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Title: Study and implementation of Hybrid broadcast broadband Television on DVB-T2 platform in an environment with or no broadband coverage

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Dedication

This thesis is dedicated to:

- my Father and my Mother, who are no longer in this world,
- my family,
- my Sister Hazin and my brother.

Abstract

In accordance with the transition from analog to digital television stated at Geneva in 2006, the present work got into digital television system description especially DVB-T2 standard that was recently developed to replace the first DTT standard DVB-T.

We firstly tackled the major principles of DTT system to show how this system can be a response to spectrum scarcity. The techniques in use to achieve this goal were developed throughout digital data compression, statistical encoding and multiplexing method.

Digital modulation types were run down as well as their application domain. In this regard QAM modulations with different sizes were describe in order to present the limits and performance of one to another. OFDM technique, as multipath modulation scheme was described to point out the uses of Fourier Transform and Invert Fourier Transform that determines OFDM subcarriers. Assuming that DTT can be implemented with different modes related to FFT size, a summary of DTT modes have been shown through some figures with normal and extended size.

Some other major parameters such Guard Interval, data bits rate, pilot patterns and FEC were described and explained how they must be chosen when designing DVB-T2 network.

In the second time, we focused on transmission channel, mixing radio-link and satellite, to deliver linear broadcasting content. Here the technical solution of T2-Edge was discussed to avoid destruction of T2-MI signal.

The new requirement of consumers using ICTs devices is mobility and interactive capability. Taking that into account, we have designed a DVB-T2 platform architecture in combination with interactive platform. Hybrid broadcast broadband (HbbTV) technology was explained as interactive application solution.

The DVB-T2 network was designed discussing the number of national digital programs and regional programs. From there we evaluated the required bit rate for the designed DVB-T2 network. Assuming that the project must be implemented in Burkina Faso, the challenge to overcome was to deliver HbbTV content in areas with no broadband coverage. For that a technical solution based on satellite transmission channel has been described and proposed.

Lastly DTT receivers issue has been pointed out in terms of technical specifications requirement and their accessibility by consumers.

We concluded this work stating the needness to take care to the deadline of DTT network implementation because of technology fastness changes; we also draw the attention of rulers on the fact that Digital Radio Broadcasting is still awaiting so that DTT network must take in account the future advent of DRB.

Keywords:

DVB-T2, Transmission channel, DTT, OFDM, bits rate, QAM modulation, encoding,

Caution:

This work is a summary of researches from e-book reading, internet sources and confidential data acquired form resource bodies. It contains some administrative information that cannot be released or published without a special authorization from the author. We highly recommend one to respect author rights according to local and international laws.

Summary

Index & abbreviations

ACE	Active Constellation Extension
AIT	Application Information Table
APSK	Amplitude Phase Shift Keying
ASI	Asynchronous Serial Interface
AVC	Advanced Video Coding
BCH	Bose Chaudhuri Hocquengham
BER	Bit Error Rate
CA	Conditional Access
CAS	Conditional Access System
CDN	Content Delivery Network
CE-HTML	Consumer Electronic Hyper Text Markup Language
CI	Common Interface
CMS	Content Management System
C/N	Carrier to Noise Ratio
COFDM	Coded Orthogonal Frequency Division Multiplex
CP	Cyclic Prefix
CR	Coding Rate
DAB	Digital Audio Broadcasting
DASH	Dynamic Adaptive Streaming over HTTP
dB	decibel
DCT	Discrete Cosine Transform
DFT	Discrete Fourier Transform
DRM	Digital Rights Management
DTT	Digital Terrestrial Television
DVB	Digital Video Broadcasting
DVB-S2	Digital Video Broadcasting Satellite Second Generation
DVB-T	Digital Video Broadcasting
DVB-T2	Digital Video Broadcasting Second Generation
EIT	Event Information Table
EPG	Electronique Program Guide
ES	Elementary Stream
ETSI	European Telecommunication Standardisation Institue
FDM	Frequency Division Multiplexing
FEC	Forward Error Correction
FEF	Future Extension Frame
FFT	Fast Fourier Transform
GI	Guard Interval
GSE	Generic Stream Encapsulation
HEVC	High Efficiency Video Coding
HbbTV	Hybrid broadcast broadband Television
IBI	Inter-Block-Interference
ICI	Inter-Carrier-Interference

ICT	Information and Communication Technology
IDFT	Inverse Discrete Fourier Transform
iDTVs	integrated Digital Television receivers
IFFT	Inverse Fast Fourier Transform
IP	Internet Protocol
IRD	Integrated Receiver Decoder
ISI	Inter-Symbol-Interference
ITU	International Telecommunication Union
JPEG	Joint Photographic Expert Group
Kbps	Kilobits per second
KW	Kilowatts
LDPC	Low Density Parity Check
Mbps	Megabits per second
MCR	Master Control Room
MER	Modulation Error Rate
MFN	Multi Frequency Network
MUX	Multiplexer
MPEG 2/4	Moving Pictures Experts Group Layer 2/layer 4
MISO	Multiple Input Single Output
MPTS	Multiple Program Transport Stream
NIT	Network Information table
NRZ	Non-Return to Zero
NRZI	Non-Return to Zero Inverted
OFDM	Orthogonal Frequency Division Multiplexing
PAPR	Peak to Average Power Ratio
PLP	Physical Layer Pipe
PP	Pilot Pattern
PSI	Program Specific Information
PSK	Phase Shift Keying
PTS	Program Transport Stream
QAM	Quadrature Amplitude Modulation
QPSK	Quadrature Phase Shift Keying
RLE	Run Length Encoding
RQD	Rotated constellation and Q Delay
RS	Reed Solomon
TV	Television
SD	Standard Definition
SDI	Serial Digital Interface
SI	Service Information
SFN	Single Frequency Network
SPTS	Single Program Transport Stream
STB	Set-Top-Box
STDM	Statistical Time Division Multiplexing
TDM	Time Division Multiplexing
T2-MI	T2 Modulator Interface
TI Block	Time Interleaving Block
TFS	Time Frequency Slicing

TR	Tone reservation
T_u	OFDM subcarrier period time
UHD	Ultra High Definition
UHF	Ultra High Frequency
VHF	Very High Frequency

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General Introduction

After the industry revolution in XIXth century, the XXI ^{rst} century is the one that can be considered as the century of ICT (Information and Communication Technology).

Indeed, upon the introduction of digital application in many professional sectors, particularly in data processing through a computer, the development of information and communication technology is continuously growing. Consumer new requirement likewise is permanently growing, the specific needs of consumers as well as manufacturers and broadcasters led to high spectrum consumption. Indeed all the innovative communication and applications are frequencies based. Unfortunately, radiocommunication spectrum is limited and must be used efficiently.

With this regards, the agreement of Geneva, in June 2006 within the member states of ITU, it was decided the transition from analog television broadcasting to digital television transmission, by 17th of June 2015 in UHF band range and by 17th of June 2020 in VHF band range in region 1 [1]. The aim of that, was first to free up a number of frequencies being used by analog television broadcasting which principle relies on single channel, single frequency to deliver a single TV program. Whereas digital television transmission will be based on multiplexing many TV programs to broadcast on a single frequency, in a single television channel. Secondly, the digital television transmission over terrestrial infrastructures will provide to consumer, more TV programs with high pictures and sound quality.

The freed up frequencies, due to digital television transmission, will be reused to develop other ICT sectors and reply to consumers' specifics needs, particularly, the need of mobility and the need of interactivity.

The combination of digital television transition and consumers new needs, led to a number of technical studies and researches which aim to reply as closer as possible, to those specifics needs. From this statement, across the world, studies and implementation of digital television transmission were running down with different DTT standards. The main relevant DTT standards in use across the world wide are: ATSC used in American's region, DVB-T, in Europe and assimilated African countries, ISDVB-T in Japan and T-DMB in Korea and other Asian countries.

Taking into account the above mentioned standards, the present thesis aims to study and implement the delivery of Hybrid broadcast broadband Television over a DVB-T2 platform, in areas with or without broadband coverage. The interest of that is: if it can be considered that in Europe, American states and Asian states, DVB-T implementation is fully or partially accomplished, it is not the case in African states especially in West African states comprising Burkina Faso my Homeland, where the year 2016 will be very critical for DTT deployment.

It is known that HbbTV delivery contains linear and non-linear contents where linear content can be transmitted together with broadcasted content, unfortunately for non-linear content and interactive application, broadband coverage is required. However, the present thesis title is to be implemented in an environment where broadband does not or weakly exist.

The so indicated environment is Burkina Faso national territory: A West African country, surrounded in the north by Mali, in the south by Ghana, Togo and Benin; in the Ouest by Cote-d'Ivoire and in the Est by Niger. The estimated Population in 2014 was 17 million habitant, with 70 % of youth across the country. Those 70% are the most active in ICT devices operation. Therefore, the pressure put on telecommunication or broadcasting operators in order to ensure their needs, is very high and permanent. Starting from this observation, it becomes essential whatever the territory is not fully covered with broadband, to design a DVB-T2 network in combination with HbbTV 2.0 standard to deliver broadcast content and HbbTV linear and non-linear content enabling interactivity. This is the challenge that we are facing to, in writing the present thesis. The following discussions will bring out the procedure to design and implement an efficient HbbTV.2.0 platform, built on DVB-T2 network to deliver both broadcast and non-linear content for about 98% of Burkina Faso national territory coverage.

CHAPTER I: Digital television transmission over DVB-T2 system

I. DVB-T2 system for television programs signal transmission

I.1. General Definitions

❖ **Definition of Television**

Television is defined as an electronic communication mean, based on distance transmission of pictures and sound: Therefore, the information to transmit is firstly produced and process into convenient television signal format in a studio, before being transmit over various existing transmission means such as radio waves, optical fiber, cable, to a transmitter where, the received TV studio signal is finally processed according to the transmission channel parameters, then broadcasted by appropriated antennas towards TV receives located in the coverage area of the transmitter.

Thereby, Television is defined as a string of activities comprising production, transmission and reception.

Implementation of television based on the above described activities is typically conventional analog television broadcasting system.

❖ **Definition of Digital television signal**

Digital Television signal refers to as digital data that can be broadcasted to a TV set via terrestrial transmitters, satellite or cable: Digital signal or data are obtained by converting analog signal into digital one. The process that leads to digital signal conversion is called digitizing; this process comprises three main operations:

- Sampling
- Quantizing
- Compression

Sampling is a process which consist of cutting out into bits or samples in a regular period of time, an analog signal using a sampling frequency (F_S)

Quantizing consist of allocating a voltage value to a single sample took at instantaneous time (t_S) corresponding to the period of sampling: The allocated decimal voltage value is then convert into binary digit that is called binary coding and represented by digits (0 or 1)

Digital compression consists of analyzing picture sources to find and remove redundancy both within and between picture frames. This technique enables reducing space for digital data storage and data transmission bit rate.

- Source coding: The entire process implying sampling, quantizing and compression is named Source coding.

❖ **Definition of Digital Television (DTV)**

Digital Television is a combination of techniques and technology to ensure the transmission of audio and video signals that are digitally processed then multiplexed into TV programs signal, to deliver digital pictures and sound to consumer TV sets.

The means used to deliver digital content can be either, terrestrial transmitters, satellite, or IP TV distribution.

I-2. Digital Terrestrial Television (DTT/ DTTV)

Both abbreviations DTT and DTTV are employed to define Digital Terrestrial Television programs transmission. In contrast to Digital television, DTT programs are broadcasted over essentially television transmitters' network installed on the ground.

In application of GE06[2] agreements, the principle of DTT relies on multiplexing a number of TV digitized program content to transmit and broadcast over a single TV channel and frequency, on a single transmitter. This technique ensures saving frequency spectrum in opposite to analog TV programs broadcasting, which requires each TV program for each TV channel and frequency, to be broadcasted over a transmitter. *Cf. Figure II-1a) and 1b)*: illustration of DTT and analog TV principles.

A TV channel of 8 MHz bandwidth can feet 20 Digital TV programs carried out over one frequency: The techniques applied to achieve this objective, rely on digital signal processing and broadcast technology. The main techniques in use are, compression techniques, modulation types and channel coding: consequently, the delivered pictures' quality and signal robustness depend on the above quoted techniques.

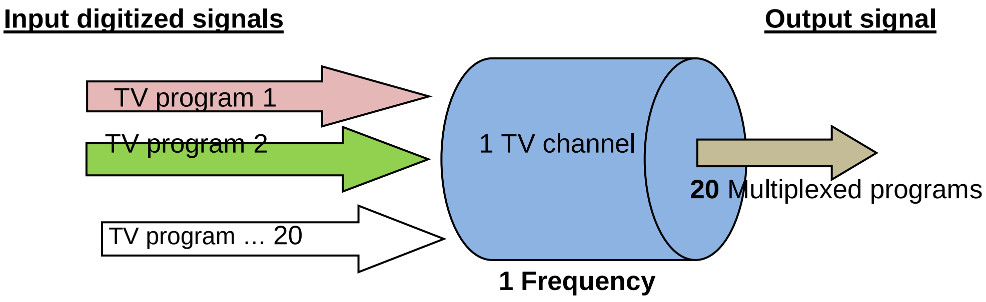


Fig.I-1 a): DTT principle: more or less 20 TV SD programs in a channel, over one frequency

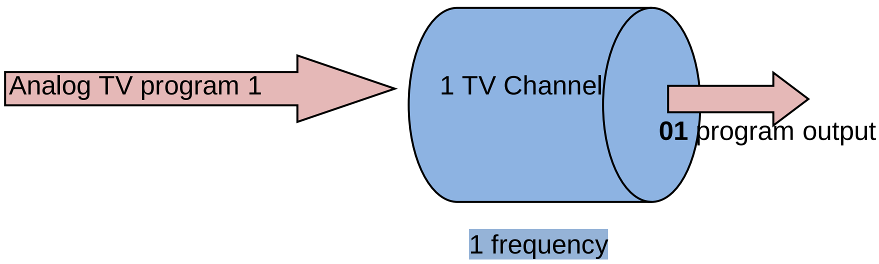


Fig.I-1 b): Analog TV principle: One program in a channel, over one frequency

I.2.1. Digital video signal processing

The first operation to apply is to convert analog signal into digital, this operation is well known as ADC (Analog to Digital Conversion), starting by sampling.

a) Sampling:

To achieve sampling operation, Shannon and Nyquist theorem must be respected: First of all, the idea of the theorem is the faithful reproduction of the signal slices or samples, resulting from analog to digital conversion; for this purpose, the samples must be taken frequently within a period of one second. In that way, the number of samples per second is called sampling frequency or samples rate and noted $\mathbf{f_s}$.
Thereby, Nyquist theorem says: “Suppose the highest frequency component, in hertz, for a given analog signal is f_{max} , the sampling rate must be at least higher than $2f_{max}$, or twice the highest analog frequency component” [3] . Accordingly, the sampling frequency must be chosen as:

$$\boxed{f_s > 2\, f_{max} \Longleftrightarrow T_S < 1 / 2\, f_{max}}$$

Where T_S represents the sampling period.

In case this condition is not respected, for example if the sampling frequency is less than f_{max}, that will cause aliasing due to some of the highest frequencies at the input which cannot be correctly represented at the output of digitized signal. The figure below, illustrates the case where: $\mathbf{f_s < 2\ f_{max}}$.

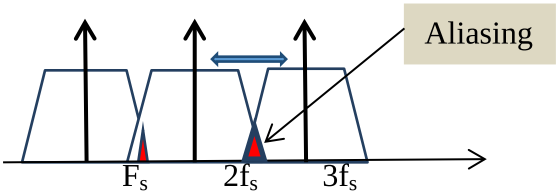


Fig I-2: aliasing signal spectrum

Applying this theorem to video signal that bandwidth is limited by 6 MHz in PAL and SECAM system and 5 MHz in NTSC system, will give two different sampling frequency: In order to harmonize and get only one sampling frequency for both systems, the Radio-Broadcasters' Committee has decided to sample analog video signal in component form, according to lines frequency of both mentioned above systems. Therefore, video sampling frequency must be a multiple of 15 625 and 15 750 Hz, respectively lines' frequencies of 625/50 Hz standard and 525/60 Hz standard. Thus the retained video sampling frequency of **13.5 MHz**.

Electronically, this operation is achieved by an electronic circuit at the input stage of an ADC as presented in figureI-3 below:

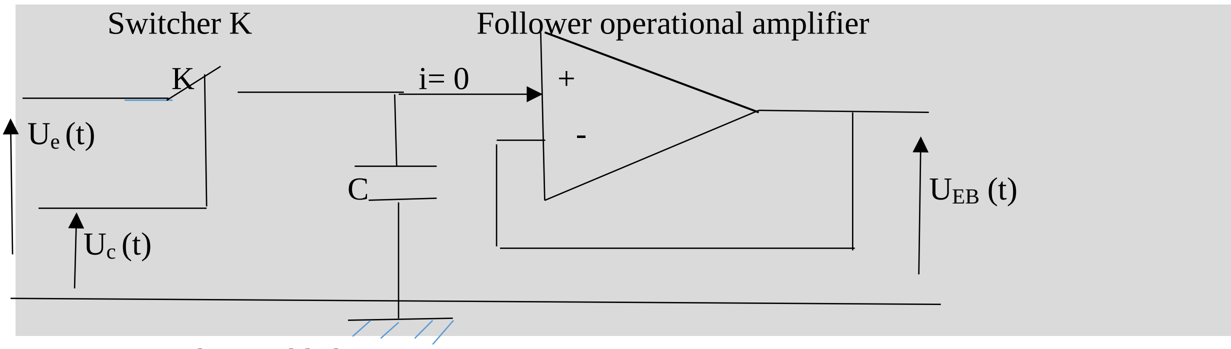
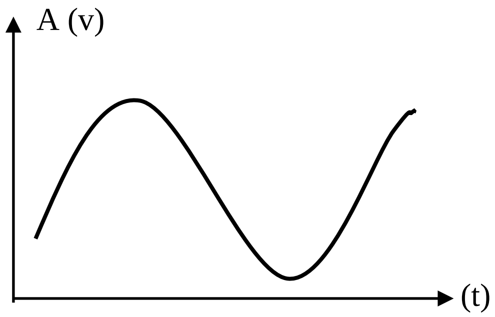


Fig.I-3 Sample & Hold electronic circuit

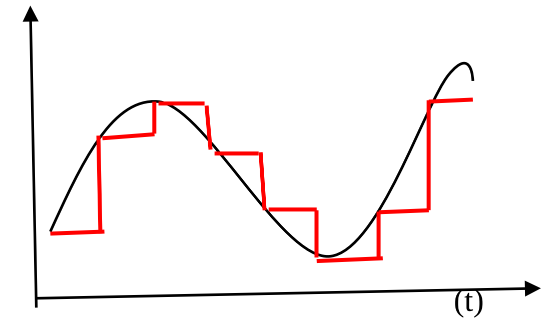
$U_e(t)$ is the voltage of analog signal at the input stage and $U_c(t)$, the order voltage of the switcher (K), "C" is the condenser which loads the input voltage when the switcher K is “on” when it is off , this voltage is held by “C”, and as the current “i” at the input of a follower operational amplifier is set to “0” (i=0), the output voltage U_{EB}(t), remains the same as the input one. This voltage represents the voltage of the sample (k) at the period (kT_e).

- Graphic illustration of a sampled signal



a) Original input analog signal

Fig I-4-a): original signal



b) sampled signal at output (red color)

Fig.I-4-b): sampled signal

The signal at the input is an analog continuous varying signal, at the output there is an analog continuous varying amplitude signal and discontinuous (discrete) time varying signal.

b) Quantizing and digitizing process

After analog signal sampling the following step of Analog to Digital signal conversion will be signal quantizing and digitizing:
Quantizing operation is the action consisting of taking a sample at a regular period of time quoted $\mathbf{T_S}$**,** and then captures the voltage of the taken sample at the instant period of time. The captured decimal digit voltage will be digitized into binary digit that means binary coded.

An example of the described process is given in figure hereafter:

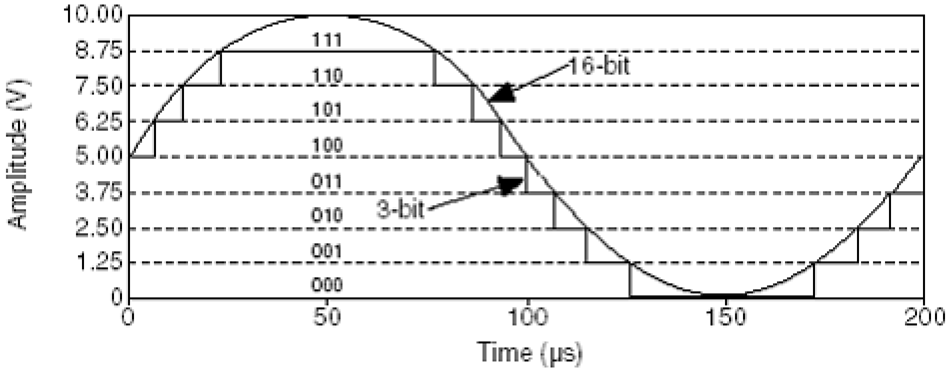
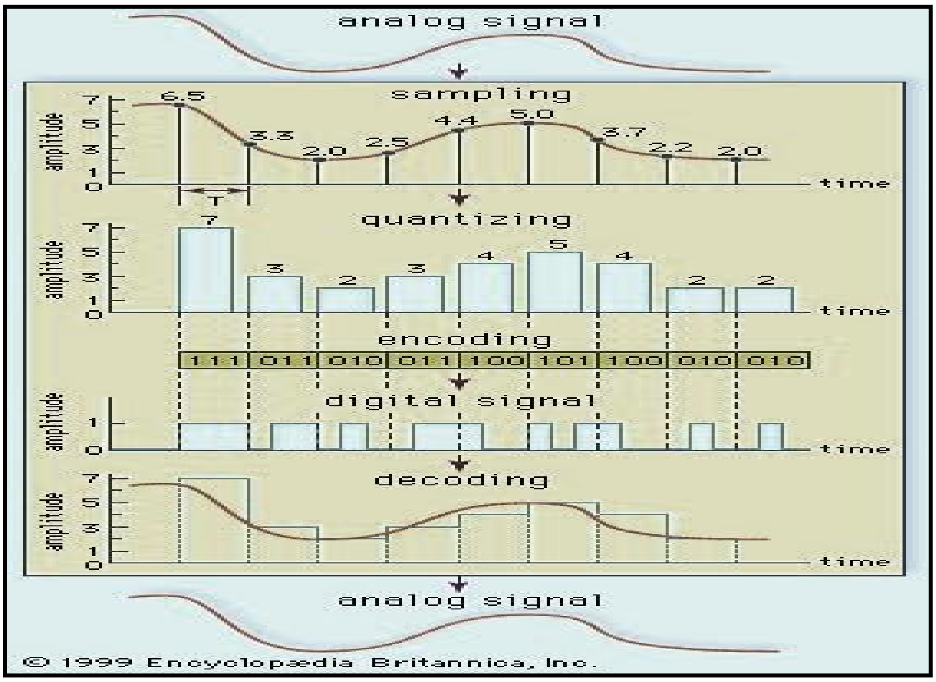


Fig. I-5: quantized and digitized 5 kHz sine wave obtained by a 3 bits ADC

On the amplitude axis, are given the samples voltage in decimal digits, those digits are binary coded by 3 bits: for standard video signal digitization, samples are coded by 8 bits and 14 bits for HD video coding. In practice, decimal numbers are rounded up to higher entire number before conversion into binary: the result of this operation induces some minor errors called quantization error. The summary of Analog to Digital signal conversion is graphically illustrated as shown hereafter: ***binary coding*** must be understood as samples representation by bits, this is also sometime called ***digitizing***.



