC

In this case since the HFC network is based on DOCSIS, the OTT streams are distributed in the Access network over Ethernet technology, as can be seen in Figure 23.

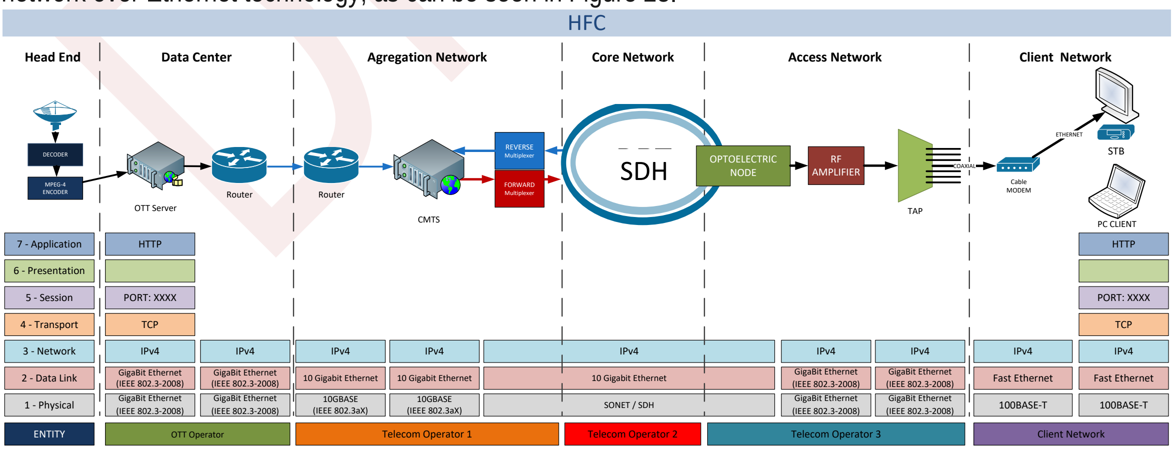

Figure 23 - Transmission scheme of OTT content over HFC

4.4 Mobile Networks

Nowadays there is increasingly more need to access different kind of services or data anywhere and anytime, only mobile networks provide this ability to the user. The importance of mobility led to the great development of mobile telephone networks and then mobile data networks.

4.4.1 GSM

The GSM (Global Systems for Mobile Communications) network is the most used mobile telephone network across Europe. This telecommunication system has the ability to transmit voice, data and message services among other supplementary services such as call forwarding or calls suspension. This network allows transmission rates up to 14.4 kbps. The GSM system made the transition from analog technology to digital technology, bringing improved security, robustness and reliability.
OTT video couldn't be transmitted over GSM networks because of its low transmission rates.

4.4.1.1 GPRS

GPRS (General Packet Radio Service) is an evolution of the GSM system, which introduced the transmission of data with packet switching. The GPRS network is implemented on the GSM infrastructure and keeps most of the network equipment and acts as a supplement to this network providing enhanced data services. Now there are two parallel networks: the GSM network responsible for voice traffic and the GPRS network responsible for the data traffic (packet switching). This system allows transmission rates up to 171 Kbps.
OTT video could be distributed over GPRS, but only the lowest profiles because of its limited transmission rates.

4.4.2 UMTS

The UMTS (Universal Mobile Telecommunications Systems) network is one of the third generation's mobile access technologies. It was designed in order to continue the success of GSM and then GPRS technology, providing higher access speed to data services. The UMTS data service supports from 144 Kbit/s (for mobile access) up to 2 Mbps (for a fixed wireless access). W-CDMA (Wide-band Code-Division Multiple Access) and CDMA2000 (Code Division Multiple Access) are modulations used in UMTS. This technology enables easy interconnection with other telecommunications systems, such as the PSTN or other data networks, allowing the user to move between different environments.
A UMTS system can be based on already existing mobile communication system and have radio equipment capable of accommodating systems such as GSM, GPRS, EDGE (Enhanced Data rates for GSM Evolution) and UMTS, in order to ease the transition from GSM to UMTS. [30]