FigI-6: binary signal encoding

(http://www.britannica.com/EBchecked/topic/585799/telecommunication/76268/Quantization

b)-1. Digital base band signal Processing

At the end of binary coding the obtained signal is made of string of bits (0 and 1), representing digital video data. These data are then modulated first applying NRZ (Non Return to Zero) or NRZI (Non Return to Zero Inverted) or using Manchester code. The aim of using those techniques is to prevent having a long string of 1's or 0's that could not be under control by the receiver; the lack of transitions between the bits, prevents the receiver to generate the right signal clock which should enable him to detect the boundaries of data information and to recover them in the right order, such as they were sent from the original information. Since the boundaries are not detectable, the transmission cannot be reliable.

The illustration of digital base band signal coding / modulation is shown in figures below:

- **Illustration of long string of bits**:

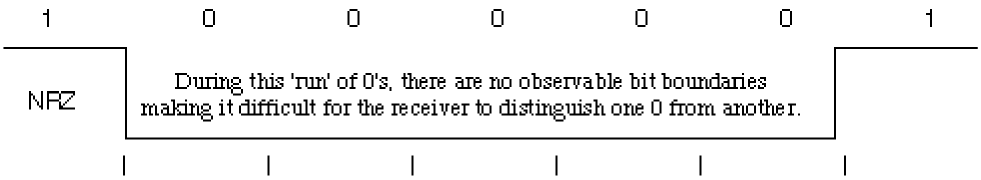


Fig.I-7: Example of long consecutive bits

- **Illustration of NRZ base band modulation encoding**

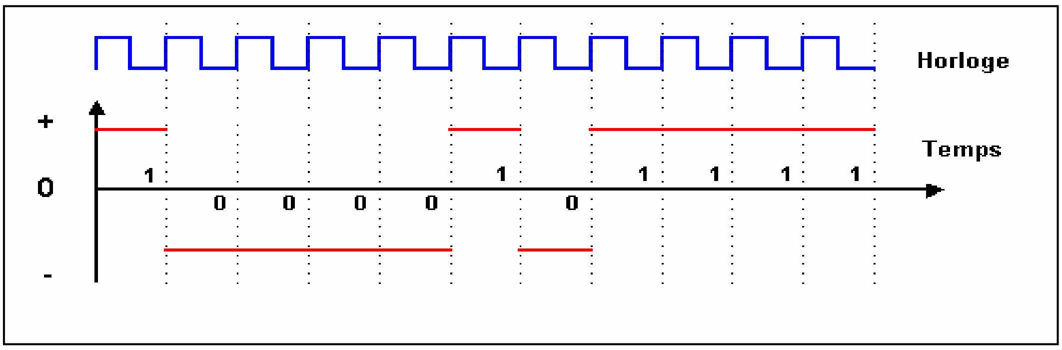


Fig. I-8: NRZ encoded digital signal

The principle of NRZ is: a bit '1' is represented by positive voltage +V, whereas a bit '0' is represented by negative voltage – V.

➢ Illustration of NRZI base band modulation encoding

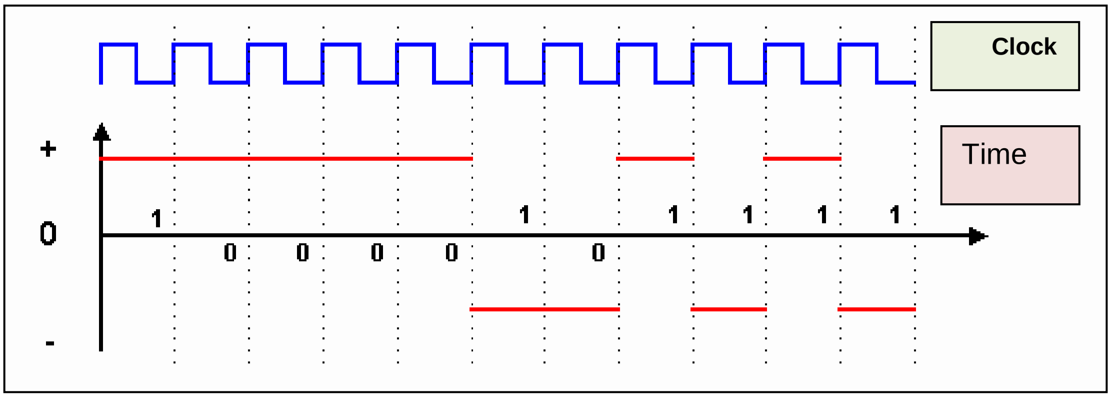


Fig.I-9: NRZI encoded digital signal

The signal front is inverted only when a bit **'1'** is encountered, in case of encountered bit '0', there is no change.

➢ Illustration of base band Manchester encoding modulation

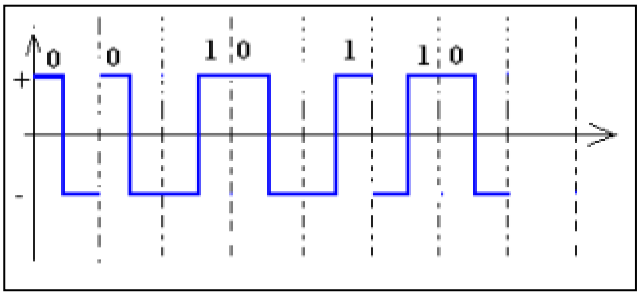


Fig.I-10: Manchester encoding signal type

The principle of Manchester digital signal encoding is to bring out a transition at each transmitted bit: from **high** level to low level when the encountered bit is **0** and from **low** to high level when the encountered bit is **1**. Manchester coding is also called diphase coding because of positive and negative signal polarity.

Remarks:

1°) In general, when using a Manchester coding, there is no DC (Direct Currant) component, and the power density efficiency is around 85% whereas in NRZ coding the power density is 90% with DC component. The presence of synchronization front enable receiver to detect correctly transmitted information and the data bit rate can be estimated as: **D = 2x F** *where F*, is the signal variation frequency in Mhz. That enables transmission to carry out more information.

2°) The process of modulation is applied on encoded Manchester or NRZ digital signal by generating sine and cosine signals, out of phase by 90° which multiply respectively odd and even digits, the sum of both sinewave and cosinewave components, represents the modulated signal according to the type of modulation in use. In DTT, Quadrature Amplitude Modulation (QAM) is widely used; it combines both Amplitude Phase Shift Keying (APSK) and Phase Shift Keying (PSK) modulation. The figure below shows an example QAM scheme using two bits per symbol that is equivalent to QPSK (Quadrature Phase Shift Keying):

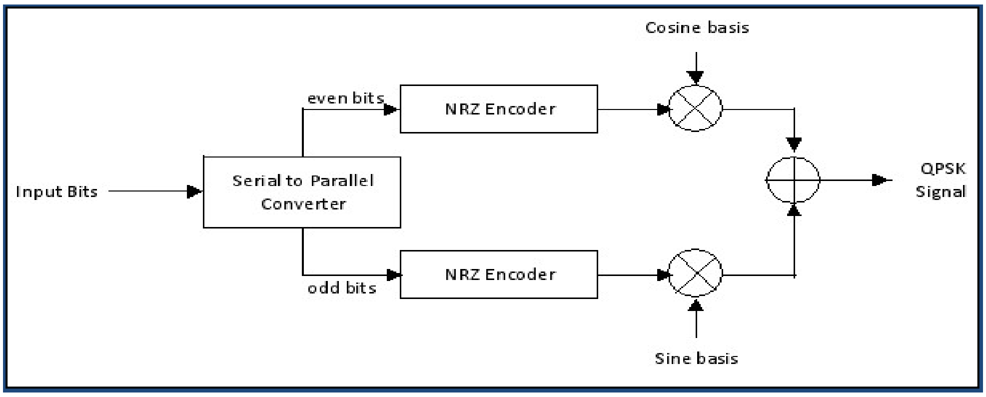


Fig.I-11: NRZ encoding application and modulation process

c) Compression: Source coding

The third step of digital video signal processing is the so called compression, which goal is to reduce the space of digital data storage and increase the data bite rate. The technique leading this process is called source coding and is achieved throughout the following steps:

1- Color component coding
2- Discrete cosine transform
3- Quantization
4- Entropy coding : example of Run Length Encoding and Huffman coding

- **The step1**, concerns the stage of color component compression, base on human eyes properties that requires treating less color component, removing undetectable color details in a picture. This technique is achieved by eliminating repeated color information contained in a picture. Those repeated information are named “redundancy” and can be either spatial (that means amplitude domain) or temporal (that means time domain). From that point of view, compression is categorized by two types:
  Lossless compression and lossy compression, in practice, lossy compression that means compression with lossy, is the one which is performed on digital pictures’ processing.

To achieve lossy compression some conditions must be respected: for example pictures’ quality must be kept after decompression, for this reason compression is achieved with upper and lower boundaries outside which, the resulting compressed picture quality is affected. The technique used to define compression limits or threshold is the **Shannon coding theorem**, named ***entropy coding*** and noted:

$C(x) \geq H(x)$, where $H(x)$ is entropy of source (bitrate), and $C(x)$ is the bitrate after compression.

c)-1. Source coding theorem:

[The average number of bits / symbol of any uniquely decodable source must be greater than or equal to the entropy H of the source][4].

Mathematically says, assuming a source that outputs symbols from S_1 …..,to S_M with occurring probability respectively P_1, …..,to P_M , the entropy of that source denoted H is the average sum (amount) of each probability noted hereafter:

$$H = P_1 \log_2 \frac{1}{P1} + \ldots\ldots + P_M \log_2 \frac{1}{PM} = \sum_{i=1}^{M} P1 \log_2 \frac{1}{Pi}$$

Once the entropy H of a source is known, it indicates in advance the codes which performance is suitable to achieve source coding: Example Huffman coding or Run-Length-Encoding in short “RLE”: This is the last stage of compression that enables the encoder to achieve the compression format.

- The second **step 2**, applies ***Discrete Cosine Transform*** on resulting compressed color component. Discrete Cosine Transform **(DCT)** is based on a finite periodical function. Such cosinewave function can be converted from time domain to frequency domain called Fast Fourier Transform in short: FFT or vice versa from Frequency domain to time domain: (IFFT).

Mathematically the above explanations are expressed:

- Fourier Transform : $x(\omega) = \int_{-\infty}^{\infty} x(t)e^{-j\omega t} \quad dt$ (Frequency domain)
- Inverse Fourier Transform : X(t) = 1/2π $\int_{-\infty}^{\infty} \times\ (\boldsymbol{\omega})\boldsymbol{e^{j\omega t}}\ \mathbf{d}\boldsymbol{\omega}$ (Time domain)

Where x(t) is a time domain signal and x($\boldsymbol{\omega}$) is the complex Fourier frequency signal spectrum; then ($\boldsymbol{\omega}$) is the frequency variable of this signal.

The upper quoted exponential form of Fourier integral is derived from **Euler's formula** who said that exponential function $\boldsymbol{e^{j\omega t}}$ can be expressed as a combination of cosine and sine functions as follow:

- exp *JA* = Cos *A* + *J* Sin *A* **(1)**

Also, based on mathematiques properties, cosine is an even function and sine an odd function it is then possible to replace ''*A*'' by ''– *A*'' *from* expression (1) as follow,

exp (- JA) = cos(- A) +J Sin(- A).

We can rewrite this latest exponential expression owing to cosine and sine properties as below:

Cos (-*A*) = Cos (*A*) and Sin (-*A*) = - Sin (*A*), thus:

- exp – *JA*= Cos *A* – *J* Sin *A* **(2)**
- $e^{-j\omega t}$ = cos(-ωt) + jsin(-ωt) = cos(ωt) – jsin(ωt)

DCT in particular is a Fourier-Related Transform [5] that is named Discrete Fourier Transform **(DFT),** but according to the expressed **Euler's formula** above, cosine function rather than sine function is applied on **DCT**: Clearer said, DCT is applied only on real numbers, i.e. cosine function which is an even function.

Thus, the obvious difference between DCT and DFT is that the latter uses both cosines and sines functions i.e. the complex exponentials function. The reason of that is: DCT implies finite different boundaries in contrast to DFT which operates over a finite domain.

In the other hand, Discrete Sine Transform **(DST)** as an odd function, requires for each data, two boundaries which are either even, either odd; this property affects the transform and offers two choices per boundaries. Those reasons led to uniquely boundaries choice such as cosine function transform.

There a number of DCT that have even boundaries such as DCT type I, II, IV and type VI; these DCT_S perform better data compression than others such as DFT_S and DST_S ; moreover, in practice, DCT type II is preferred for digital pictures compression because of computation convenience.

c)- 2. DCT-Part II technique for data compression

DCT- part II or DCT-II is a one dimension often referred to as DCT [6] which transform equation is given as follow:

$$X_k = \sum_{n=0}^{N-1} X_n \cos\left[\frac{\pi}{N}\left(n+\frac{1}{2}\right)k\right] \quad \textbf{(3)}$$

where k = 0, 1…, N-1 and N- real numbers are transform respectively to $\mathbf{X_0, ..., X_{N-1}}$ this transformation occurs ,multiplying X_0 by 1 / $\sqrt{2}$, this achieve to get a DCT-II orthogonal matrix.

For wide image description, the DCT represents the frequency domain as a combination of cosine functions in the x axis and y axis directions: that means it transform a signal of image from spatial domain into frequency domain.

Assuming a spatial image made of **N by M pixels** at the encoder input, and described by a function **f (x, y)**, where, **x** and **y** represent respectively the pixel energy intensity in rows and columns; f(x, y) will be transformed into frequency domain by the function **F(u,v)** where, **u**, and **v** are the DCT coefficient in row $\mathbf{K_1}$ and column $\mathbf{K_2}$. The equation **(4)** that expressed the DCT "conversion" from spatial domain to frequency domain is defined as follow:

$$F(u, v) = \left(\frac{2}{N}\right)^{1/2} \left(\frac{2}{M}\right)^{1/2} \sum_{x=0}^{N-1} \sum_{x=0}^{N-1} f(x,y) cos \left[\frac{\pi(2x+1)u}{2N}\right] cos \left[\frac{\pi(2y+1)v}{2N}\right] . f(x,y) \quad (4)$$

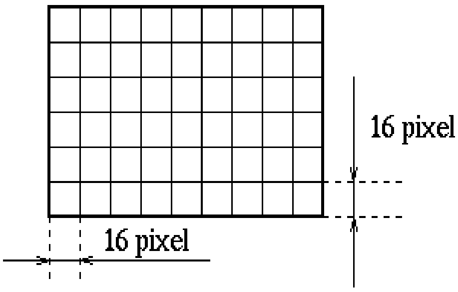
In the above formula (4), f(x, y) is a two-dimensional function and generally for DCT uses, which aims to separate image into parts of N by N sub-bands often called blocks of pixels, is then quoted N x N matrix of pixels: where N- is the width and height of the image,

x,y = 0,1,2,...N-1 and u,v = 0,1,2,...N-1

- The coefficients $(\frac{2}{N})^{1/2}$ and $(\frac{2}{M})^{1/2}$ are normalization constants.

In the case of basic functions compression technique such as JPEG, DCT is not applied in the whole image but it is performed on set of blocks of 8x8 pixels as said upper (NxN) blocs for luminance component (Y) while chrominance component Cr and Cb are divided, depending of chrominance subsampling in 8x8 or 16x8 either by 16x16 blocs, corresponding respectively to a quarter, half and 100% of luminance blocs number.

For MPEG compression, blocks are made of ***16 x16 pixels called Marco-blocks*** of which, luminance component (Y) is made of 4 blocks of 16x16 pixels while each color component C_r and C_b are made of one block of 16x16 pixels, this is illustrated as below:



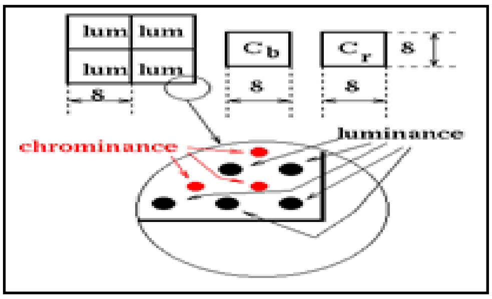


Fig.I-12 a): Marco blocks 16x16 **Fig. I-12 b)**: Luminance and chrominance macro-block

Every *macro block* contains 4 *luminance blocks* and 2 *chrominance blocks*. Every block has a dimension of 8x8 values. The *luminance blocks* contain information of the brightness of every pixel in *macro block*. The *chrominance blocks* contain color information. Owing to that human eyes are less sensitive to colored details, it is not necessary to give color information for every pixel. Thus, four (4) pixels are related to one color information (C_r and C_b).

DCT compression technique lies on that much of the signal energy is concentrated on low frequencies [7] and on high frequencies information, energy is so small so that these parts of information can be neglected without noticeable distortion on the final picture. With this regard, DCT requires to treat less data, treating only Fourier coefficients of low frequencies, neglecting coefficient of high frequencies.

Step 3 of Compression: DCT matrix Quantization

Quantization is the process of reducing the number of bits needed to store an integer value by reducing the precision of the integer[8]

In fact, MPEG format matrix quantization consist of dividing the matrix of DCT coefficients (16x16 pixels) by another matrix called quantization matrix. Indeed, for every DCT value in the matrix, the Quantization matrix defines a quantum value corresponding to the step size for each DCT element position. In that manner, low frequencies coefficients generally closer to the upper left corner, are encoded with a small quantum i.e. a small step size and coefficients of high frequencies, located in the lower right corner are encoded with larger quantum value, i.e. with big size step:
The result of this, tends to reduce the high frequencies, moreover, some of the high coefficients are rounded down to zero after division.
Quantization formula is given as follow:

$$F^*(u,v) = \left\lfloor \frac{F(u,v) + \left\lfloor \frac{Q(u,v)}{2} \right\rfloor}{Q(u,v)} \right\rfloor \cong \left(\frac{F(u,v)}{Q(u,v)} \right) \qquad \textbf{(5)}$$

That means ***F*(u,v)** becomes equivalent to* $\left(\frac{F(u,v)}{Q(u,v)} \right)$ in case where the quotient **Q(u,v) / 2** (component of high frequencies) is negligeable and reduced to zero**.** In the picture below, is shown location of low frequencies on top at left, and high frequencies at lower in right corner.

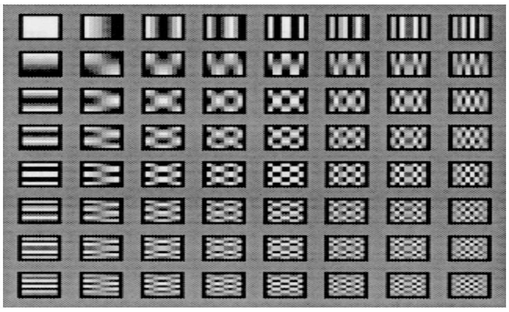


Fig. I-13: DCT image transform

The above image is composed of DCT blocks ready for compression: For compression standards such as MPEG-2, MPEG-4 AVC or H264, DCT quantization matrix coefficients and frequency matrix Quantization illustration examples are given as below :

This is an example of DCT matrix F(u,v):

$$F(u,v) = \begin{bmatrix} -415 & -33 & -58 & 35 & 58 & -51 & -15 & -12 \\ 5 & -34 & 49 & 18 & 27 & 1 & -5 & 3 \\ -46 & 14 & 80 & -35 & -50 & 19 & 7 & -18 \\ -53 & 21 & 34 & -20 & 2 & 34 & 36 & 12 \\ 9 & -2 & 9 & -5 & -32 & -15 & 45 & 37 \\ -8 & 15 & -16 & 7 & -8 & 11 & 4 & 7 \\ 19 & -28 & -2 & -26 & -2 & 7 & -44 & -21 \\ 18 & 25 & -12 & -44 & 35 & 48 & -37 & -3 \end{bmatrix}$$

Fig. I-14 a): example of DCT coefficient matrix

This is an example of quantization matrix Q(u,v):

$$Q(u,v) = \begin{bmatrix} 16 & 11 & 10 & 16 & 24 & 40 & 51 & 61 \\ 12 & 12 & 14 & 19 & 26 & 58 & 60 & 55 \\ 14 & 13 & 16 & 24 & 40 & 57 & 69 & 56 \\ 14 & 17 & 22 & 29 & 51 & 87 & 80 & 62 \\ 18 & 22 & 37 & 56 & 68 & 109 & 103 & 77 \\ 24 & 35 & 55 & 64 & 81 & 104 & 113 & 92 \\ 49 & 64 & 78 & 87 & 103 & 121 & 120 & 101 \\ 72 & 92 & 95 & 98 & 112 & 100 & 103 & 99 \end{bmatrix}$$

Fig.I-14 b) common quantization matrix

This is the resulting quantized frequency matrix $F^*(u,v)$:

$$F^*(u,v) = \begin{bmatrix} -26 & -3 & -6 & 2 & 2 & -1 & 0 & 0 \\ 0 & -3 & 4 & 1 & 1 & 0 & 0 & 0 \\ -3 & 1 & 5 & -1 & -1 & 0 & 0 & 0 \\ -4 & 1 & 2 & -1 & 0 & 0 & 0 & 0 \\ 1 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \end{bmatrix}$$

Fig.I-14 c) Frequency quantized matrix

The principle of frequency Quantization is to divide the DCT coefficients matrix with the defined quanta, which represent the quantization matrix, and round off to integers, example:

For a DCT coefficient of -58 and a quantum element of 10, the frequency quantized element will be obtained as hereafter:

$\frac{-58}{10}$ = **- 5.8** which are rounded off to **– 6** (see first row, third column in upper matrix).