4.4.2.1 HSPA

High-Speed Packet Access (HSPA) is a set of technologies that defines the migration path for 3G/WCDMA operators worldwide. This technology was standardized by the 3GPP, it uses the FDD transmission scheme and includes the variants: HSDPA (High Speed Downlink Packet Access), HSUPA (High Speed Uplink Packet Access) and HSPA Evolved. Unlike UMTS, HSPA provides very efficient voice services in combination with mobile broadband data, consequently filling the UMTS broadband gap allowing the user to enjoy speeds of at least 1Mbps on the uplink and 14.4 Mbps on the downlink. HSPA Evolved introduces Multiple-Input Multiple-Output (MIMO) capabilities and higher order modulation (64QAM), enabling greater throughput speeds of up to 21Mbps on the downlink.
This technology was developed to cover a flaw existing in UMTS networks, i.e. to make the link between 3G mobile network and Internet services, allowing to overlay various protocols that enable high-speed data communications to several users served by same cell.

4.4.3 LTE

Long Term Evolution (LTE) is a 4G wireless broadband technology developed by the 3GPP and it represents an evolution of the mobile access technology from GSM, a 2G standard, to UMTS, the 3G technologies based upon GSM. This technology is also known as Evolved UMTS Terrestrial Radio Network (E-UTRAN).

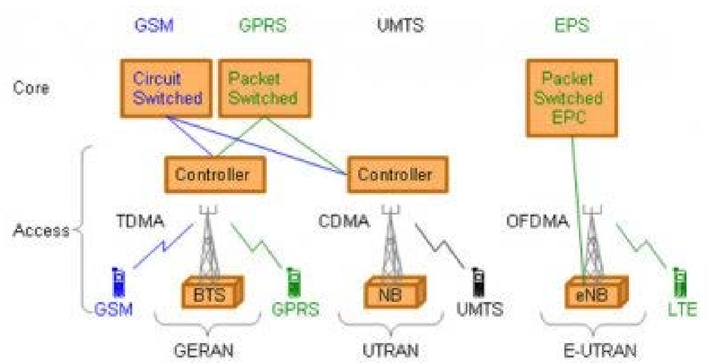

Figure 24 – Mobile Network Evolution from GSM to LTE [31]

The capacity of each sector is substantially increased improving the bit rate and mobility of each end use, leading to a lower latency in the network. With the rise of the IP protocol as a transport protocol carrying all types of traffic, LTE upper layers are based upon TCP/IP which results in an all-IP network with point-to-point QoS. LTE supports mixed data, voice, video and messaging traffic, they all run over IP, for example the voice service will be supported by VoLTE (Voice Over LTE).
LTE uses OFDM (Orthogonal Frequency Division Multiplexing) and MIMO (Multiple Input Multiple Output) antenna technology, similar to that used in the IEEE 802.11n wireless local area network (WLAN) standard. The higher signal to noise ratio (SNR) at the receiver enabled by MIMO, along with OFDM, provides improved coverage and throughput, especially in dense urban areas where signal is harder to propagate.
It is expected that this technology can achieve peak data rates of around 100 Mbit/s upward and 50Mbit/s downward, these maximum values for optimal conditions that can hardly be achieved in commercial wireless networks today.

4.4.3.1 OTT over LTE

LTE is the most recent wireless access technology and the broadband available is the ideal to access OTT multimedia content. Today LTE offers a broadband downlink of approximately 100 Mbps.
In Figure 25, it is illustrated the transmission of an HTTP video stream over a LTE access network. The last hop of the stream goes through an all IP network, based on IPv4 and Ethernet over a RF link.
In LTE when data flow of information leaves the core network, enters the E-UTRAN Networks and is routed by a Radio Network Controller (RNC), than the information is transmission over the air through an RF link by an eNodeB.