The obtained DCT frequency coefficients quantization matrix is serialized using a zig-zag scanning procedure: This technique is named Huffman encoding.

❖ Step 4, Entropy encoding: Huffman encoding

The technique in use for quantization matrix coefficients serialization is a sort of particular coefficients scanning, describing zig-zag lines: Starting on top at left corner of coefficients array and ending up at the right lower corner.

The illustration of an example of Huffman zig-zag scanning is shown in *figure I-15* below.

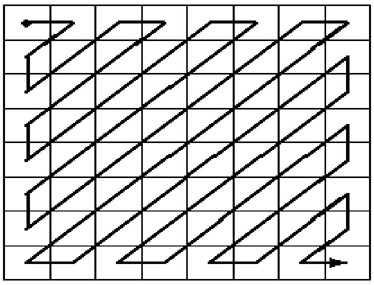


Fig.I-15 zig-zag frequency coefficients, Huffman scanning

The idea of using a zig-zag scanning is to order then group the non-zero entries and apply the appropriated encoding technique, accordingly to entropy encoding theorem. Once the coefficients are zig-zag scanned and serialized, the final action to accomplished compression, is to apply the so called **Run Length Encoding (R L E)**, which a Huffman encoding technique.

c)- 3. RLE-Huffman encoding step

Huffman encoding is a method of statistical digital data compression, it enables reducing the code length of code words: Its principle is to recode the Bytes at a source knowing the entropy of that source. In this manner data with high repeated frequency will be encoded with short length code and data with less or no repetition will be coded with longer code: Thus the encoding name Run Length Encoding (R L E).

For example, observing results in the quoted zig-zag scanning in the above figure, we can notice that many coefficients are equal; that means they are repeated, and also many noticeable zero coefficients noticeable, Huffman compression can be performed; moreover, when there is a string of zero, a common RLE encoding can be applied as follow from ***figI-14 c):***

-26 -3 0 -3 -3 -6 2 4 1 -4 1 1 5 1 2 -1 1 -1 2 0 0 0 0 -1 -1 0 0 0…0 till the end.

Note : the string of zero is code as **EOB** (End Of Bits).

Therefore, if the quantization is too high (big number of quantum), the resulting compression ratio also will be too high, then there will be a very few number of frequency coefficients non-zero to represent faithfully the dedicated DCT block; then at the decompression site, on the corresponding picture, the subdivision in blocks will be visible by human eye: this is called ***picture's artifacts or "blockiness***": That is considered as a disadvantage of RLE. The illustration of that is shown in figures below:

Fig.I-16 a): Blockiness / artifacts picture

Fig.I-16 b) normal compressed picture

Regarding the description of the above section, we can define an overview block diagram of MPEG compression format process: MPEG format compression is globally made of following steps as shown in the figure bellow:

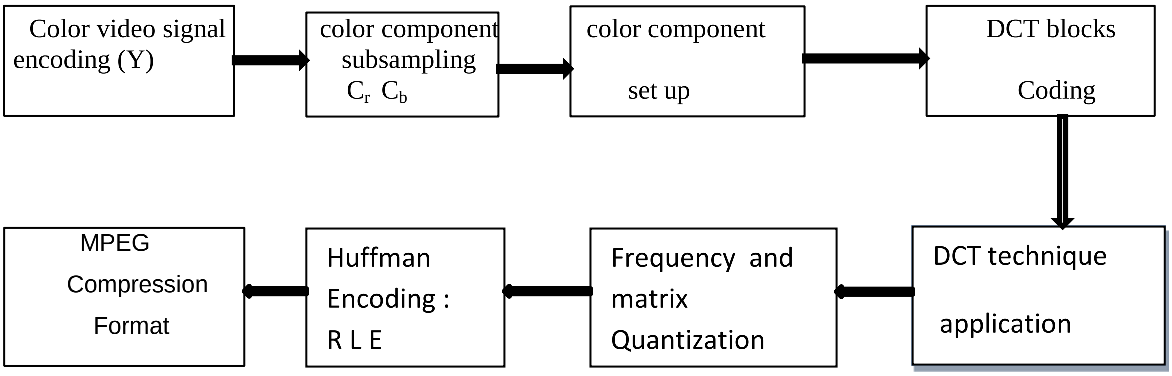


Fig: I-17 MPEG compression Formant Bock Diagram overview

Note: Color video signal is digitized taking in account that human eyes is less sensitive to color details, so it is no need to have the same number of samples as (Y), white and black component , this first stage of samples reduction is sort of data compression. The second stage concern the way to organize samples co-siting (pixels co-siting) that refers to as sampling format (ex 4:2:2, 4:2:0 etc.), then comes DCT Blocks composition (8X8 or 16 X16, 64X64 etc.); The application of DCT on real part of cosine function leads to composition of matrix on frequency coefficients basis, followed by frequency matrix Quantization; This other resulted matrix is scanned and ***serialized by Huffman encoding technique***; the final action on this process is to use the appropriate encoder like RLE to output a digital MPEG compressed video format.

I.2.2. DVB-T2 Compression standard

In the previous section, we discussed source coding importance in digital signal processing, one of the advantages of DVB-T2 is the standard use for data compression, indeed there are many standards for pictures and video compression, the most popular are: JPEG for still pictures, MPEG1, MPEG2, MPEG4, and MPEG7 are mostly use for moving pictures.

For efficient compression, DVB-T2 widely uses the MPEG-4 AVC as ETSI compression standard and H.264 as SMPTE equivalent MPEG4-AVC standard.

The reason why this standard is suitable lies on its capacity to save considerably storage space and increase data rate transmission while keeping excellent picture's quality.

I.2.3. MPEG-4 Part 10/AVC or H.264 Standard

The ITU-T compression standard H.264, also known as MPEG-4 Part 10/AVC which stands for Advanced Video Coding, is the most recent MPEG video data compression format. Owing to its high performance in contrast to MPEG-2, the MPEG-4AVC / H264 is nowadays the most popular video compression format in use for both SDTV and HDTV pictures. An H264 encoder can compress up to 50 % compared to MPEG-4 part 2 format. With this format, high compression can be performed without picture quality degradation. The result of this is: A video file requires less space for storage as well as band width.

H.264 or MPEG4-AVC has more advanced compression methods than the basic **MPEG-4 part 2** compression: [One of the advantages of H.264 is the high compression rate. It is about 1.5 to 2 times more efficient than MPEG-4 encoding**]** [9]**.** In comparison to MPEG4 space saving, MPEG4-AVC compression rate is higher, that enables it to store more information in the same space than MPEG 4.

The biggest advantage of MPEG4-AVC is that it requires low bit-rate on transmission network, enabling it to be used for internet transmission.

For the present DTT network designing, a new compression format will be proposed for implementation; that is ***HEVC*** (High Efficiency Video Encoding), which refers to as ***MPEG-H Part 2*** the equivalent ***of H.265***, it was recently approved by ITU-T in January 2013 and recognized as the successor standard to H.264 or MPEG4-AVC. The high performances of the new compression standard, in terms of bandwidth saving which is approximately [40-45%][10] over the previous MPEG4-AVC compression standard, and other features such as: storage capacity, frame rate (up to 300 fps), enable this standard to support UHDTV 2K, 4K, 8K resolution etc. The newest digital video compression is more efficient and more technically flexible for future needs such as added values content delivery in addition to classic television broadcasting.

I.3. DVB-T2 video data modulation

I.3.1. Quadrature Amplitude Modulation: QAM

The serialized and encoded digital video data after process of quantization, must be "prepared" to suit to the transmission channel, for this reason encoded data need to modulate analog signal named carrier: The main goal of modulation is to squeeze as much as possible, data into the least amount of spectrum: The aim of this, is known as ***spectral efficiency***, it enables to measure how quickly data can be transmitted in an assigned bandwidth.

The unit of measurement is bits per second per 1Hz (**b/s/Hz).** Multiple modulation techniques have emerged to achieve and improve spectral efficiency in accordance to the transmission domain: Quadrature modulation is the common modulation for all of them, the difference from one type to another, relies on the size of data to modulate and to carryout. In this sense, a number of modulation options are available: QPSK (Quadrature Phase Shift Keying); (16, 32, 64, 128, 256) QAM which is M-size Quadrature Amplitude Modulation: This is the widely adopted and suitable for digital wireless terrestrial transmission, especially for digital television and radio broadcasting (DVB-T/ DVB-T2 and DAB transmission).

M-size QAM noted M-QAM (M Quadrature Amplitude Modulation) is a combination of amplitude and phase modulation with n-bits and M-size. The possible combination of amplitude and phase level (in a diagram of x and y axis) is defined by the size or order M of QAM. **M** corresponds to the amplitudes and phases states in the diagram.

To achieve QAM modulation it is first required to know the number **(n)** of bits to code symbols: a symbol is a sample of information, represented by a number of bits: when this sample is bits coded, it is named **codeword**. Knowing the size of the QAM modulation, the number of bits which is necessary to code a symbol, is defined by base 2 logarithm of M-bits size: for example, in case of 64-QAM, a symbol is coded by: lb(64) = ln(64) / ln(2), where ln(64) = 4.158 and ln(2) = 0.693 then the ratio 4.158 / 0.693 give **6 bits per symbol**.

The reverse way gives 2 power n-bits/s to define the size of QAM modulation (2^n where n-represents the number of bits to code a symbol) for the chosen example where n is equal to 6 bits, 2^6 yield to 64 QAM.

Quadrature Amplitude Modulation is a technique widely used for wireless terrestrial communications with high bit rate transmission; it implies two carriers out of phase by 90° and having the same frequency. These carriers are described by functions

I(t) = cos(2πf0t) and Q(t) = sin(2πf0t) , where **I-**represents the In-phase and **Q** the Quadrature amplitude. The resulting modulated signal to transmit S(t), is the sum of I(t) + Q(t) i.e.; ***S(t) = cos(2πf0t) + sin(2πf0t)***.

The technique of QAM enables to transmit a larger number of bits in a few frequency bandwidth (first aim of digital modulation: spectral efficiency) and this amount of bits are transmitted once a time. That means increases of high data rate transmission.

The representation of amplitudes and phases combination in a diagram with (y) and (x) axis, is named ***constellation.***

I.3.2. principle of QAM modulation

The principle of Quadrature Amplitude Modulation is illustrated as follow:

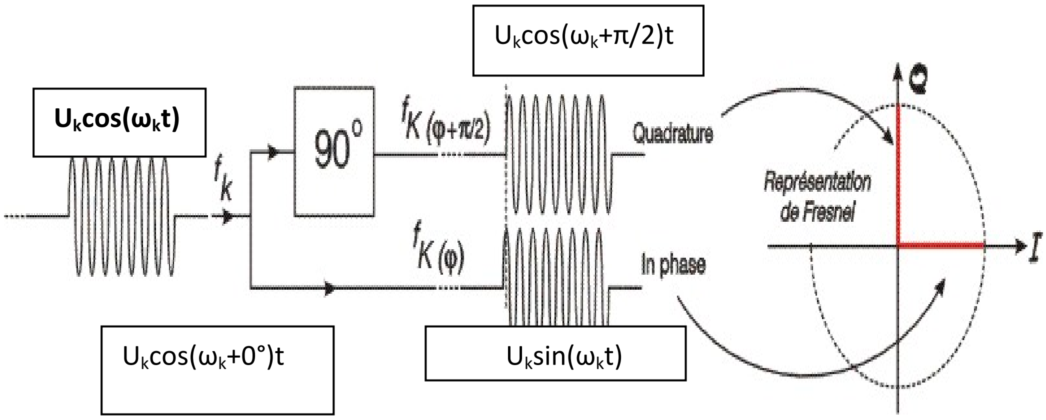


Fig.I-18 QAM I and Q carriers' representation

A cosine wave carrier signal is shifted at 90° yielding two carriers sine and cosine waves to be modulated respectively by I and Q data signals on axis (x,y) and produce a QAM carrier S(t) with an amplitude **ρ** and a phase **φ,** to transmit ; this operation is shown on figure II-17 below:

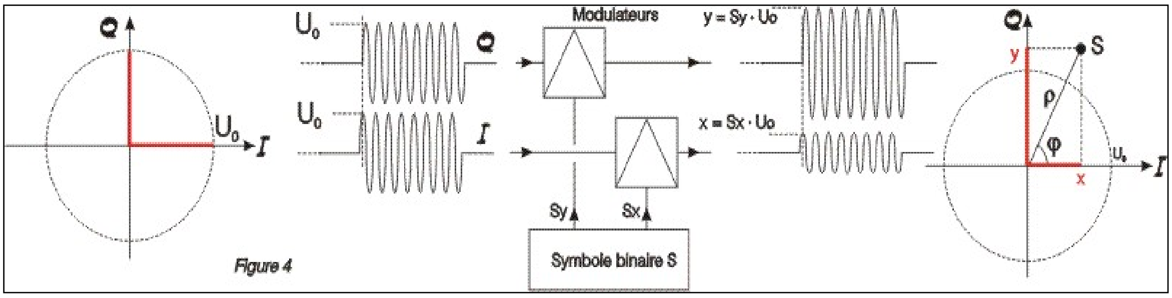


Fig.I-19 QAM modulated signal S(x,y)

The above technique of modulation allows the modulator to reduce symbols' rate. As said in the previous section, a symbol is a number of bits coding a sample of information, the coded symbols are then mapped in an orthogonal diagram on (y) and (x) axis, accordingly to their amplitude and their phase: such of diagram is called constellation.

To provide how QAM operates, an example of 16-QAM constellation is illustrated in the figure I-18, normally 16-QAM is the lower order of QAM, because 2-QAM is

typically the same as BPSK (Binary, Phase Shift Keying) using only two bits: one bit per symbol (0 and 1), in the same way 4-QAM is the same as QPSK (Quadrature Phase Shift Keying) using 4 combinations of phase such as: 00, 10, 01, and 11.
(Two bits per symbol)
8-QAM is almost synonym of 16-QAM in terms of Bit –Error-Rate performance which is 0.5dB better [11].

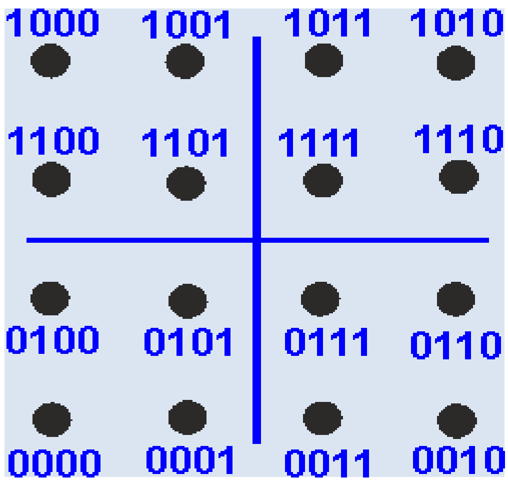


Fig.I-20 **16-QAM** constellation diagram

In the above 16-QAM constellation diagram, a number of bits stream is grouped and associated to the different states of phase and amplitude; 16-QAM mean 16 combinations of amplitude and the number of bits associated to each symbol (Black points) in the diagram is easily defined by the relation 2 power n bits = 16, thus
n = ln(16) / ln(2) that yield 2.772 / 0.693 = **4 bits :**
Neighboring symbols differ only by one bit.

Note: When the number of bits to code a symbol increases, the bit rate also increase and is better, but in contrast when this number is reduced, the transmission is more reliable because of less bit error rate (**BER**)**.**
A tradeoff between BER, QAM order and SNR is shown in the figure below, giving a comparison to one another**.**
[The bit error rate (BER) of QAM can be calculated through Monte Carlo simulations][12].In the figure below is shown a theoretical approach of BER calculation for Gray coded QAM, for even number of bits per symbol.

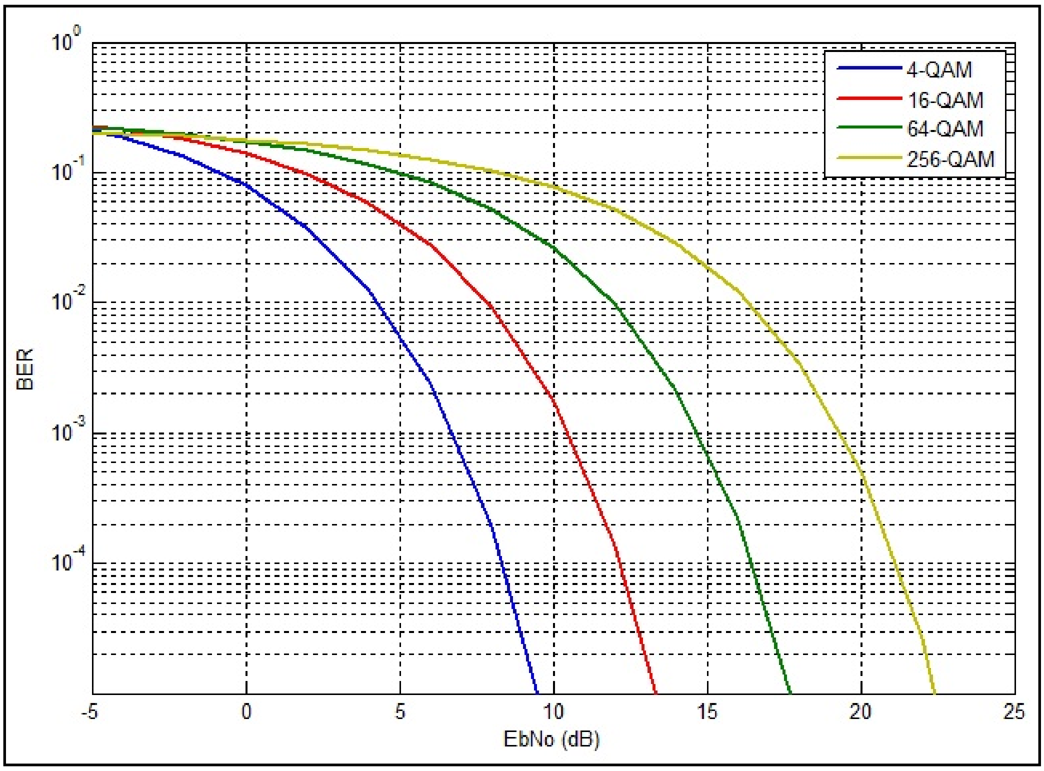


Fig.I-21 BER of 4-QAM, 16-QAM, 64-QAM, 256-QAM in **AWGN**

[Gray coding ensures that a symbol error results in a single bit error][13]. Globally said more the BER is low more the Eb/No (Energy per bit to Noise density ratio) increases and more the QAM size is bigger: Consequently, the transmission is reliable.

The distance from axis origin, to a given symbol represented by a sequence of bits stream in the figure above, determines the amplitude of signal information to transmit and the angle from axis origin to the sample is the phase of that signal.

Therefore, amplitude and phase are simultaneously changed accordingly to transmitted information.

A number of QAM constellations exist, among them; we can quote rectangular, circular and Hierarchical constellation. Rectangular constellation is a sort of Pulse Amplitude modulation that is simpler to transmit and to demodulate, but is not optimal in terms of BER because there is not enough space between constellation points for a given energy. Circular constellations are constructed over line axis and are not often used because of the difficulties to estimate the BER.

Hierarchical QAM is often used in DVB (Digital Video Broadcasting) where the constellation points are grouped into a certain priority making them irregular. This irregularity of constellation points improves the probability of information reception in noisy environment.

The consequence of having higher order QAM that means higher data rate for RF (Radiofrequency) or microwave application in broadcasting or telecommunications field, induce multipath interference. In that case, the spreading of spots in the constellation, reduce the distance between two adjacent phase and amplitude states. When that occurs, it becomes complicated for the receiver to decode the signal properly: In this sense, one can say that noise immunity is reduced for higher order/size QAM.

d) I.3.2.1. Digital modulation parameters assessment

Mostly, noise parameter is measured or determine by Carrier-to-Noise ratio, noted $\frac{C}{N}$ (dB) where **C** is the power of the central carrier and **N**- the noise power , this value enables to analyze if the carrier can still be detected by the demodulator. This method of C/N measurement is shown of figure below (spectrum analyzer display): more the ratio is higher, less is the noise and more the transmission quality is better.