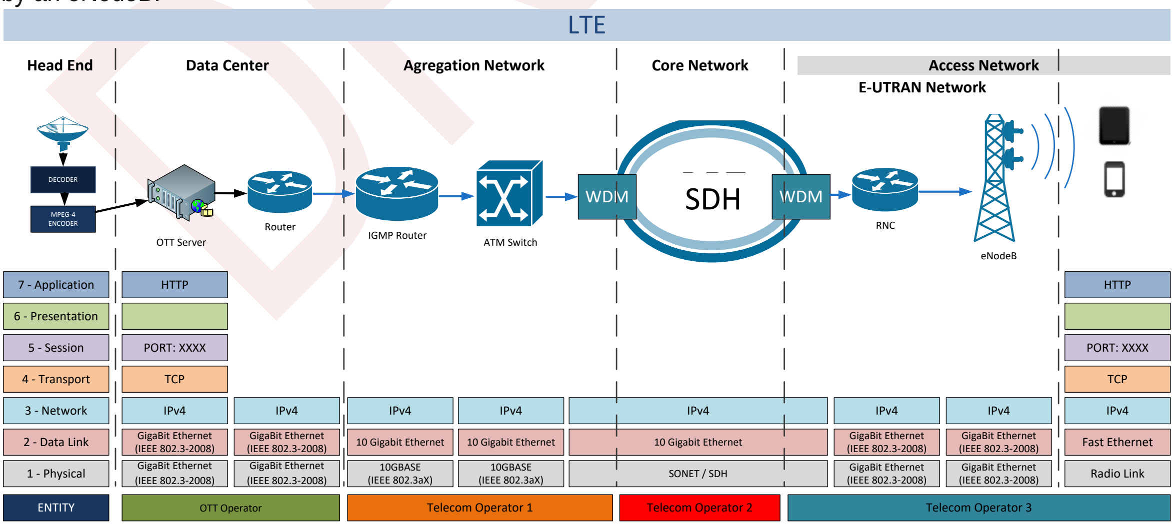

Figure 25 - Transmission scheme of OTT content over LTE

We can have multiple telecom networks involved in the transmission of OTT streams. We can have a first Telecom Operator that the OTT service provider contracted to connect him to the Internet. Telecom operator 1 can be connected to a Tier 1 operator (Telecom Operator 2) to transmit information to the access telecom operator (Telecom Operator 3) where the client is connected. It means that the stream flow can go through

multiple operators in the whole end to end connection. In some cases the whole transmission can also be done by only one operator.
Typically Tier 1 operators are operators who transmit high quantities of data between telecom operators.
This multiple telecom operator end-to-end concept also applies to the other access networks already described, where an OTT video can be streamed.

4.4.4 WiMAX

WiMAX (Worldwide Interoperability for Microwave Access) is a wireless technology and it's defined according to the IEEE 802.16 standard. This access network technology is intended as an alternative to xDSL or cable in the last mile access.
This technology has a much greater range than Wi-Fi (IEEE 802.11), providing wireless broadband access coverage up to 50 km for fixed stations and 5-10 km for mobile stations with the same performance of Wi-Fi but with the same coverage and quality of service as a traditional cellular network. It works in the 2 to 66 GHz range and enables connectivity without a direct line-of-sight to a base station, providing data rates up to 70Mbps.