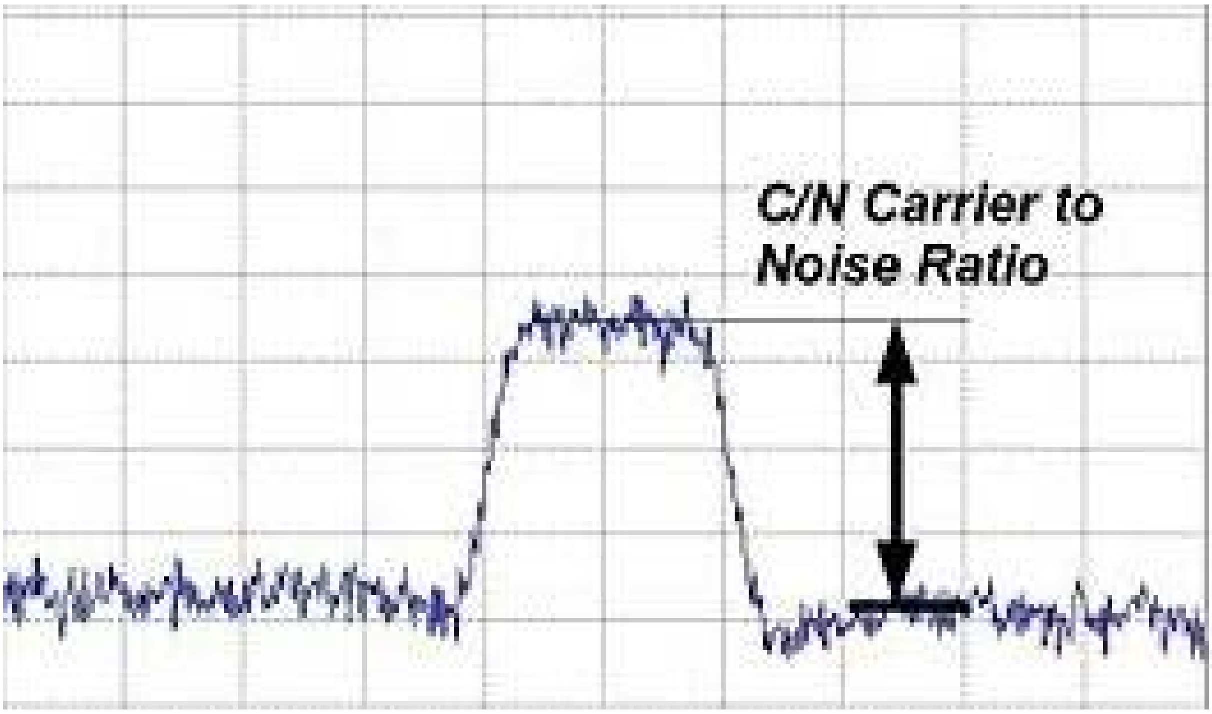


Fig.I-22 C/N measurement on central frequency signal spectrum

In vertical axis is given the amplitude (the power in dB) and in horizontal axis is given the carrier frequency. The C/N ratio is obtained by making the subtraction power carrier minus noise power. It can also so be calculated by the following relationship:

C/N = Eb/No * (R/B), where R = bit rate, B = channel bandwidth

It exist another method of transmission quality or reliability analyses which is based on Energy per bit and white Noise density ratio: this parameter is noted:

Eb/No, where Eb represents the energy per bit before channel correction (FEC); Eb can be assessed as carrier power (**C**), divided by information bit rate (**R**):

Eb = C/ R (C= Eb.R) this parameter can be expressed in Joules unit (J) or in watts per hertz (W/Hz or Ws). The fact of using Eb power instead of the whole carrier power enables to easily compare different modulations schemes in order to make an efficient choice.

No is the noise power density, measured on a bandwidth unit that means **1 Hz**; and not the average power of noise on the total bandwidth of the carrier. In doing so, it enables to compare BER plots performance of different digital modulations schemes as shown in ***figure II-21*** of previous section. More the QAM size is large, more the Eb/No is high while the BER decreases.

For digital transmission the information bit rate parameter is important to determine if a given transmission can be reliable; for this reason Shannon has established a theorem that determines the bit rate threshold as below:

[*The Shannon–Hartley theorem says that the limit of reliable information rate (data rate exclusive of error-correcting codes) of a channel depends on bandwidth and signal-to-noise ratio according to]* [14]**:**

$$\mathbf{R < B.\ log_2\ (1+ \frac{S}{N})} \qquad \mathbf{(1)}$$

R- is the information rate in **bits per second** excluding error-correcting codes (FEC);

B -is the bandwidth of the channel in hertz;

S- is the total signal power (equivalent to the carrier power ***C***); and

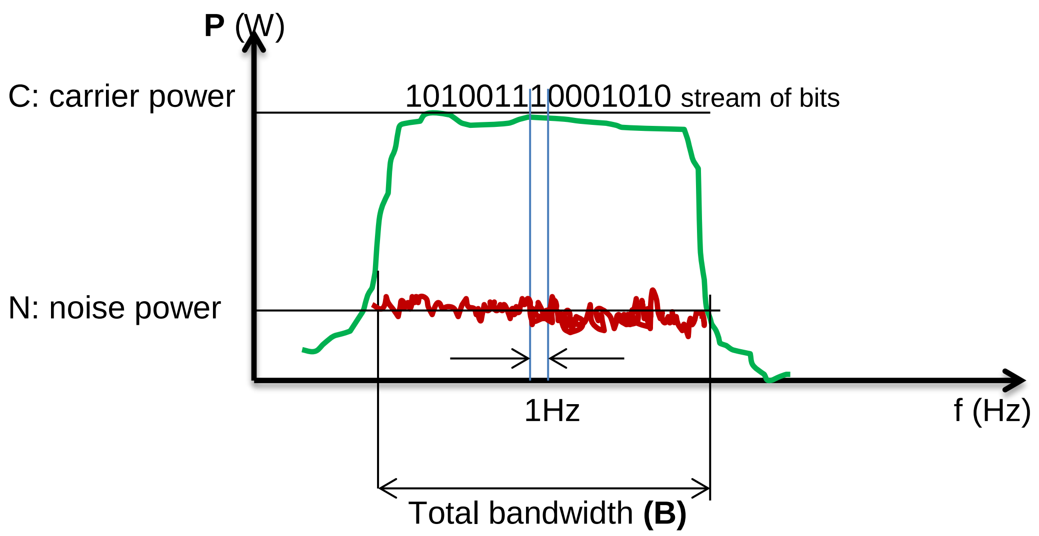
N- is the total noise power in the bandwidth

In the other side, Shannon has determined a transmission channel capacity, in terms of accurateness of a transmission channel to transmit a number of bits in a time unit (1second) without error.

This capacity is given by the expression below known as [***Shannon channel capacity]*** [15] quoted ''**C**''

$$C = B.\ \log_2 (1+S/N) \quad (2)$$ Note: S can be replaced by C: the carrier power.

The technique of **Eb/No** measurement is shown on figure II-22 below:



<u>**Fig. I-23**</u> **Eb/No** graphical spectrum measurement

N.B: A frequency bandwidth is the interval of frequencies comprised between lower and upper limits inside which, the signal attenuation is not greater than 3 dB: that corresponds approximately to 50% of signal power reduction.

There are a number of relationships between the parameter C (Total carrier power) and N noise power on the spectrum bandwidth; these relationships are expressed as follow:

$$\boxed{C / N = C / (No^* B) = (Eb / No) * (R / B)} \quad (3)$$

Knowing that Eb = C / R we can re-express Eb / No as follow:

*Eb / No = (C / N) * (B / R)*

*No= (N * Eb*R) / B * C)*

Where:
R- Information rate in bits per second;
B- Total channel bandwidth in Hertz;
C- Total carrier power
N- Total noise power in the bandwidth

Apart graphical determination of C/N, this parameter can be calculated by the formula given as below, in relation to formula (3):

$$\boxed{C / N\ (dB) = 10 \log (Eb / No) + 10 \log (R / B)} \quad (4)$$

This formula indicates that the parameter of noise C/N depends on noise density in the bandwidth, and also to the QAM size, it is therefore recommended to find out the best tradeoff between the information bits rate, the QAM size and the BER, in order to have a reliable transmission with a high information bits rate.

e) I.3.2.2. QAM modulation quality assessment**:**

Modulation quality is assessed through its Error measurement: the parameter that defines this quality is named Modulation Error Rate (MER) measured in dB.

The relationship between modulating symbol power and error power is given as follow: **MER (dB) = *10log (Average symbol power ÷ Average error power)***; In the case of MER, the higher the number, the better [16].

When using a constellation diagram to plot symbols in phase and in quadrature, the amplitude of each symbol is a vector targeted on the given symbol. This is normally an ideal case, in practice; due to channel impairments the transmitted symbol vector is a little bit different than ideal one. In this case, modulation error is calculated as follow:

Modulation Error vector = Transmitted symbol vector - Target symbol vector

Illustration of this expression is given in figure bellow.

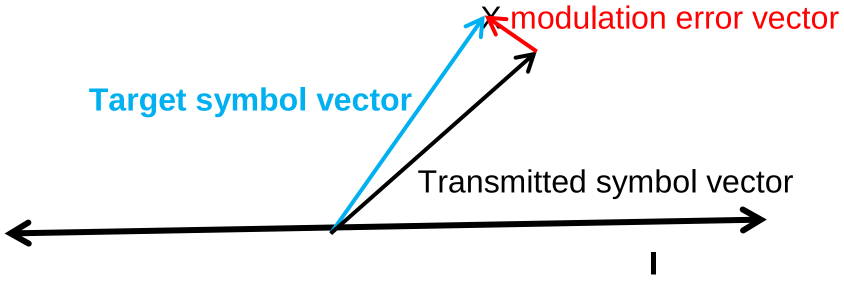


Fig. I-24: Graphic of modulation error vector

Mathematically, the definition of MER as a ratio is given by following formula:

$$\textbf{MER} = 10\textbf{Log}\left[\frac{\sum_{J=1}^{N}(I^2+Q^2)}{\sum_{J=1}^{N}(\delta I^2+\delta Q^2)}\right] \quad (5)$$

[16] Broadcom : digital carrier tonoise ratio and modulation error ratio

Where, I and Q are the real (In-phase and imaginary (quadrature) parts of each sampled ideal target symbol vector, $\delta I\ and\ \delta Q$ are the real (in-phase) and quadrature imaginary parts of each modulation vector.

Equalization is applied on received transmitted symbols in order to improve MER value after this filtering operation the upper threshold of MER is comprised between 40-45 dB

I.3.3. Orthogonal Frequency Division Multiplex as DVB-T2 specific modulation

Wireless terrestrial telecommunications as well as digital broadcasting television are facing to multiple interferences and obstacles surrounding the earth: interferences and obstacles are derived from buildings, road and air traffic, other wireless telecommunications' transmission etc. In order to mitigate interferences, and information fading, DVB-T2 has adopted a technique capable to improve noise and interferences effects: That is achieved on frequency selective channel by Orthogonal Frequency Division Multiplex, named OFDM.

I.3.3.1. Channel with Frequency selective fading

In high bite rate communications, transmissions are physically limited due to the imperfection of the system itself and also due to signal distortion along propagation channel.

One of transmission constraint is that, it requires a temporal separation of transmitted information at the reception. In addition to that, signal frequencies are differently attenuated in the same spreading channel. This occurs when the bandwidth of the transmitted signal is wider than the bandwidth of consistency spread channel which is defined as the minimum of bandwidth in witch; two attenuations in the channel are independent**.** A channel where frequencies independently attenuated is called ***frequency selective channel***.

For frequency selective channel, one of the techniques in use to transmit digital data is a multi-carriers modulation, known as OFDM (Orthogonal Frequency Division Multiplex).

The principle of OFDM technique relies on a block of information data (a number of symbols) to modulate a carrier named Fast Fourier Transform [17] in short: ***FFT.***

I.3.3.2. OFDM Technique principle

OFDM is a multicarrier modulation that was defined as a wireless digital transmission modulation standard under the standard specification [IEEE 802.11a][18] and is very efficient for spread channels with multipath environment, compared to other modulation which are more sensitive to inter-symbol interference.

It first subdivides the entire available spectrum, into many carriers: the number of carriers varies according to the system (2K, 8K, etc.) of transmission. This number roughly can be 2000 or 8000 sub-Carriers.

In OFDM technique, the entire channel is subdivided into sub-channel using sub-carriers modulated by low-bit rate data stream. To avoid interference between adjacent sub-carriers, they are closely spaced and orthogonal each other.

The condition of frequencies orthogonality is required to enable the spectrum of each modulated carrier to be arranged in a manner so that it has a null at the central frequency of each adjacent carrier: that means, there is no crosstalk from other sub channels.

The figure bellow illustrates an example of OFDM four (4) sub-carriers spectrum representation:

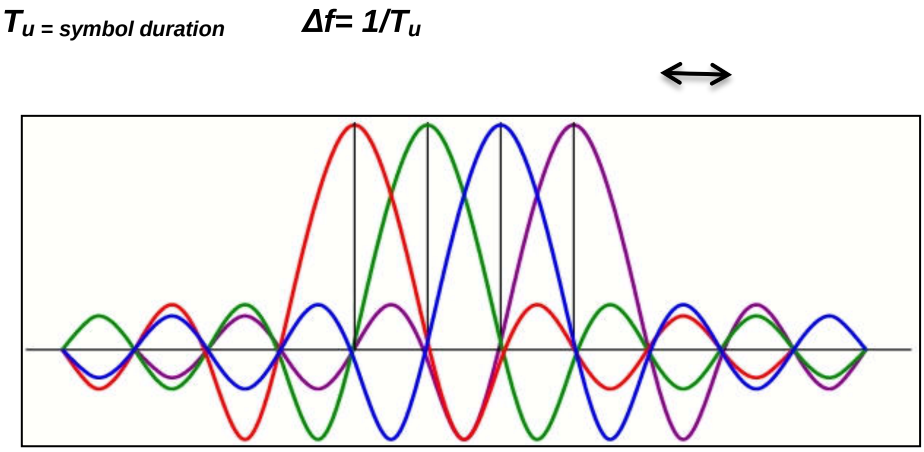


Fig. I-25 Four (4) OFDM Subcarriers spectrum

Observing OFDM signal spectrum it can be noticed that the bandwidth of one OFDM signal i.e. the spectrum of one sub-carrier is very narrow (1 kHz) precisely said 1,116 KHz in digital television broadcasting and the spacing between each sub channel noted ***Δf*** is the reverse of useful period **(T_u)** between adjacent subcarrier . In this manner, ***the symbol rate of each carrier is very low***: consequently such of signal can tolerate a multipath delay. The Inter-symbol interference quoted ***ISI***, may occurs only in case where the delay spread is too long. For example the duration of the delay must exceed 500 ns

The orthogonality condition is achieved in case when the space between two consecutive sub-carrier frequency ***$\Delta f = k / (T_U)$*** hertz, where T_U in seconds is the useful duration of an OFDM symbol: one sub-carrier spectrum duration, this corresponds to the size of a captured window in the receiver; k- is a positive integer which is often equal to **1**.

Therefore, using **n**- subcarriers, the total bandwidth can be estimated as **$B \approx n \cdot \Delta f$ (Hz).**

The illustration of the above description is shown in figure hereafter:

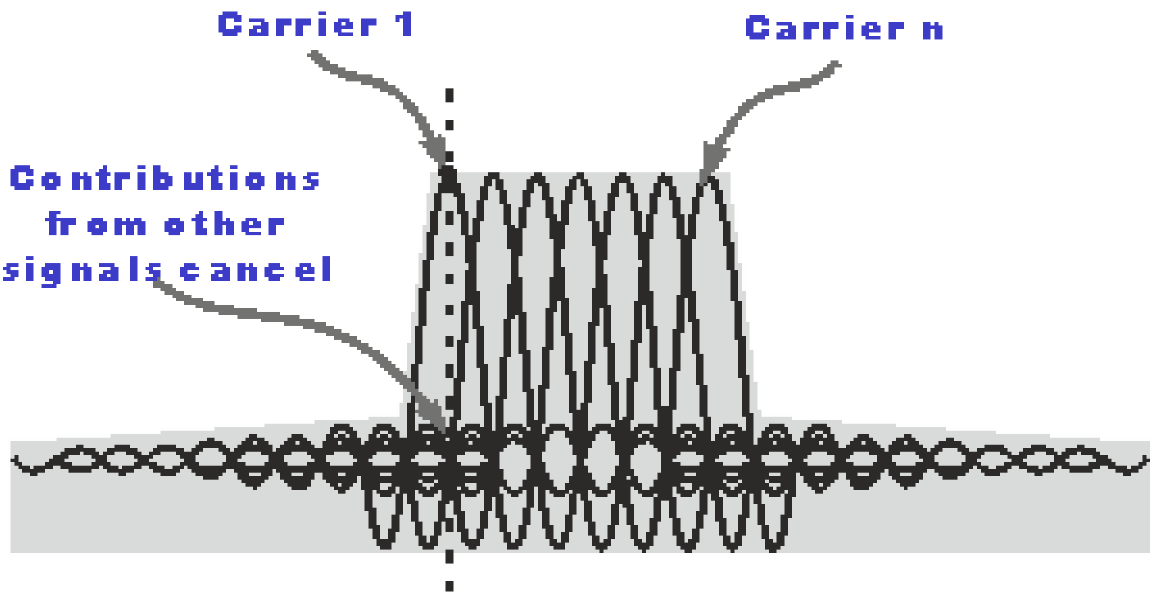


Fig. I- 26: OFDM spectrum with n-subcarriers

Observing the figure above, there is no guard band in between each sub-carrier spectrum in contrast to a simple FDM (Frequency Division Multiplex) where each spectrum is separated by a guard band that protect against leakage interference from adjacent sub-channels.

OFDM does not need guard bands which increase the total bandwidth, whistle OFDM tend to get a narrow–bandwidth. The frequency orthogonality eliminates leakage interference in OFDM.

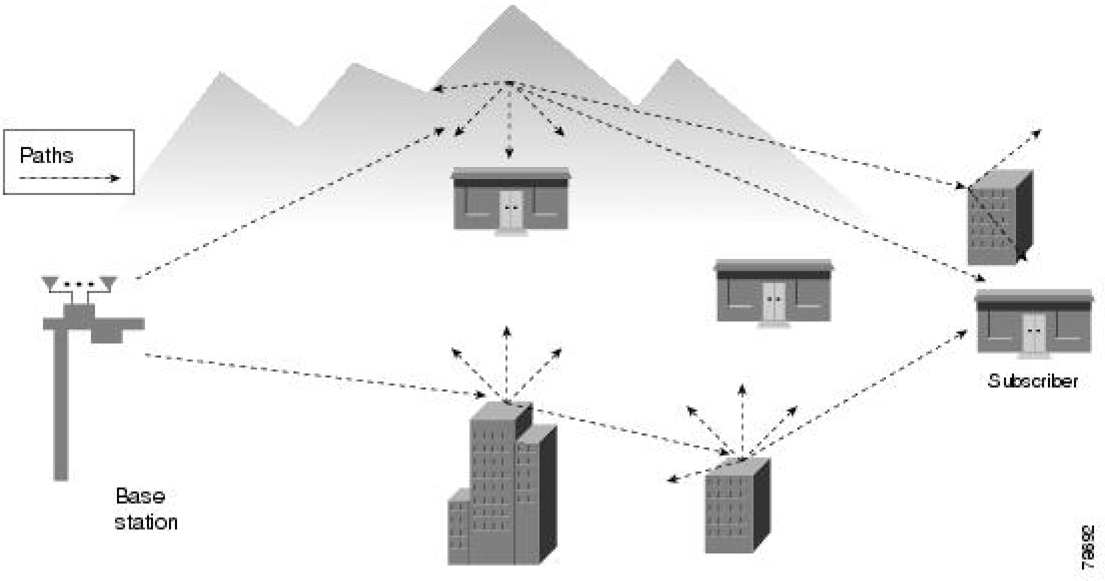
f) I.3.3.3. OFDM effect on Multipath wave propagation and fading

Multipath wave propagation for wireless telecommunications result on that, more than one radio wave are received from different direction on receiver antenna. This situation derives from radio waves reflection and refraction on various obstacles such as: mountains, buildings, bodies, cars, planes, ionosphere, rivers' water etc.

The resulting effect of waves' reflection causes constructive and destructive interference and phase shifting of the transmitted signal.

Consequently in digital radio communication such as DTT, multipath propagation induces errors at the reception and affects the transmission quality due to the brief changes (fluctuation) of amplitude or phase on a short period time, this effect is known as ***fading***.

The illustration of obstacles inducing multipath propagation can be represented as shown in the figure I-26.



<u>Fig.I-27</u> : Multipath waves propagation illustration

Despite the disposal taken to improve multipath signal reception using OFDM technique (transmission of many carriers at a time), it stays to be prone to errors due to channel noise, frequency selective fading and propagation environment effects as shown in the figure above. To overcome errors in OFDM modulated signal, error-coding techniques based on data redundancy are added before transmitting OFDM signal. This latest data coding refers to as channel coding: and the resulting ODFM signal plus error coding data, yield the so called **COFDM** (Coded Orthogonal Frequency Division Multiplex) modulated signal instead of OFDM. Other said ODFM is a non-pre-coded signal, but adding an error correction based on redundancy techniques involves increasing transmitting data number, that lead unlikely to data rate and spectral efficiency reduction. The operation that consists on forward error correction is called channel coding, to be discussed later.

I.4. OFDM Signal processing concept.

I.4.1. OFDM signal make up

As previously defined, OFDM is a digital multi-carrier modulation scheme introduced in digital communication domain such as DTT to extend the concept of single subcarrier modulation by using a multiple sub-carriers within the same single channel.

The aim of doing so is: rather than using a single subcarrier to transmit data stream with a very high data rate, OFDM utilizes a large number of orthogonal subcarriers that are first QAM modulated and transmitted in parallel.

Each of those sub-carriers is individually modulated at low symbol rate, by Quadrature Amplitude Modulation (QAM) technique. The final result of many sub-carriers combination within the same bandwidth enables to reach the same data rate as the one of single sub-carrier modulation scheme. The described process is illustrated in figure bellow:

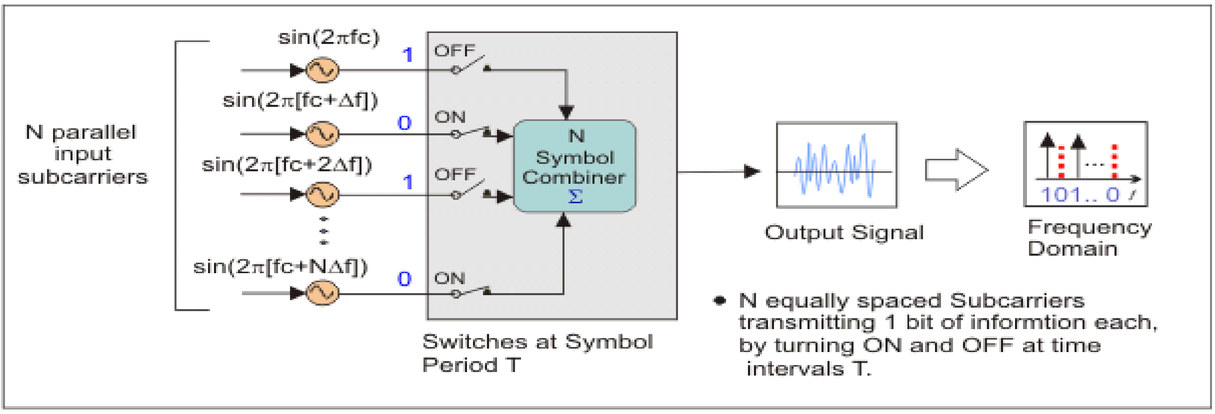


Fig.I-28: **Simple OFDM Generation**

At the input of the multiplexer (symbol combiner), source symbols that was generated to modulate (QAM) each sub-carrier, are complex values that means real and imaginary number, representing the mapped constellation with **N-** associated sinusoids.

Each associated sinusoid having a phase and amplitude represents a set of sub-carriers that are parallel transmitted at the input of a source symbol combiner where there are frequency multiplexed orthogonally. In fact this last operation is not a proper modulation scheme but a stage where the transmitter modulator perform a sinusoid signal from time domain to time frequency by summation of a set of sinusoids. Such of sinusoid signal conversion is achieved throughout mathematiques calculation named ‘Fast Fourier Transform’ in short **FFT**.

The frequency of each sub-carrier is selected in a manner so that they are spaced by $\Delta f = 1/T_u$: a condition of orthogonality.

In the figure above, to make simpler, N-equally spaced sub-carriers transmit a symbol carrying out by ‘1’ bit.

In general, according to the QAM type, a symbol is carried out by at least 4, 6, or 8 bits.

At each period **T** shown in the figure above is output a signal representing the summation of all N-sinusoids i.e.**N**-Sub-carriers summations.

I. 4.2. Inverse Fast Fourier Transform (IFFT) implementation

Instead of using different stages of individual modulators to perform OFDM symbol (different from source symbol) that will lead to high energy consumption and affect cost effectiveness, the Inverse Fast Fourier Transform **(IFFT)** is achieved on the *set of* ***N-samples modulating N-sub-carriers, to make up a single OFDM symbol.***

The principle of IFFT is to take frequency-domain input data (complex numbers representing the modulated subcarriers) and converts them into time-domain output signal that is analog OFDM symbol waveform, carrying out source symbols by many sub-carriers. Other said these source symbols are mapped onto N-group of orthogonal frequencies at a time period **T**, where T represents the OFDM symbol period.

The explanation of IFFT implementation is illustrated throughout the steps in the figures bellow:

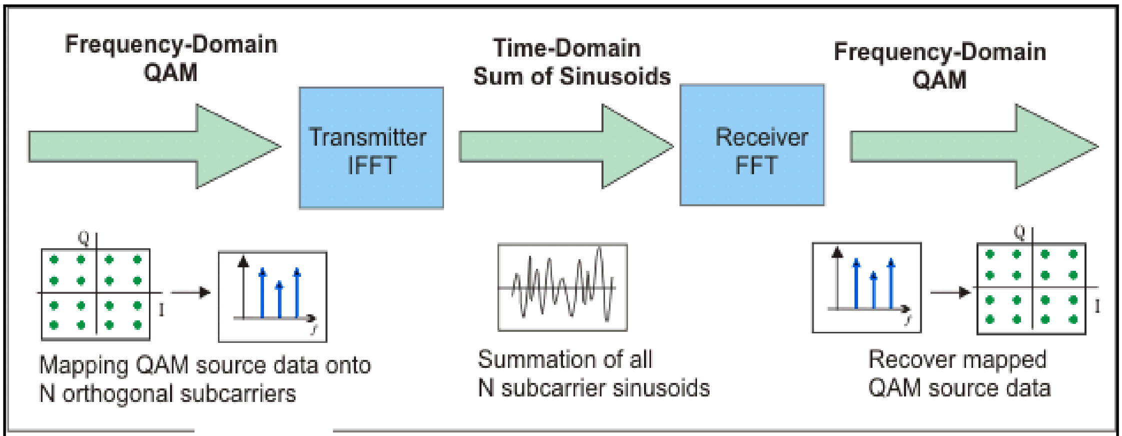


Fig.I-29 Simplified OFDM System Block Diagram

The block diagram bellow gives the steps of OFDM signal process: a serial stream of bits d_o, d_1, $d_{2.....}$dn, obtained after quantization step in compression process, is transmitted in parallel in QAM constellation mapper to output source symbols s_0, s_1, s_2, .. s_n. These source symbols carrying out a number of bits depending to the type of QAM size, modulate individually a sub-carrier $\sin(2\pi f_n t)$ at low symbol rate. The combination of many sub-carriers as shown in the bock diagram, enables data rates (symbol rate) similar to conventional single-carrier modulation schemes within equivalent bandwidths.

The resulting modulated sub-carriers named k_o to k_n providing complex numbers are transmitted at the input of a multiplexer/combiner where they are summed orthogonally in frequency to output an OFDM signal.

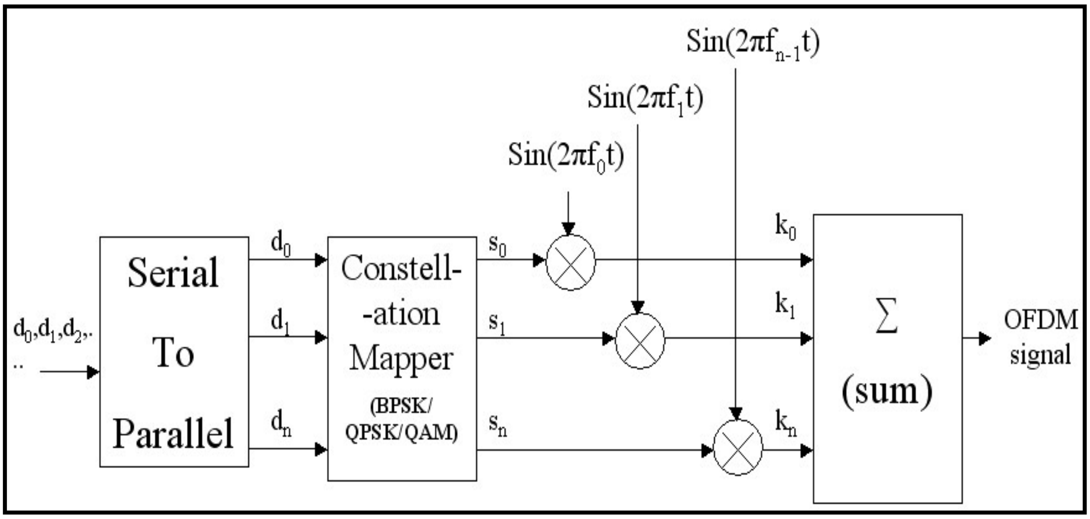


Fig.I-30: OFDM modulator simplified block diagram operation

The upper described process to obtain OFDM signal is technically achieved by mathematiques calculation known as IFFT, as previously said: OFDM Modulator in practice is achieved by, a simple FFT inverse.

Its mathematical definition for an OFDM signal **S(t)** of **N** samples/symbols is described as follow:

❖ **Fast Fourier Transform (FFT) of N samples is expressed by:**

FFT: $\mathbf{S(\boldsymbol{\omega})} = \int_{-\infty}^{+\infty} \boldsymbol{s(t)e^{-i\omega t}}\,\mathbf{dt}$ where $\omega = 2\pi f$

This operation converts signal from time domain to Frequency domain it can be mathematically solved as the sum of complex subcarriers expression which is hereafter:

S(k) = $\boldsymbol{\sum_{n=0}^{N-1} S(n)e^{-2i\pi k\frac{n}{N}}}$ with $\mathbf{0 \leq \boldsymbol{K} \leq \boldsymbol{N}}$ **,** n-comprised within integer 2^N

That condition enables Fast algorithm of sequence N calculation.

n.k represents the number of cycles / periods to collect sequence N of samples

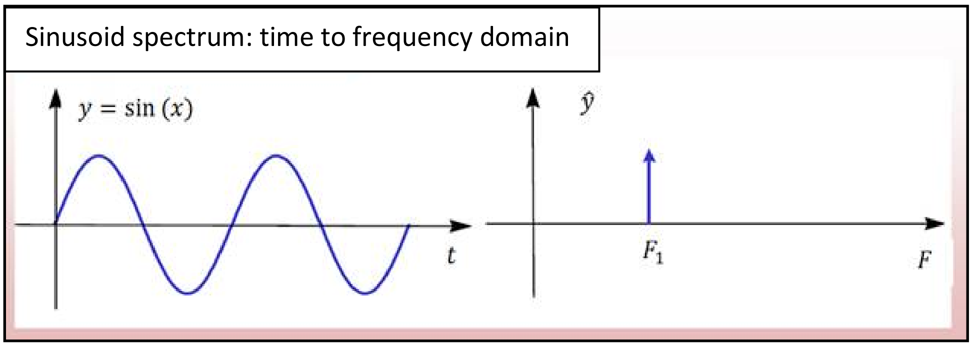


Fig. I-31: Time to frequency domain representation

❖ Inverse Fast Fourier Transform (IFFT)

In **OFDM** signal process, the inverse operation is required; that means, convert a signal from frequency domain (set of complex subcarriers) onto time domain

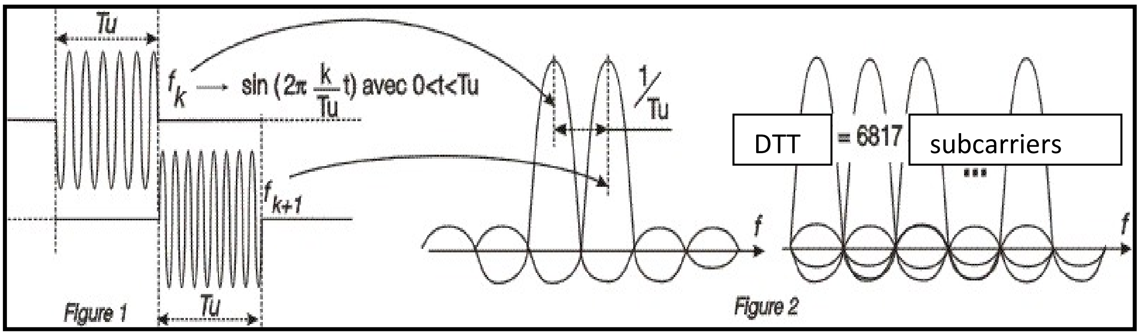
IFFT: $s(t) = (\frac{1}{2\pi}) \int_{-\infty}^{+\infty} S(\omega)e^{-i\omega t}\, d\omega$

The resulting OFDM signal is obtained by the equation below:

$$S(n) = \frac{1}{N} \sum_{K=0}^{N-1} S(k).\, e^{2i\pi.n\frac{K}{N}}$$

The obtained OFDM signal through IFFT process is a discrete spectrum, representing a discrete sampled signal shown in the ***figure II-30***.

<u>**Note**</u>: It is important to underline that Discrete Fourier Transform does not compute continuous varying signal.



<u>**Fig.I-32**</u>: IFFT for OFDM spectrum representation

I. 4.3. Guard Interval in OFDM signal

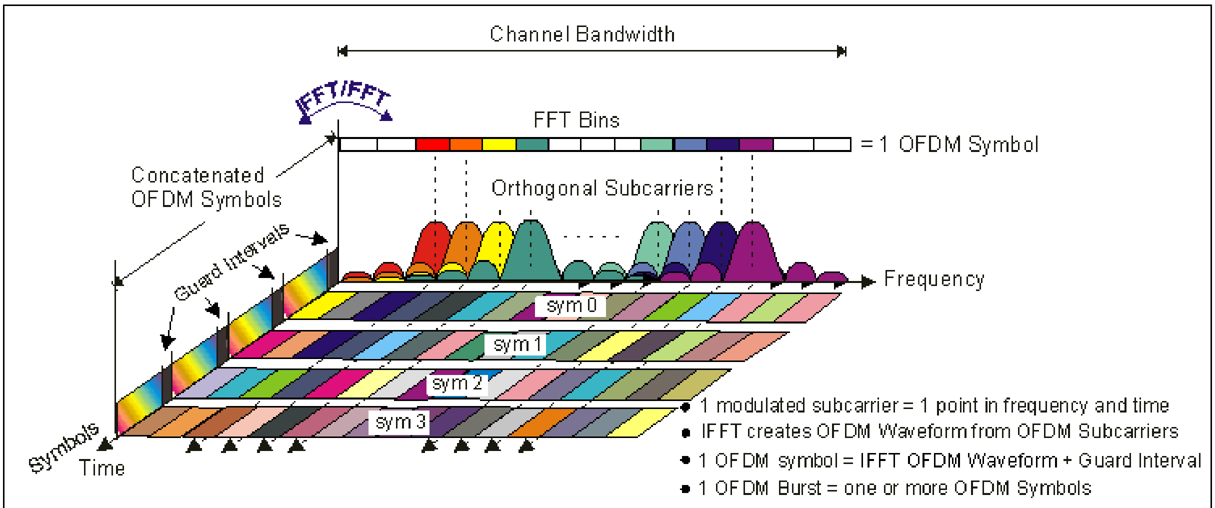
In OFDM modulation, not all the above mentioned subcarriers are modulated by symbols, some of them are used to allow ***guard interval*** frequency insertion, that means the duration which separate two consecutive OFDM symbols.

This guard interval **(GI)** is required in multipath environment to avoid reflected information contained in a delayed symbol, to interfere with the direct received symbols. The rest of subcarriers are utilized for pilot synchronization symbols, the need of these pilots is to provide required information to the channel, for signal demodulation at the receiver side. Other said "Synchronization Pilots" intended to help receivers to lock onto the useful signal.

Likewise, insertion of Guard Interval is to avoid "inter-symbol" interference: The duration between each neighboring subcarrier must be chosen in such way, at the reception the receiver must get enough time to separate each subcarrier carrying data information. Unfortunately, the insertion of guard interval introduces some losses in transport capacity, reducing the spectral efficiency and synchronization markers introduce other losses in transport capacity.

In practice, usually, in 2K mode the useful subcarriers are defined at 1705 and 6817 subcarriers are defined for 8K mode: 2K or 8K or … 32K represents the size of Fourier coefficients after sinusoid signal decomposition into frequency domain.

The structure of OFDM signal (a sum of N source modulating symbols) is given in the figure II-31, after subdivided the whole channel of transmission bandwidth into many sub-channels bandwidth, corresponding to each sub-carrier modulated by a source symbol, separated each other by a Guard Interval.



Frequency-Time Representative of an OFDM signal

FigI-33: Frequency & time OFDM signal representation

Guard intervals are inserted between each of the symbols, to prevent inter-symbol interference which is provided by multi-path delay spread in the radio channel at the reception. Multiple symbols are concatenated to create the final OFDM burst signal as shown in the figure above.

DVB-T2 uses OFDM modulation because of subcarriers orthogonality, which allows more subcarriers per bandwidth resulting in an increase in spectral efficiency. When the conditions of subcarriers orthogonality are perfectly fulfilled, it prevents interference between overlapping carriers.

I.4.4. Inter-symbol-interference (ISI)

The following description will explain how inter-symbol- interference occurs: in case of multipath spread, a delayed symbol signal overlaps with the neighboring symbol signal as shown in the figure bellow:

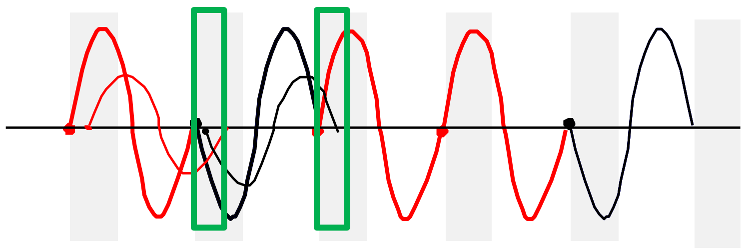
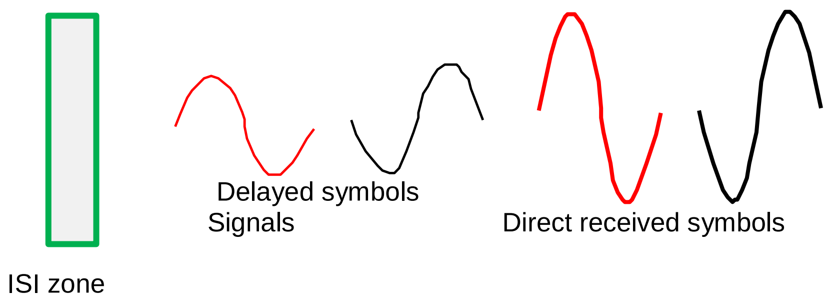
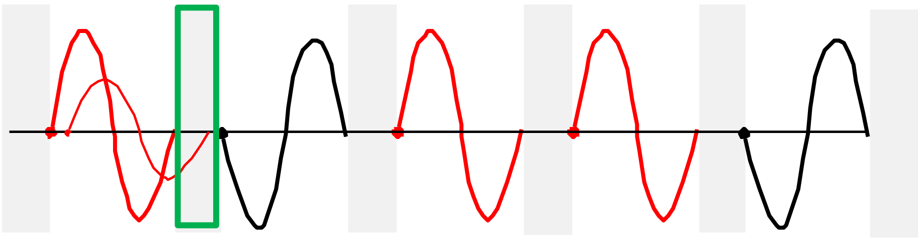


Fig.I-34: ISI illustration



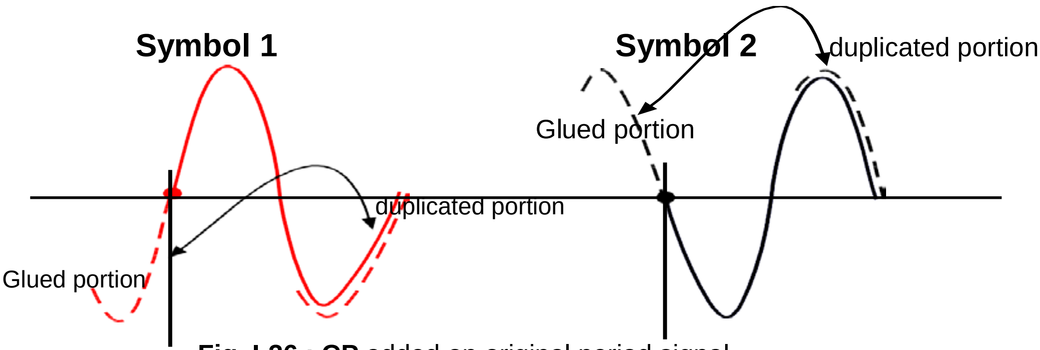
The insertion of Guard Interval (GI) is a solution that enables to avoid ISI as well as Inter-Block-Interference (IBI); this solution is shown in the figure bellow:



<u>Fig. I-35</u>: GI insertion

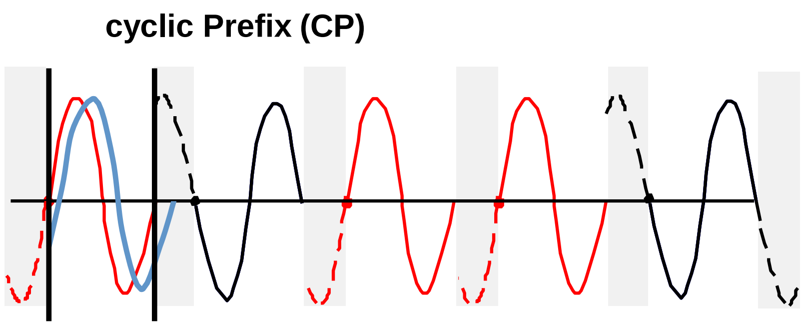
As it is impossible to know in advance how long will be the delay duration, the GI duration is taken higher than the period of direct received symbol. To achieve that, the so called **Cyclic Prefix (CP)** is added to the period of original transmitted symbol: The reason of this is, the hardware does not allow blank space transmission, it needs to send out signals continuously.

The solution of ***Cyclic Prefix*** consists on making the symbol period longer, by duplication of the last information's of this symbol and glues them in the front of the signal.



<u>Fig. I-36</u> : CP added on original period signal

The final resulting signal including cyclic prefix and delayed signal is as follow:



<u>Fig.I-37</u>: CP+ original symbol signal

A shifted phase is observed on the above signal due to delayed signal, this delayed phase is compensated to ensure the periodicity of the final signal by using Fast Fourier Transform calculation that adds a cyclic Δf (orthogonality condition of subcarriers) frequency to the central frequency $\mathbf{f_c}$. In this manner the delayed signal stayed in phase with the original signal because of longer period and is added to this one instead of destroying it.

The benefit of Cyclic Prefix is to allow the signal to be decoded by the receiver even if the packet of information is detected after some delay.

One of the major differences between OFDM and FDM (Frequency Division Multiplexing) is that, any overlap in the spectrums of adjacent signals will result in interference. [In OFDM systems, the subcarriers will interfere with each other only if there is a loss of orthogonality][19]: this occurs in OFDM only in case when there is a frequency error, causing the subcarrier frequencies to shift in a manner so that the spectral nulls are no longer aligned on the central frequency of each spectrum.

I.4.5. OFDM Transmitter structure overview

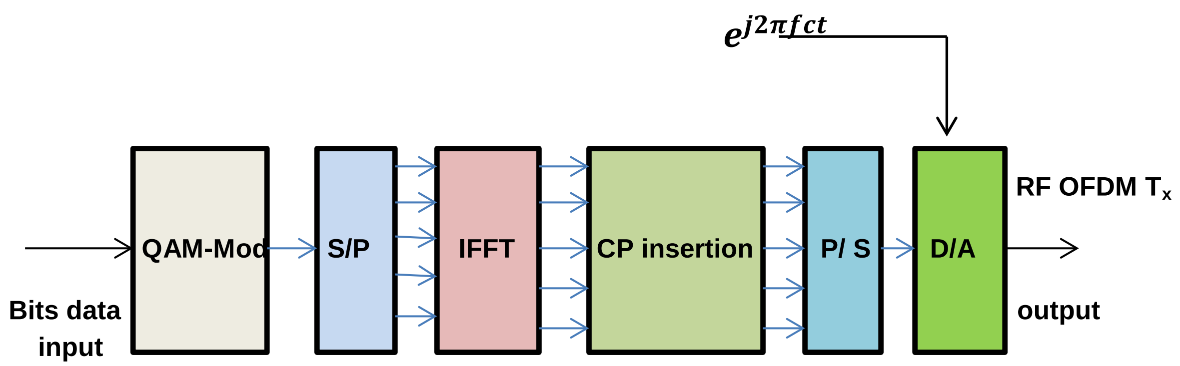


Fig. I-38: OFDM transmitter block diagram

- **S/P:** Serial to parallel selector of N complex symbols
- **IFFT:** Inverse Fourier transform for complex symbols summation into time domain
- **CP**: Cyclic prefix insertion to get longer signal period to avoid ISI
- **P/S:** To get serial packetized N- OFDM signal to transmit
- **D/A** : RF modulated OFDM signal to transmit over a channel

I.5. DVB-T2 Channel coding process and interleaving

In the previous section, it was discussed OFDM efficiency in wireless digital communications including Digital Video Broadcasting system. It was shown how better OFDM technique enables multi-paths spread reception. To improve DVB signal reception in addition to OFDM (to create a COFDM), channel coding is performed to mitigate signal attenuation in noisy spread channel. One of the methods in use is the so called Forward Error Correction (FEC) based on part of information redundancy transmission.