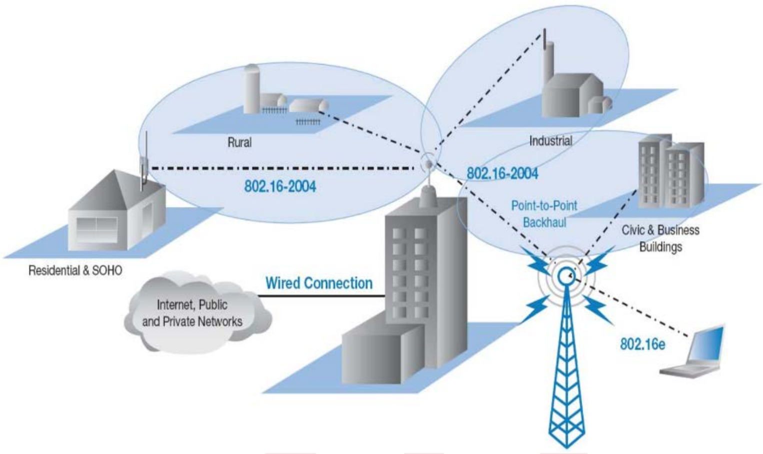

Figure 26 – Fixed WiMAX deployment and usage models

However, the available bandwidth is also shared by all the users that are connected to the network simultaneously, so greater the number of users, smaller the bandwidth available for each.
Wireless networks offer some advantages over wired ones because they can be helpful to connect remote areas, where wired networks are not yet installed or are too expensive to deploy. We can see this happen in some developing countries, where WiMAX is being adopted in areas that had no previous broadband infrastructures.

4.4.5 Wi-Fi

The Wi-Fi technology was developed to provide wireless short range, giving users greater convenience in their daily lives. This technology is generally used for distances of 30 meters indoors and 90 meters outdoors. Transmission rates evolved over the years with many amendments introduced into the original standard and today we can achieve connection speeds up to 300 Mbps (using IEEE 802.11n, the fastest standard in optimal conditions [9]). However, under "normal use" it operates at lower speeds, probably around 130Mpbs or less. These speeds are, mainly influenced by the number of users on the network (shared medium, shared timeslots) and on the number of different Wi-Fi networks on the same physical space (radio signal interference).
This technology is viewed as a complement and an essential part of the Home Network and is wide spread and well established over the world.

4.5 IPTV

IPTV – Internet Protocol Television - is a technology that uses Internet Protocols (IP) to deliver television services through packet switched networks, instead of other traditional networks such as terrestrial broadcast, satellite signal, and cable television formats. The official definition approved by the group focused on IPTV of the International Telecommunication Union (ITU-T FG IPTV) is:

"IPTV is defined as multimedia services such as television/video/audio/text/graphics/data delivered over IP based networks managed to provide the required level of quality of service and experience, security, interactivity and reliability." [11]

Nowadays IPTV services aren't used to deliver only television channels, but they deliver also a large amount of other contents, such as applications, games, information content, radio streams, among others. This service being usually part of Triple Play bundles, including also voice and data services it becomes a challenge for telecom operators, where they have to provide these services through their already existing networks with high standards of quality of service (QoS).

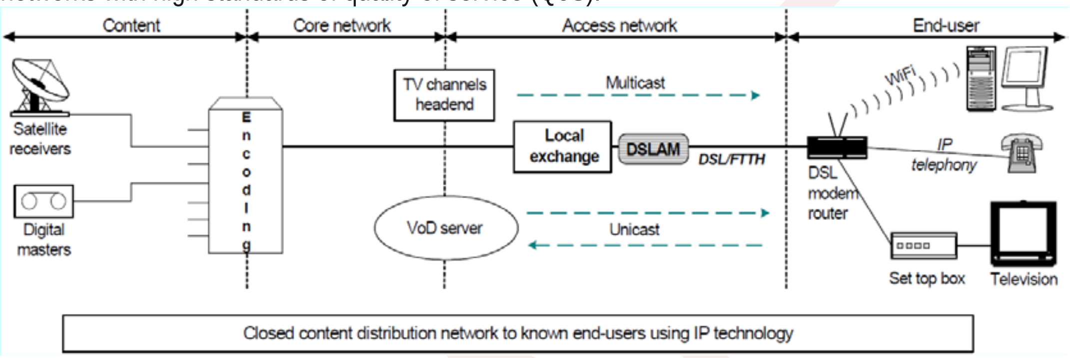

Figure 27 – IPTV network [10]

The main features of this technology are:

- Support for interactive television - the ability to transmit information in both directions, server/client and client/server, allows IPTV service providers to offer a larger quantity of interactive television applications such as video on demand
- Customization - an IPTV system, through its bi-directional communication allows users to personalize their television content in order to see what they want and when they want to see, according to their interest programming this can be achieved with on demand content.
- Optimized bandwidth management - instead of sending all channels available to all users, the IPTV technology allows service providers to send only the channel requested by the user. This allows network operators to save a lot of bandwidth on their networks.