The purpose of channel coding is to give more robustness to the transmission so that transmitted data stay less affected by disturbances present on transmission channel. The technique which can enable transmission robustness is to add extra or redundant data before their transmission: The most popular techniques in use to achieve channel coding are:

- Reed-Solomon (R S) code , known as Forward Error Correction
- LDPC-BCH code
- TURBO code

I.5.1. Reed-Solomon codes

As said previously, the aim of channel coding is to add extra data which can protect transmitted information in the channel, this technique is widely applied in digital television (DTT) radio communication transmission; furthermore, it is used in digital Hi-Fi systems.

For example Reed-Solomon code is used to correct scratches and dust effects on a music CD. The transmission channel in this case becomes the CD itself: it contains extra data to replace when reading, data which are unusable because of dust or scratches.

For Digital Television Terrestrial Transmission (DTT), Reed-Solomon code was applied in **DVB-T** standard to improve transmission robustness, in so far as the transmission channel is subject of multipath reflexion effects.

R S code principle in DTT can be described hereafter:

Reed-Solomon codes are block-based (OFDM signal/sequence of data symbols) error correcting codes, in the receiver side, each block is individually processed and erroneous data are corrected as much as possible, so that the receiver can recover the original data. Anyhow, not all the erroneous data are correctable, the capability of R S code to correct a number of errors, depends on its features.

The type of RS code is a linear code and is specified as RS (n, k, s) where n–represents the length of the total code word given in bytes, k- represents the length of a taken block of data symbols and **s-** is the number of bits in a single symbol. The illustration of this explanation can be given as follow:

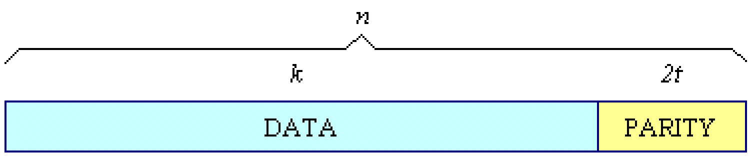


Fig:I-39 RS code length

To find out the parity symbols that must be added, the encoder performs the difference between the n-bytes minus the taken k data symbols: this can be expressed by the relation hereafter: **2t = n - k** and the meaning of this for a receiver is: the decoder can correct up to **t** symbols that contain errors in a code word[20].
The described Reed-Solomon code in DVB, especially in DVB-T, is performed by the following example:
RS (204, 188, t = 8): For a given 204 bytes code word, made of 8-bits per symbol, the parity symbols that must be added are: **2t = 2x8 = 16** symbols, among them, the decoder can correct up **to t symbols**: that means **8** symbols, which contain errors in a **code word of 188** data symbols. More than 8 data symbols, the decoder cannot correct errors contained in the other symbols of the code word, the other rest of symbols are used to control the encoder function. In one word, Reed Solomon code consists on: for a given code word made of k symbols to transmit, RS code will transmit a code word **n = (k +2t**). A code working in this manner is called convolutional code and has efficiency R corresponding to following fractions: ½ , 2/3, ¾, 5/6, 7/8.

I.5.2. Turbo codes

Like Reed-Solomon code, Turbo codes are error-correcting codes; the encoder is made of two convolutional codes: that means outer code and inner coder, between the two codes there is an interleaver, the rate of such code depends on the configuration scheme, for example the figure below shows a parallel mounted codes configuration:

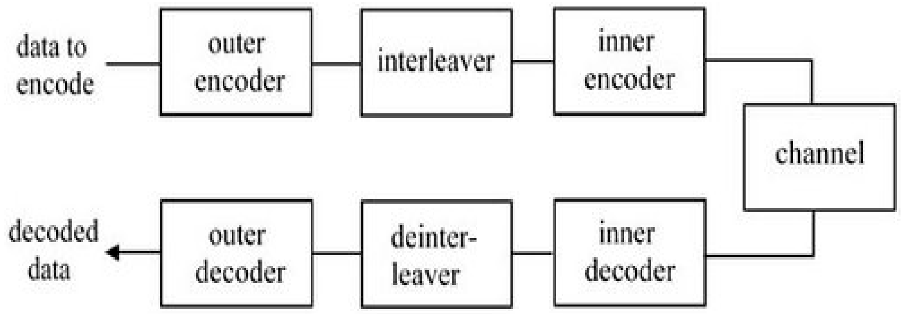


Fig.I-40: Turbo codes parallel configuration

Source : http://www.scholarpedia.org/article/Turbo_code

If the codes are parallel mounted, the resulting rate of parallel concatenation combining two codes with rates *R*1 and *R*2 gives a global rate [21] *Rp* equal to:

$$\boxed{Rp = \frac{R1R2}{1-(1-R1)(1-R2)}}$$

Note: analyzing this formula comparing to the case of serial concatenated codes, where the Resulting codes rates R_s is equal to $R_S = R1 \times R2$; the parallel codes rate is higher than that of serially concatenated code.

Moreover, using parallel concatenated turbo codes, that enable bandwidth efficiency, compared to serial concatenated turbo codes. The interleaver stage enables to scatter information contained in a single symbol to others symbols, to prevent a completely loss of information of that symbol in case it it might be destroyed in the spread channel.

The natural coding rate of a turbo [22] code is *R*=1/3 that means: for one bit at the input of the encoder, turbo code will add two extra data and output 3 bits. Consequently this can be used for low data transmission, but for high data transmission like television broadcasting this significantly reduce data modulation rate that means: symbols rate.

In noisy environment, RS code or Turbo codes are limited, especially in case of digital television transmission field using a multipath reception due to multicarrier modulation, to mitigate this RS code limitation, new channel codes are implemented: Low Density Parity Check code (LDPC) and Bose-Chaudhuri-Hocquenghem(BCH) code.

I.5.3. Low Density Parity Check: LDPC and BCH codes

Low Density Parity Check codes group are considered as more performed for error correction, although all of error correcting codes are based on linear block technology: they are serial concatenated linear codes, where the ***outer code*** is handled by ***LDPC*** code and the ***inner code*** is handled by ***BCH***.
In Digital Video Broadcasting (DVB) system the two codes are combines to improve error correction; for that BCH code is added to eliminate long errors in cascade that are typical for LDPC codes at low error rates [23].
The length of code word made by LDPC and BCH is given as follow:

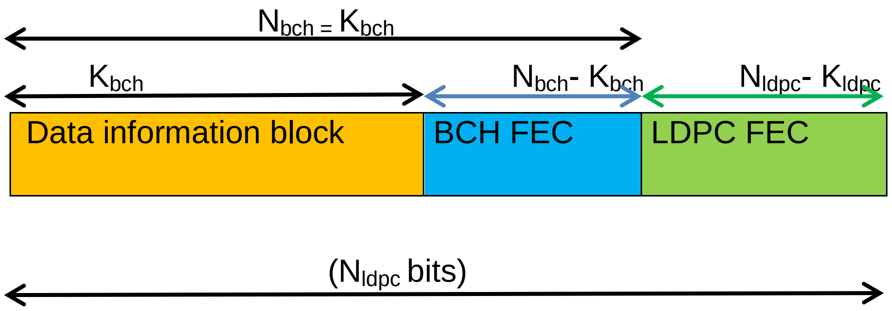


Fig.I-41: LDPC –BCH code word structure

For a given block of information K, the parity symbols to be added in BCH code is N_{bch}- K_{bch} whereas LDPC parity symbols is N_{ldpc}- K_{ldpc} . The total length of N_{ldpc} = 64800 bits at 8K OFDM DVB-T2 mode and 16200 symbols 2K OFDM mode
The combination of the two forward error correction schemes LDPC-BCH gives a very robust signal reception in DVB transmission, for these reason convolutional codes such as Reed-Solomon codes have been replaced by LDPC-BCH codes in Digital Television Transmission, Second generation **(DVB-T2).**

The commonly FEC in use for LDPC-BCH codes are: 1/4, 1/3, 2/5, 1/2, 3/5, 2/3, 3/4, 4/5, 5/6, 8/9 and 9/10 are available, depending on the selected modulation and the system requirements. [*Coding rates 1/4, 1/3, 2/5 have been introduced to operate, in combination with QPSK, under exceptionally poor link conditions, where the signal level is below the noise level*] [24]

The yield of LDPC-BCH code for a given matrix of symbols and chosen parity bits accordingly to a certain calculation is given by the formula:

R= C / (L+C), where C is the number of column and L the number of rows in the matrix.

The principle of LDPC-BCH, involves punching technique that means, for a given number of bits at the encoder input, this number is doubled for the output, but not the doubled number of bits will be transmitted: only a part of them will be transmitted, this technique leads to comprehension of FEC codes usage in the table below:

Table I-1: LDPC-BCH FEC versus transmitted bits

Input number of bits	Number of bits present at the output	Transmitted number of bits	Given FEC
1	2	2	1/2
2	4	3	2/3
3	6	4	3/4
3	6	5	3/5
4	8	5	4/5
5	10	6	5/6
7	14	8	7/8

Example: for a given FEC = 3/5 means 3bits at the input, 2X3 = 6 available at the output, but only 5 bits will be transmitted. Two bits were punched. The effect of punching enables to improve spectrum efficiency for a given C/N ratio is up to **30% - 35%** in comparison to RS code usage with the same C/N.

The relationship between FEC and bandwidth occupation can be seen on the following diagram:

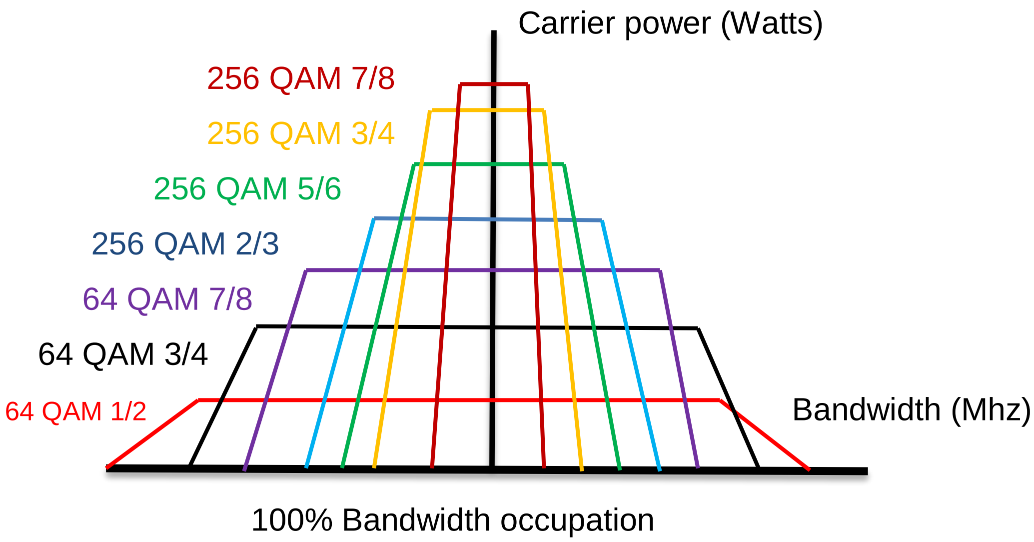


Fig. I-42: FEC to Bandwidth relationship

I-6 Technique of multiple data signal multiplexing in DVB-T2

In general three main principles of data multiplexing are widely used: that are Frequency multiplexing, Time Division multiplexing and statistical multiplexing.

I-6-1 Frequency Division Multiplexing (FDM)

The technique of Frequency Division Multiplexing (FDM), consist on forming a composite signal by frequency translation of different signals. This technique is illustrated as below on figure I-36.

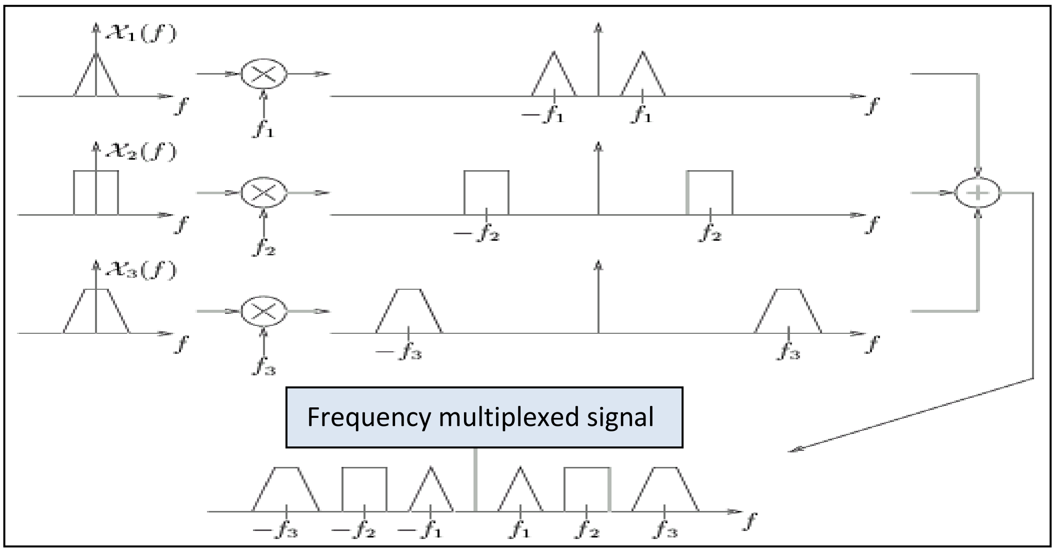


Fig.I-43: Principle of frequency multiplexing.

Assuming a number of signal $x_i(f)$ in base band to transmit once a time: a number of signal mixers tuned on a central frequency f_c in a manner to obtain the spectrum of each signal shifted along the frequency axis to cover the total frequency range without crossing each other, respecting the inter-band introduced to separate each spectrum.

In this way, the FDM multiplexed signal is transmitted to the receiver that will remove one by one every frequency using mixers tuned to the same frequencies as at the transmission side.

This return path of multiplexing is called demultiplexing shown in figure II-35.

The principle of FDM is the one used for analog television transmission where the receiver connected to an antenna, received all the channel frequencies and one must tuned to a given central frequency to get the dedicated program. That means, one program occupies the total bandwidth of the channel, avoiding the possibility to add more programs in the same channel.

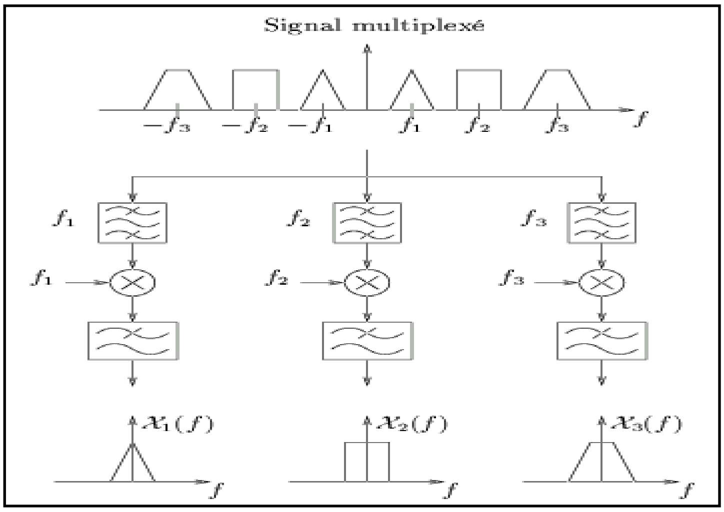


Figure I-44: Principle of frequency demultiplexing.

A series of mixers are tuned to individual central frequency to demultiplex single signal frequency. An overview of that process is shown below:

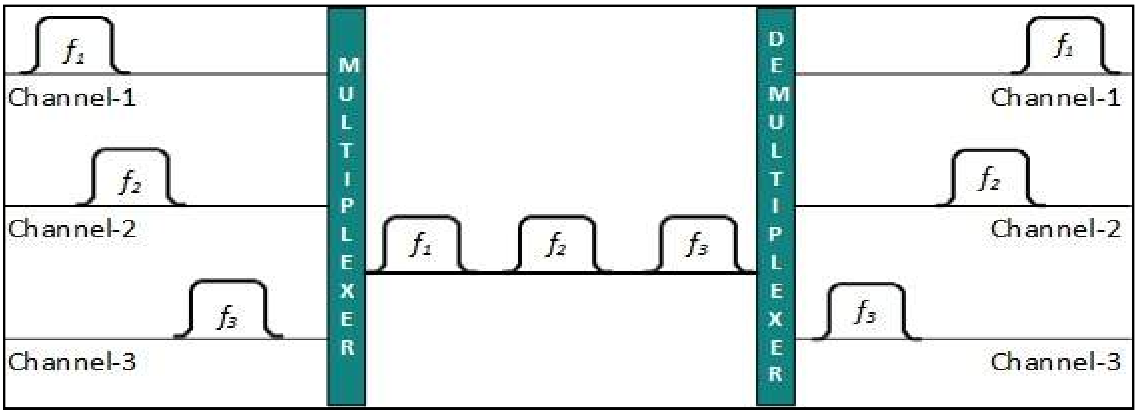


Fig. I-45: FDM multiplexing and demultiplexing process

As previously said, FDM is an analog technology: It divides the spectrum or carrier bandwidth in logical [24] channels and allocates one user to each channel. In analog television transmission each program can use the channel frequency of 8 MHz for example independently and has exclusive access of it. To avoid adjacent channel interference, Channels are separated by guard bands for about 250 KHz, that frequency band must not be used by either channel.

I.6.2. Time Division Multiplexing (TDM)

g) I.6.2.1. Principle.

In contrast to FDM, TDM is applied primarily on digital signals transmission with the aim to save channel spectrum occupation and share it with other users. More than digital signal transmission TDM can also be used for analog signal transmission.
In TDM, the same channel spectrum to be shared is subdivided in time slots corresponding to Elementary Stream (ES) in case of digital signal or frames in case of analog signal. Each ES are transmitted within the provided time slot only, at a given time.

The principle of Time Division Multiplexing (TDM) is a technique that enables mixing data from different channel, with low rate by temporally dividing those data and transmits them on a high rate channel. This type of data multiplexing is called Time division multiplexing because data from different channel at low rate are time interleaved collected. This principle is illustrated in figure I-36.

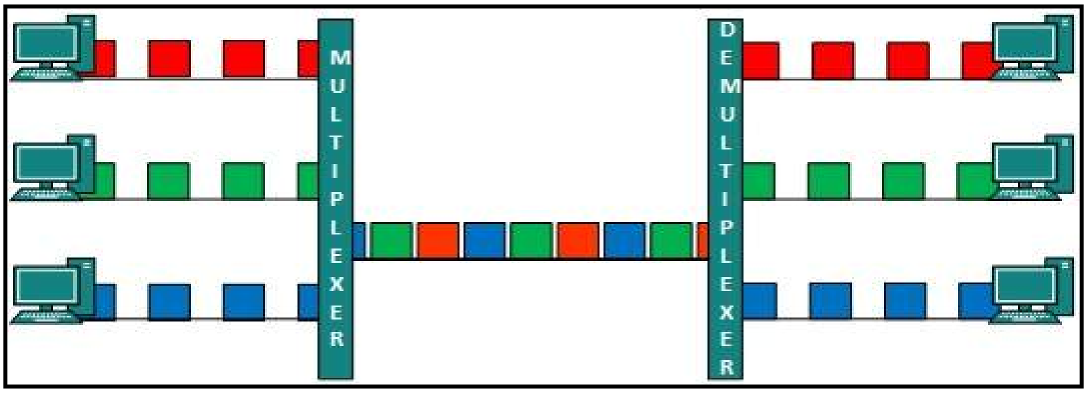


Fig.I-46 Time Division Multiplexing principle.

As it ca be seen in the figure above, TDM works in synchronized mode with the receiver. To enable the demultiplexer in the receiver to recognize on time each original signal, both ends Multiplexer and demultiplexer are timely synchronized and both switch to next channel simultaneously. Interleaved signals from different channels are high rate transmitted in a single channel.

The obtained interleaved signal at the output of the multiplexer is named Transport Stream (TS) or SPTS (Single Program Transport Stream) in case where, only programs are transmitted. But in case where extra data are added to programs data, the signal is named MPTS (Multiple Program Transport Stream)

I.6.3. Statistical Time Division multiplexing (STDM)

I.6.3.1. Principle

In contrast to TDM which continuously transmit time slot even if there are no ES frames, STDM does not transmit empty time slot. In fact sending empty time slot is a waste of space that is not efficient and is considered as a drawback of TDM.

A Statistical Time Division Multiplexer named **STDM or STATDM either STAT MUX** does not assign specific time slots for each device. It uses dynamic time slot allocation with longer lengths to the programs which are more active. But in case if the processing program by the encoder is idle, it will not receive any time slot.

This is achieved by adding an address field to the generated time slots.

Consequently, STDM works dynamically, with variable time slot lengths, enabling him to be much more efficient in bandwidth saving. Moreover it has a buffer memory for temporally data storage in case the processing program signal is much active like a football match for instance.

The recent digital broadcasting video encoders possess in addition to TDM, STDM technology. The usage of STDM in DVB-T2 for example, enables to fit up to 20 SD programs in a single multiplex.

The principle of STDM is illustrated in the figure below:

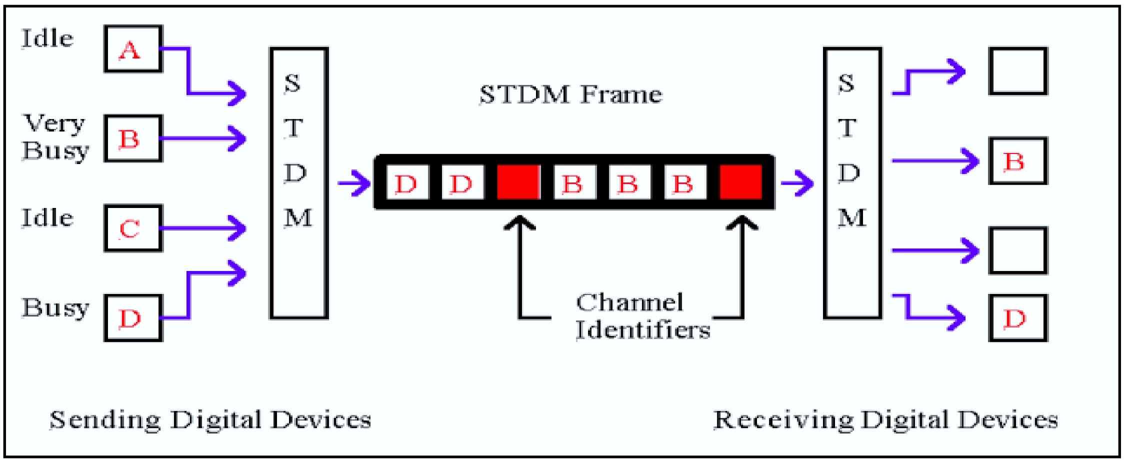


Fig.I-47: STDM multiplexing principle

In the figure above, it is indicated channel identifiers, those identifiers include information about encoder address and also information about the count of the number of data characters belonging to dedicated encoder address.

In this manner only active programs, represented in the figureII-38 by letters B and D are transmitted, idle letters A and C are not transmitted: as we can notice, program B which is more active has longer time slot length than letter D. The total data rate at the output of the multiplexer will be the summation of data rate allocated to program B and D. At the receiver side, the demultiplexer will output only program B and D at the synchronized time with the transmitter.

I.7. General overview of a DVB-T2 modulator

The generated TS signal by the multiplexer, is connected at the input of a DVB-T2 modulator which uses some innovative stages in contrast to the simple DVB-T modulator. The new signal processing stages introduced by DVB-T2 system are to improve transmission protection. The following figure describes the different stages of the T2 modulator.

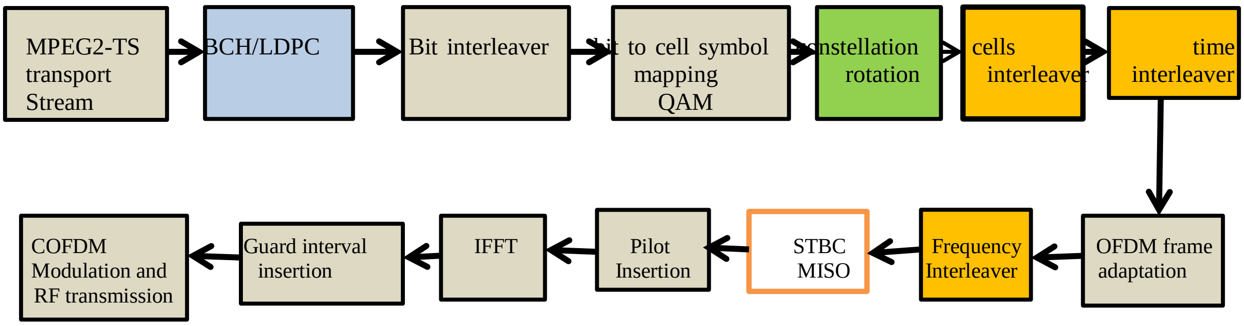


Fig. I-48: Simplified modulator chain of DVB-T2.

Observing the figure above, it is noticeable that the first difference with DVB-T modulator is the introduction of channel coding strategy which is the same for DVB-S2 and lies on the combination of LDPC and Bose-Chaudhuri-Hocquenghem (BCH) codes, previously discussed. It offers better performance than convolution RS code in DVB-T.

At the output of the LDPC-BCH block, bit interleaver intervenes to interleave bits in the same manner as for Physical Pipes Lines (PLP).

After this step, bits are then mapped on their corresponding symbols (cells): The action of mapping bits on their symbols corresponds to Quadrature Amplitude Modulation (QAM) step. More than fixed QAM constellation, DVB-T2 can uses in option Rotated constellation and Q Delay (RQD) in case of very noisy environment. This new technique is provided as an option: It offers additional robustness and diversity for terrestrial broadcasting which enables challenging with noise reception.

According to Figure I-46 DVB-T2 modulator introduced some important innovative features such as the use of three cascaded forms of interleaving: Bit, Time and Frequency. All those interleaving stages aimed to mitigate errors burst in the transmission channel. The frequency interleaver is a pseudo-random block interleaver it operates on OFDM symbols and is used to scramble different PLPs data if any.

The previous interleaver, time and bit operate only on single PLP data.

Another aim of frequency interleaver is to contribute in breaking up any artefact burst that can occur at the channel output.

At the output of frequency interleaver, pilots are inserted into the OFDM symbol: The goal doing that is to enable synchronization and tracking as well as channel estimation at the receiver.

Then the resulting time interleaver signal is output to OFDM frame builder (OFDM symbol composition) sector where parallel to serial operation is achieved. The modulation itself is achieved by using IFFT (signal conversion from frequency domain to time domain).

I.7.1. Some practical parameters for DVB-T2 system designing

I.7.1.1. FFT size and carriers duration

More than DVB-T, DVB-T2 OFDM symbols are extended to 16K and 32K whereas DVB-T OFDM symbols are limited to 1K, 2K, 4 and 8K mode. The understanding parameters of different modes are given in table below for a channel bandwidth of 8 MHz in DVB-T2.

Tableau I-2: FFT size parameters for DVB-T2 at 8 MHz bandwidth:[EN 302 755][25]

Parameters		1K mode	2Kmode	4Kmode	8Kmode	16K mode	32K mode
Total number of carriers K_{total}	Normal carrier mode	**853**	**1705**	**3409**	**6817**	**13633**	**27265**
	Extended carrier mode	*n/a*	*n/a*	*n/a*	6913	13921	27841
Value of carrier number *Kmax*	Normal carrier mode	852	1704	3408	6816	13632	27264
	Extended carrier mode	*n/a*	*n/a*	*n/a*	6912	13920	27840
Duration T_U Elementary period T µs and elementary period T = 0.1094 µs		1024T	2048T	4096T	8192T	16384T	32768T
Duration T_U µs		112	224	448	896	1792	3584
Carrier spacing 1/TU (Hz)		*8929*	*4464*	*2232*	*1116*	*558*	*279*

The results in table 2 above, the parameters given by '*'EBU document 3348 TECH"*. (European's Broadcasting Union) show that more the FFT size is increasing, higher is the number of subcarriers to transmit information. At the same time, the carrier's spectrum is narrow that means many different carriers to carry out much information, enabling more fading resilient.

Table I- 3: Length of guard interval for DVB-T2 in 8 MHz bandwidth channel

FFT size		**GI-Fraction**						
		1/128	**1/32**	**1/16**	**19/256**	**1/8**	**19/128**	**1/4**
FFT	T_U (µs)	GI (µs)						
32k	3.584	28	112	224	266	448	532	n/a
16k	1.792	14	56	112	133	224	266	448
8k	0.896	7	28	56	66.5	112	133	224
4k	0.448	n/a	14	28	n/a	56	n/a	112
2k	0.224	n/a	7	14	n/a	28	n/a	56
1k	0.112	n/a	n/a	7	n/a	14	n/a	28

The results of Guard Interval are given in the table above in relation to FFT size in 8Mhz bandwidth channel: a combination of GI-fraction and FFT size must be taken in account to find out the optimized subcarriers period T_U and GI duration to carry out much information while avoiding ICI: Increasing the FFT size results in a narrower sub-carrier spacing and consequently in a longer symbol duration.

An example of various data rate given in the following table II-3 is to be analyzed when designing a DVB-T2 network.

I.8. Modulation type, code rate and bit rate tradeoff

Table I-4: Maximum bit-rate and recommended configurations for 8 MHz, 32k, 1/128, PP7 [TS 102 831][26]

Modulation	Code rate	Absolute maximum bit-rate			Recommended configuration		
		Bitrate (Mbit/s)	Frame length L_F	FEC blocks per frame	Bitrate (Mbit/s)	Frame length L_F	FEC blocks per frame
QPSK	1/2	7.49255	62	52	7.4442731	60	50
	3/5	9.003747			8.9457325		
	2/3	10.01867			9.9541201		
	3/4	11.27054			11.197922		
	4/5	12.02614			11.948651		
	5/6	12.53733			12.456553		
16-QAM	1/2	15.03743	60	101	15.037432	60	101
	3/5	18.07038			18.07038		
	2/3	20.10732			20.107323		
	3/4	22.6198			22.619802		
	4/5	24.13628			24.136276		
	5/6	25.16224			25.162236		
64-QAM	1/2	22.51994	46	116	22.481705	60	151
	3/5	27.06206			27.016112		
	2/3	30.11257			30.061443		
	3/4	33.87524			33.817724		
	4/5	36.1463			36.084927		
	5/6	37.68277			37.618789		
256-QAM	1/2	30.08728	68	229	30.074863	60	202
	3/5	36.15568			36.140759		
	2/3	40.23124			40.214645		
	3/4	45.25828			45.239604		
	4/5	48.29248			48.272552		
	5/6	50.34524			50.324472		

The main difference between DVB-T and DVB-T2 is the extension of FFT size to 16k and 32k FFT. Those sizes do not exist in DVB-T. According to EBU document, it is not quiet recommended to use a very long symbol time that may result in a poor Doppler performance. It is also recommended to use 32K only for fixe rooftop reception and not for mobile reception at UHF bands. In the other hand it makes sense to implement mobile services using DVB-T2 system because of four time better Doppler performance in this band.

DVB-T2 has introduced additional GI fraction using 1/128, 19/256 and 19/128, which enables further possibilities to use Single Frequency Network (SFN) in DVB-T2 system.

The figureII-40 is an example of code rate, constellation size and bit rate recommended configuration shown in figure below, using an 8 MHz bandwidth channel 32k mode and GI Fraction of 1/128, with Pattern Pilot 7 (PP7).

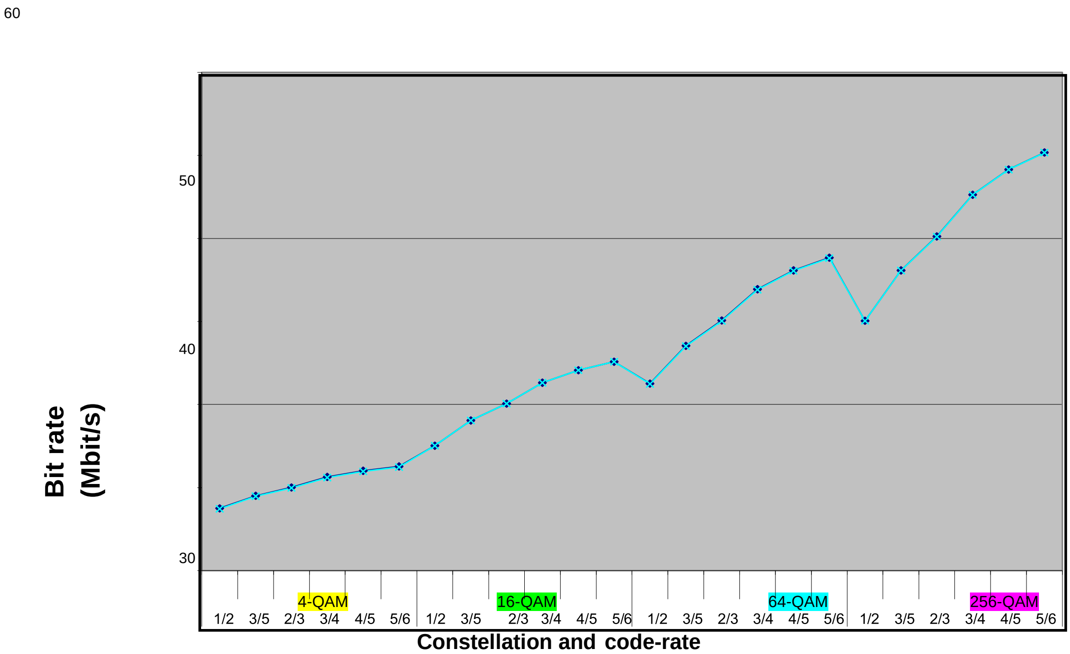


Fig. I- 47: Recommended DVB-T2 parameters configuration

Recommended configuration line

I.8.1. Extended Carrier and data capacity gain

One of the worth of DVB-T2 is that it allows the extension of a number of used carriers in 8k, 16k and 32k mode without modifying the original RF channel bandwidth limits. This technique is called Extended Carrier Mode and can be seen in **Figure I-48** which shows the spectral density of the extended carrier mode for the various FFT modes.

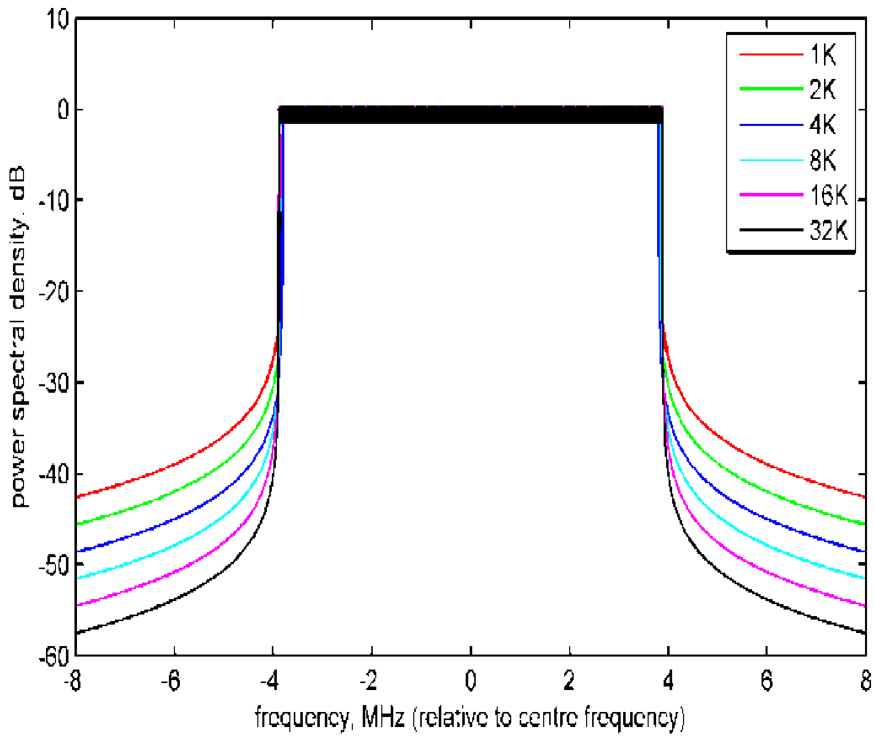


Fig.I-50: Theoretical DVB-T2 signal spectrum for guard interval fraction 1/8 (for 8 MHz channels and with extended carrier mode for 8k, 16k and 32k)[27]

The capability to extend the carrier in comparison to the normal carrier enables to benefit on data transmission capacity; this benefit is estimated in terms of channel gain that is summarized in table below:

Table I-5: Data capacity for **E**xtended **C**arrier **M**ode

	Carrier Mode		
FFT	**Normal**	**Extended**	
Size	**Carriers**	**Carriers**	**Gain**
1k	853	-	0.00%
2k	1705	-	0.00%
4k	3409	-	0.00%
8k	6817	6913	1.41%
16k	13633	13921	2.11%
32k	27265	27841	2.11%

I.8.2. Scattered pilot pattern to be used for each allowed combination of FFT size and guard interval in SISO mode

Table I-6: scattered pilot pattern [28] choice

FFT size	Guard interval						
	1/128	**1/32**	**1/16**	**19/256**	**1/8**	**19/128**	**1/4**
32k	PP7	PP4 PP6	PP2 PP8 PP4	PP2 PP8 PP4	PP2 PP8	PP2 PP8	n/a
16k	PP7	PP7 PP4 PP6	PP2 PP8 PP4 PP5	PP2 PP8 PP4 PP5	PP2 PP3 PP8	PP2 PP3 PP8	PP1 PP8
8k	PP7	PP7 PP4	PP8 PP4 PP5	PP8 PP4 PP5	PP2 PP3 PP8	PP2 PP3 PP8	PP1 PP8
4k, 2k	n/a	PP7 PP4	PP4 PP5	n/a	PP2 PP3	n/a	PP1
1k	n/a	n/a	PP4 PP5	n/a	PP2 PP3	n/a	PP1

According to ITU the combination of Guard Interval Fraction (GIF) in relation with FFT size allows to indicate the suitable pattern pilot to be inserted in OFDM frame for receiver synchronization.

I.8.3. Conclusion

DVB-T2 Standard is considered as the recent standard with a set of new parameters that make him more robust and more efficient for wireless communication and video broadcasting system. The main features that ensure this robustness and efficiency can be summarized as follow:

1) DVB-T2 system uses OFDM modulation with a larger size of QAM modulation and FFF size including Extended Carrier Mode; then it inserts system a classic Cyclic Prefix (CP), making him a very resilient system to transmission over terrestrial frequency selective fading channel.

2) The use of BCH and LDPC codes combination and three cascade interweavers in DVB-T2 system, provide an improved physical layer system (PLP) performance.

It can also employs optionally the concept of rotated constellations in severe in frequency-selective channels.

3) DVB-T2 standard uses the technique of PLP, a very interesting technique which provides the possibility to carry out different services which differently treated in terms modulation scheme, symbol rate, content nature, coding type and robustness. However it maintains constant the output symbol rate of the modulator.

4) DVB-T2 system allows signalling which enables synchronized transmission and reception, while allowing optional transmission by two transmitters for one antenna reception know as MISO (Multiple Input Single Output).

CHAPTER II: Designing DVB-T2 Network with interactive platform

This chapter will discuss DVB-T2 designing issues in Burkina Faso, pointing out the encountered challenges to overcome in order to implement successfully the present project. The target here is first the territory coverage with digital television signal, the second target to attain is to design a reliable network with the best choice of equipment technology and the best of DVB-T2 parameters.

The designed DVB-T2 network must be upgradable and compliant with the recommendations of ITU (International Telecommunications Union) and finally it must be compatible with interactive HbbTV platform enabling rural dwellers with no broadband coverage to access to added value services and interactive services.

II.1. DVB-T2 Network designing in Burkina Faso

II.1.1. Burkina Faso territory, Population and ICTs presentation

The following data are presenting the physical territory area, the population growth and the development of ICTs (Information and Communication's Technologies) in Burkina: These data were given in 2014 year, by the World Bank website [29].

The estimated population of Burkina Faso is about 17 419 615[30] habitants with an annual growth rate of 2.8% [31]. This population is composed as follow:

- 50.2% of women

- 79.7% of dwellers that makes 12 363 74 of habitant living in campaign

- 45% of the total population is 0-14 years old and represents the most active ICTs users.

The population average density is 51.8 habitants per km².

The administrative territory organization of Burkina Faso is made of:

-13 regions, 45 provinces, 351 departments, 302 rural districts, 49 Urban districts and 8 435 villages.

Table II-1: Burkina Population and ICTs statistics

INDICATORS	2010	2011	2012	2013	2014
Internet and fixe Broadband Subscribers	13 705,00	14 063,00	14 328,00	12 962,00	5 381,00
Mobile phone subscribers	5 707 850,00	7 682 100,00	9 976 105,00	11 240 886,00	12 496 391,00
Population access to electricity in %	13.10	13.10	---- -	---	---
GDP (Gross Domestic Product) in annual %	8,45	6,52	6,45	6,67	4,05
population density	56,80	58,46	60,16	61,90	63,67
Fixed line telephones	143 963,00	141 529,00	141 358,00	137 421,00	124 595,00
GDP per habitant ($ USD)	577,85	670,45	678,37	716,06	720,05
Population	15 540 284,00	15 995 313,00	16 460 141,00	16 934 839,00	17 419 615,00
Most populated town (capital Ouagadougou)	1 914 162,00	2 059 511,00	2 215 896,00	2 384 157,00	2 565 194,00
Dwellers population	12,32	12,88	13,46	14,08	14,73
Safety internet Servers	4,00	10,00	10,00	14,00	11,00

Source : http://data.worldbank.org/indicator/SP.POP.TOTL/countries/BF?display=graph

Regarding the data contained in the above table, Burkina Faso appears as a very weak country in terms of economy, and is classified among the less advanced countries group in the world according World Bank program.

It is also noticeable that the major part of the population is living in rural areas with a few access to electricity and broadband. Thus the challenge for as to give them chance to access not only to information but to access to ICTs applications by covering the country with digital television technique and by enabling interactive services.

II.2. Analog Television network presentation

In Burkina Faso, public Television was created in 1963 three years after its colonial independence; The broadcasting network up to now is composed of 30 analog transmitters dispatched all over the territory and whose powers are varying from 30 Watts to 2KW. The territory coverage is estimated at 52% in 2015 in VHF band.

14 of these transmitters are collocated in national telecommunication operator infrastructures; consequently the towers sharing are often in disadvantage of television which transmission antennas are low height located, resulting in a few area coverage. In normal conditions, the analog broadcasting network was design to cover 80% of national territory but due to transmitter's amplifiers recurrent failures and bad transmission infrastructures sharing with the national telecommunication operator, the actual coverage is estimated to less than 50%.

The poor television service coverage does not enable the major part of the population to access to information's nor to communication services.

The transition from analog to digital television is appreciated as an opportunity to remake a new television transmission network creating its own infrastructures with the reasonable dimensions and location space.

Apart public television, 24 [32] private televisions are coexisting with public television mainly in the capital city and very few in the others regions.

In this point of view, DVB-T2 standard is chosen to implement digital television terrestrial and the analog network must be extended to 35 transmitters to attain efficient transmitters dispatching across the territory, that is to say 5 five more transmission sites are to create in addition to the previous 30 analog transmitters, in new area of the country. The map hereafter shows the analog transmitters dispatching across the country and the planning 5 new transmitters to add on analog network.

II.2.1. Digital Television Terrestrial DVB-T2 network design

The present DVB-T2 network target a UHF band utilization in its portion comprised between 470-694 MHz, such as lastly adopted during the World Radio Conference 2015 in Geneva (WRC-15).The frequencies are to be planned in Multi-Frequencies Mode (MFN).

II.2.2 Territory coverage strategy

As seen in the previous section the analog television transmission network does not cover the entire national territory, for this reason 5 additional transmitters are needed and are to be located in areas which do not receiving at all analog television signal. Creating new transmission stations, implies new towers installation, in this regard, 21 guyed towers with a height varying form 120 meters to 150 meters according to their geographical location in relation with the territory borders.

The transmitter's power is chosen in comparison to analog transmitters power, keeping in the mind that analog transmitter power is the half of digital one for same area coverage.