In order to take advantage of the already existing copper networks, the operators have improved the efficiency of these to be able to quickly provide the contents to the end user without errors. One of the reasons for the increasing need of more bandwidth is due to the size of the content that is distributed. Video data requires large storage space, so if we want to transmit this data in the shortest time possible we need a higher rate of transmission and consequently more bandwidth.

The IPTV architecture evolution can be summarized through the following steps:

1. IPTV architecture not based on next-generation networks - the first generation of IPTV architecture consisted in one IPTV headend and middleware platforms for distribution services. This is the solution that is currently implemented in the IPTV market. You can interact with this architecture subsystems NGN (Next-Generation Networks) but generally the service control is done separately and is used a new application layer.
2. IPTV architecture for next generation networks not based on IMS - allows interaction at specific points between IPTV functions (such as control functions) and some existing elements of next generation networks (such as control elements of transport). In this step, a dedicated IPTV subsystem is used to provide all the IPTV functionality (IPTV control and user management) to integrate IPTV components in NGN architectures.
3. IMS-based IPTV architecture - specifies IPTV functions based on subsystem IMS (IP Multimedia Subsystem), and allows reuse of IMS functionalities, initiation of services and control mechanisms based on SIP (Session Initiation Protocol).

4.5.1 IPTV Distribution over ADSL and FTTH Networks

The distribution of IPTV over ADSL or FTTH networks can be summarized in in Figure 28 and Figure 29, and described by:

- IPTV uses RTP (application layer) over UDP (transport layer);
- The signal goes normally through 3 different Networks:
  - Service Provider network: usually inside the data center or between the head-end and the data center. This network is used to acquire, encode and broadcast the content.
  - Network Provider: since we're talking about a service where it's mandatory to own a managed network to guarantee quality of service. The network has to be owned practically end-to-end by the IPTV operator.
  - Customer or Home Network: owned by the client inside its premises and usually installed by the network and/or service provider.

The main difference between the distribution of IPTV over ADSL or FTTH lies on the access network:

- Because the ADSL (Figure 28) is based on the old POTS technology it offers slower speeds in the access network, this can restrict the access to channels with HD quality. In order to maintain quality IPTV reserves a content bandwidth to deliver the TV channels, usually 4 Mbps for SD channels, this normally interferes with the customer internet signal since they are sharing the same network. So if we have 2 SD signals over an ADSL network with a top speed of 16 Mbps, we are consuming with the IPTV service half of the bandwidth (8Mbps).
- FTTH is fiber based (Figure 29), and offers higher access speeds. Nowadays the access speed is set to 100 Mbps, more than 4 times the offered by ADSL. With this technology we've got no constraints offering multiple HD and SD signals.

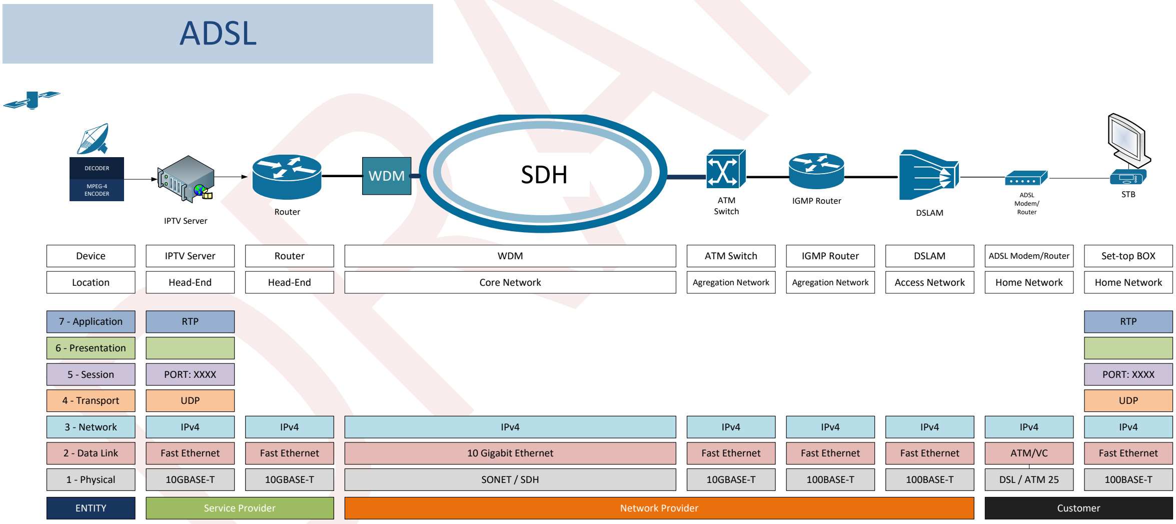

Figure 28 – IPTV transmission scheme over ADSL Networks

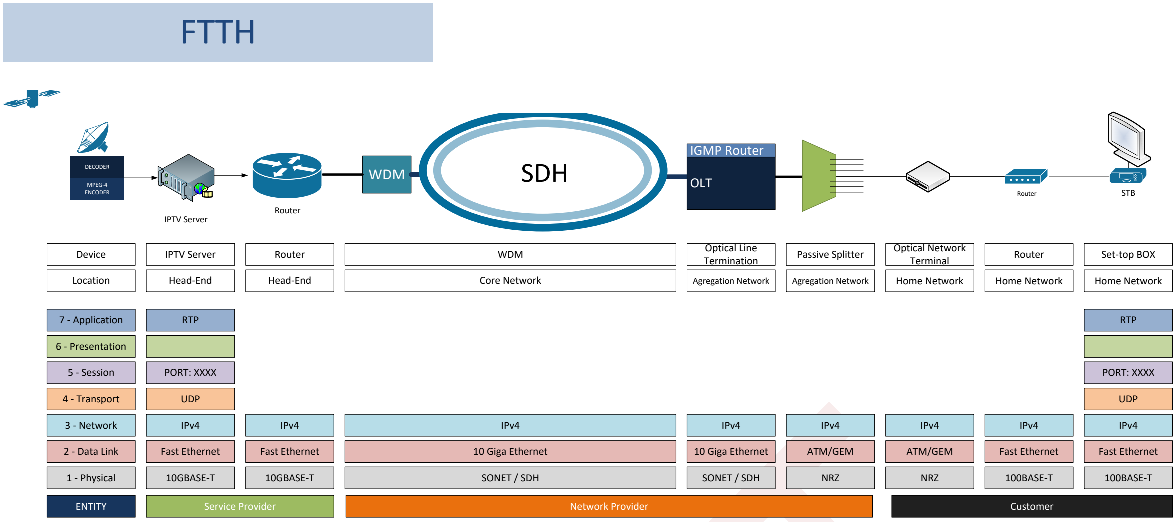

Figure 29 - IPTV transmission scheme over FTTH Networks

4.5.2 IPTV vs. OTT

IPTV and OTT are two technology mediums to distribute television and video over IP networks. However the main difference between them is that IPTV was designed to be used over a managed network, the service provider has to own the infrastructure in order to provide the service; and OTT was intended to deliver video over unmanaged networks.
IPTV had formerly the advantage of quality control over the video delivery since it was running over a managed network, but nowadays QoS over OTT can be controlled with content management systems and advanced CDN solutions.
The main advantage of OTT over IPTV is that it was designed to reach any connected device, giving a key competitive advantage in terms of customer and device reach.
Other differences between these two technologies are resumed in Table 2 below.

	IPTV	OTT
Type of network	Managed (“walled gardens”)	Un-managed (open internet)
Network ownership	Service Provider has to own the network	Service provider may or not own the network
Quality of Service	Guaranteed (control over quality can be easily achieved)	Guaranteed if some delivery techniques are used such as Adaptive Streaming and CDNs.
Protocols	Transport Streams (TS) over UDP	Mainly based on HTTP over TCP
Routing Topology	Multicast	Unicast

Table 2 – Comparison of IPTV vs. OTT]

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