Taking in account that inter-carrier-interference could arises with neighboring countries (Cote-D'Ivoire, Ghana, Togo, Benin, Niger and Mali) transmission antennas are directional orientated at the transmission stations located near the borders.

The summary of territory coverage strategy parameters chosen by myself are given in table below:

TABLEAU II-2: New created broadcasting stations in addition to the 30th existing stations:

Order	Stations location	Transmitter nominal power	Tower height (m)	Antennas directivity	GPS Coordinates Alt / L. N / L. W	Frequency / channel allocation
01	Mangodara	1KW	120	Directional	294/9°55'/4°21'	
02	Solenzo	2 KW	120	Directional	325/12°08'/4°03'	
03	Arbinda	1 KW	120	Omnidirectional	315/10°55'/3°15'	
04	Gayerie	1 KW	120	Directional	292/12°40'/0°29'	
05	Sebba	1 KW	120	Omnidirectional	294/09°55'/4°21'	
06	Bâtié	200 W	120	Directive	279/9°53'/2°56'	

TABLEAU II- 3: Relocation of existing stations sharing broadcasting infrastructures with national telephone operator

Ordre	Stations location	Transmitter nominal power	Tower Height (m)	Antennas directivity	GPS coordinates Alt / L. N / L. W	Frequency / channel allocation
01	Gaoua	4 KW	120	Omnidirectional	310/10°20'/3°10'	
02	Dédougou	2 KW	120	Omnidirectional	340/12°26'/3°25'	
03	Boromo	2 KW	120	Omnidirectional	286/11°43'/2°58'	
04	Diébougou	2 KW	120	Omnidirectional	315/10°55'/3°15'	
05	Dori	4 KW	120	Omnidirectional	298/10°33'/4°46'	
06	Tenkodogo	2 KW	120	Directional	302/11°46'/0°25'	
07	Fada	4 KW	120	Omnidirectional	293/12°03'0°24'	
08	Ouahigouya	2 KW	120	Omnidirectional	329/13°34'/2°25'	
09	Nouna	2 KW	100	Omnidirectional	273/12°44'/3°52'	
10	Banfora	1 KW	120	Omnidirectional	300/10°35'/4°46'	
11	Kaya	1 KW	120	Omnidirectional	312/10°35'/4°46'	
12	Koupéla	700 W	120	Omnidirectional	306/12°11'/0°19'	
13	Yako	1 KW	120	Omnidirectional	331/12°56'/2°14'	
14	Po	200 W (1+1)	120	Omnidirectional	325/11°10'/1°10'	
15	Léo	4 KW	100	Directional	347/11°06'/2°06'	
16	Tougan	1KW	120	Omnidirectional	307/13°06/3°02'	

Note: The above mentioned caracteristiques are to be completed by spectrum regulator who will assigned bandwidth channel and UHF central frequency for each transmission station, in accordance to MFN network frequency planning.

II.2.3. DVB-T2 network modes

The present DTT is designed for fixed rooftop reception and portable reception; it must carry out 10 television programs with national content and 5 programs which are local contents, produced in regions to be broadcasted in the given region.

Assuming that regional contents are limited to the region's area and also assuming that the maximum distance between two transmitters located inside the same region is roughly 70 Km**, MFN** mode becomes suitable instead of SFN (Single Frequency Network) mode. Indeed SFN mode is more frequency saving than MFN but due to the minimum distance that must be respected (70 km at least) when using SFN mode it would be impossible to avoid ICI (Inter-Carrier-Interference) with neighboring stations in the same area.

II.2.4. DVB-T2 transmission parameters

- **Channel bandwidth**

The analog television was SECAM (Sequential Color with Memory) standard with 8 MHz bandwidth in VHF band for public television broadcasting and PAL (Phase Alternated Line) standard with 8 MHz bandwidth in UHF band IV and V for private television broadcasting.

With this regards, the channel bandwidth remains the same, that means 8 MHz but the frequency band is harmonized to UHF band at 470 to 694 MHz. The color system is also be harmonized to PAL.

- **Video format**

The Standard Definition (SD) is defined as a prior video format definition; however High Definition (HD) can be examined and be accepted in the multiplex if any.

- **Digital video Compression format**

The aim of this project is to enable carrying out a large number (15) of television programs, it will be mandatory that the final produced television program to be encoded in **H265 or HEVC**, recognized by ITU as **MPEG-H part II**, this is the latest efficient encoding adopted by ETSI. It enables earning more than 40% bit rate reduction than its predecessor H264 / MPEG4-AVC: it potentially works with SD, HD and even UHD.

- **Modulation scheme**

The planned modulation scheme obviously is a digital one: a Quadrature Amplitude Modulation of 256 constellation size **(256-QAM), without rotation** this option is to maximize the data rate transmission that can attain **50 Mbps** for a code rate fraction

of 5/6 according to ITU data. More the constellation size is high more the spectral efficiency is better.

The chosen FEC (Forward Error Correction) fraction seems to be high and fragile for signal protection (One redundant bit for 5 transmitted bits) in the other hand the carrier power is high resulting to a high value of Carrier to Noise ratio (C/N) dB: as previously discussed.

- **Fast Fourier Transform Size (FFT–size)**

This parameter determines the number of subcarriers to use in multicarrier modulation scheme; it can be a normal size or an extended size. Using a normal size of FFT, means to remain in the allocated channel bandwidth with a relative high power of spectral density. But in case of Extended FFT size especially in case of using **32K Extended**, the power of the spectral density is very low: around **-58dB** compared to 16Kor 8K where this value is higher**: - 50 and - 45 dB.**
With 32 K Extended, the number of subcarriers is **27.841** yielding a bandwidth gain of **2.11%.** The large number of subcarriers carrying out information in one time makes the transmission reliable in multipath environment. For delivering high-bit-rate services to fixed rooftop antennas, in VHF or UHF, the 32k FFT mode is more appropriate. With that regard time variations in the channel are minimized, and 32k offers the very highest [33] bit rates achievable using DVB-T2.

- **Guard Interval (GI)**

It ensures avoiding Inter-Symbol-Interference (ISI) and Inter-Block-Interference, it must be taken higher than the time which separates the transmitter and the receiver, and this time is the result of distance over light celerity. That means to get a long GI, the distance between receiver and transmitter must be longer.
Assuming that the useful time of one period $\mathbf{T_u}$ in (µs) for 32 K FFT size mode given by ITU is **3584 µs ($\mathbf{T_{u\,=}}$ 3584 µs),** the corresponding longest time is the **GI fraction** of **19/128** that gives **GI = 532 µs .**

A larger FFT size such as 32K, implies a longer symbol duration, for this reason $T_{u\,=}$ 3584 µs seems to the best choice in our case, which means that the guard interval fraction is smaller.

- **COFDM features**

The technique to ensure multicarrier modulation will be OFDM type that uses Discrete Fourier Transform (DFT) enabling a summation of s(n) complex symbol's functions to form OFDM signal called **Frame:**

The length of the given frame depends on QAM size, the data bit rate and the FEC blocks per frame: For effective tradeoff between those parameters, ITU has recommended some values pointed out in the table **II-4 above**, assuming the data of this table, 256-QAM, 32K, with 5/6 FEC corresponds to a **Frame length of 60**.
Channel coding to protect transmission will be achieved throughout **LDPC-BCH codes of 5/6 ratio.**

- **Pilot Pattern choice**

The choice of pilot patterns (PP) determines the performance for delayed signals arriving outside the guard interval: With respect to Nyquist limit, a delayed signal received outside the GI cannot be equalized although the Inter-Symbol-Interference fraction is small.

In case of rooftop reception with a directional outdoor antenna such as the planned reception system, Doppler effect is low. Therefore **PP7** that is less robust to Doppler effect and with low overhead pattern is chosen to maximized channel capacity.

The planned network combined both rooftop and portable reception in urban areas: Assuming that in a portable reception, the channel characteristics do not change rapidly so it makes sense to have less overhead pattern, for these reason**, PP7** remains suitable for both fixed and portable reception**.**

II.3. Multiplex capacity evaluation

The planned DTT network is to be designed for SD video format with an aspect ratio of 4:3 however a space for 01HD video format must be reserved for any sudden need.

II.3.1. bit rate allocation

To plan multiplex total bit rate, we need first to plan audio video an ancillary data bit rate per channel, from there will derives the multiplex bit rate.

❖ **Standard and High Definition channel bit rate determination**

Expected video compression format is H265 or HEVC/ MPEG-H part 2, and HE-AAC as audio compression format, with this regard, the following data rate are summarized in the table below:

Table II-4 Data bit rate details

MPEG-H Part 2 or H265/ HEVC SD service in Kb/s	
SD video	1370
Stereo Audio (HE AAC)	128
Teletext	80
DVB subtitle	92
Total Audio Video SD service	**1670 (1.67 Mbps)**
MPEG-H Part 2 or H265 / HEVC HD service in Kb/s	
HD video	4500
Stereo Audio HE AAC 2.0	192
Teletext	120
DVB subtitle	100
Total Audio and Video HD service	4912 (4.912 Mbps)
Common (Ancillary) data in Kbps	
PSI (PAT, PMT, CAT)	200
SI (NIT, SDT, TOT/TDT, MIP)	20
SI EIT pf + EIT schedule	300
CA ECM	200
Software Download	30
TOTAL Common Data	750 Kbps
Radio broadcasting channel	92 Kbps

❖ **Data bit rate summary**

- **SD** audio and video bit rate per program channel (720x576i):

1.67 Mbps+ 750 kbps **= 2.42 Mbps**

- Radio broadcasting channel bit rate:

0.92 Mbps

- Expected, available SD television programs in 2016: **18**
- Required bit rate: 18 x 2.42 = **43. 56 Mbps**
- Expected Radio broadcasting Channels in 2016: **02**
- Bit rate need for radio: 02 x 0.92 = **1.84 Mbps**
- **18 SD TV + 02 Radio** will require: 43.56 + 1.84 = **45.4 Mbps**
- Adding **01 HD TV** (1920x1080i) channel will provide 45.4+ 4.912 = **50.312 Mbps**

According to ITU recommendations contained in the *Table II-4*, the maximum recommended bit rate is **50.324 Mbps,** other said**,** the planed multiplex bite rate is acceptable, it is not out of range. Although we are using H265 compression standard, the allocated bit rate per program, remains slightly high, moreover a statistical multiplexing will be performed in order to improve transmission efficiency that means no bit rate allocation for idle program. In general, once a week is broadcasted a sport program, the rest of the time, movies and talk shows programs are broadcasted.

In terms of required encoders number, using HD/SD–SDI H265 encoders with at least 2 video inputs and 8 stereo inputs will yields **10** (9+1) encoders**+02** redundant encoders.

About the multiplexer, 01 active multiplexer with at least **20 ASI** inputs and a redundant **one** in standby switchable by automatic change over.

II.3.2. DVB-T2 National interactive platform architecture design

II.3.2.1. National Head-End architecture design

The national DVB-T2 network will take in account the possibilities to deliver at a time broadcasted program and streaming programs that means live or downloadable programs, for this reason the national Head-end must be designed so that it can achieve both broadcasted and streamed programs.

Satellite reception system will be needed to ensure program reception from abroad: these programs are performed by content providers for added value purpose.

For that an IRD with descrambler capability will be needed. Another decoder ensuring IP streaming content will provide streaming content from providers utilizing transmission over IP.

A high capacity server will stock added value programs which need to be downloaded by consumer, other way some of those programs will be live transmitted. In addition to added value services, the content provided by Editors will be classified into three main parts: Governmental Public television content, private's televisions content and regional public as well as private's televisions content.

The particularity of regional content is that: Those contents must be broadcast only in the region's area in addition to national content programs. Taking in account this particularity, regional subhead-ends will be installed in each Head-region's place.

The subhead-ends are planned for **5 local contents** in addition to **10 national** contents. In total, fifteen programs are planned to be broadcasted in regions.
Technically, three Physicals Pipes Layers (PLPs) will be used: One for public television content, the second for private content and the third for HD program channel:
In the regions, a T2-Edge will enable to combine national and local content.

- **Program's content transportation means**

According to the actual situation, optical fiber as transmission channel is unlikely not recommended: The first reason for that is the very few existing optical fiber network across the territory. A project to create a national backbone is now discussing, it may be later a greatest mean for digital data transportation comprising DTT transmission.

The second reason why the existing optical fiber network is not recommended is that at any time this network is frequently damaged causing troubles or interruptions on communications' systems. This derives from farmers activities in campaign and other human behavior.

In connexion to the above inconvenient, satellite will be first used for national multiplex content link to head-region's places. And secondly microwaves will be used to link national head-end and also to link inter-plaques (stations from the same region's area)

In one word, the transmission channel is planned to be a mixed one, satellite and microwave: more than transmission channel, the satellite will play a double role: linkage function and broadcasting function. The linkage function will be done in C-band because of severe atmospheric conditions (dust, heavy rain, 40°C Temperature etc.) in contrast the broadcasting function will be performed on DTH (Direct To Home) in Ku-band. That will complete the non-terrestrial coverage in areas where terrestrial coverage is impossible because of the relief. The DTH will also enable to target international coverage where the diaspora is important such as European's region, Middle East region and later American's continent.

The block diagram of the described DVB-T2 platform is presented as follow:

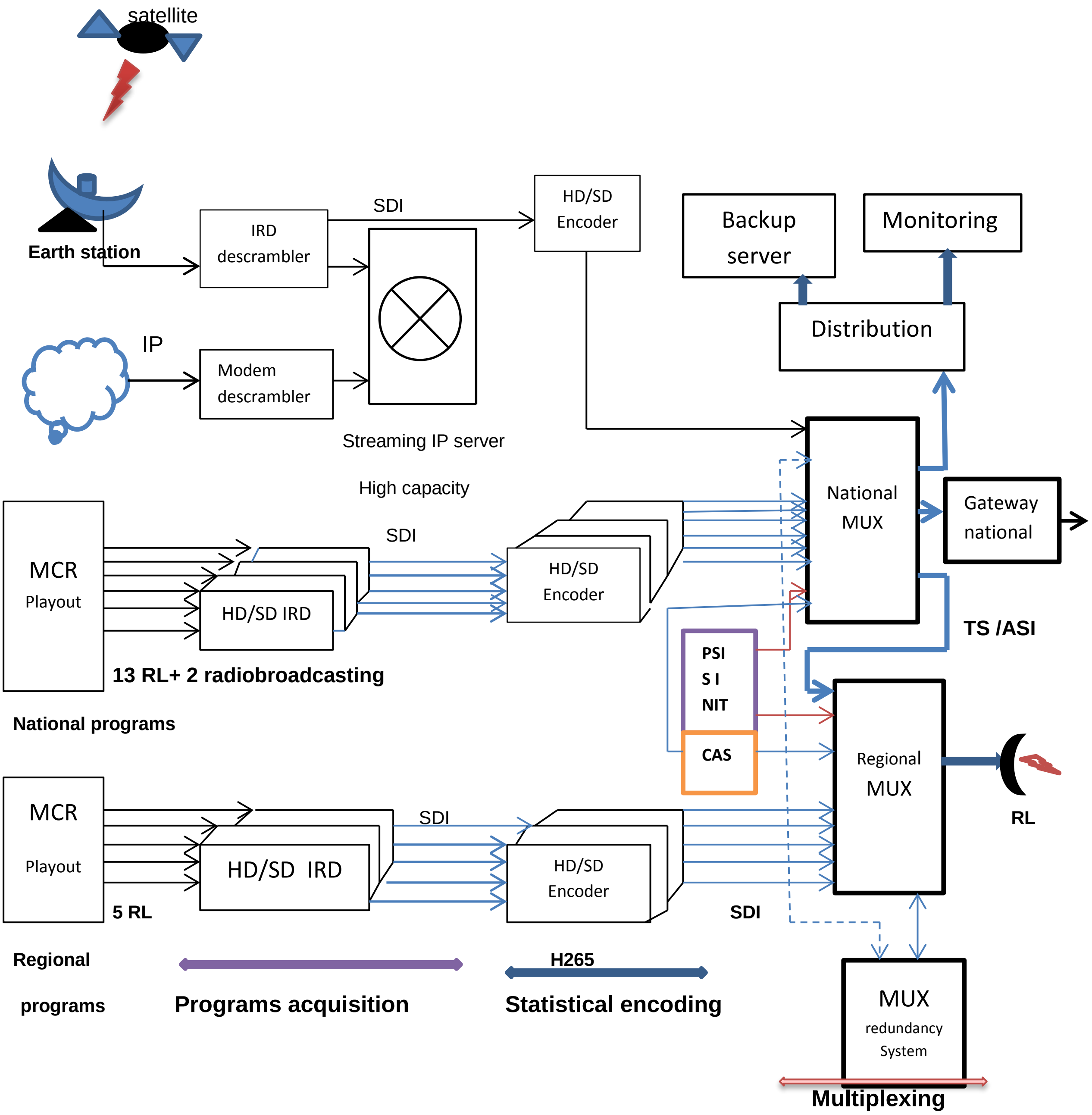


Fig.II-1: DVB-T2 headend Platform block diagram

The above block diagram shows the DVB-T2 signal processing, starting from programs acquisition, these programs are received from different sources: Those from local editors are conveyed from Masters their Control Rooms (MCRs) throughout Radio microwave Link (RL) to the head end.

Streaming programs are stored in a dedicated server from internet protocol (IP): These programs are payed programs for added value purpose.

A part of satellite programs are to be mixed with live broadcasted programs while the other part of the same programs will be stored for streaming purpose. Programs received from satellite are considered as abroad programs, a contract must be established with their providers and will not be free to air.

A backup file server, receiving multiplexed signal will enable archiving and catch up television programs.

The programs dedicated for national coverage, at the output of the multiplexer, the obtained MPTS (Multi-Programs Transport Stream) signal is conveyed to a gateway for T2-MI signal generation.

The programs *dedicated (13+5TV programs+2 radios)* for local regional broadcasting (the area around the capital) are output to a Radio-microwave link towards the DVB-T2 transmitter located at about 10 km far away from the national head-end building.

h) II.3.2.2. Regional Head-End architecture design

To provide head-region's places with multiplexed programs signal, satellite will contribute conveying this signal to the sub-head-end in regions'. For that, a gateway at the national head-end will first generate a T2-MI signal to transmit over C-band satellite, the downlink from this satellite will be multiplexed to the regional programs provided locally by each regional editors. The architecture of the regional head end can be designed as follow:

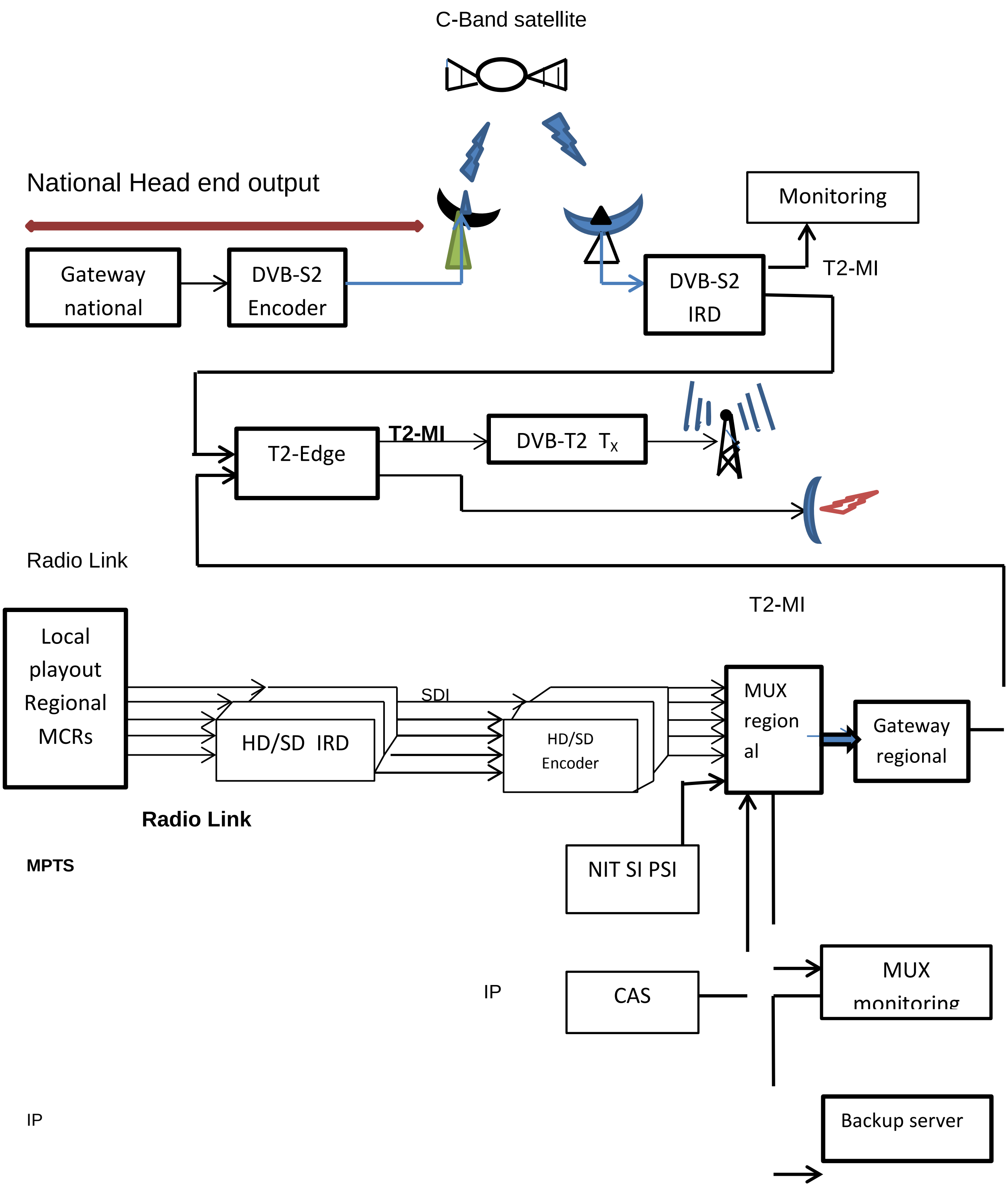


Fig. II-2: Regional head-end architecture block diagram

The above regional head-end architecture shows that the multiplexed programs in regions' are composed of national programs (13 TV +2 radios) plus 5 regional programs. National programs are transmitted to regional head-end over C-band satellite transmission channel.

Then the 5 local programs are transmitted to the regional head-end via radio-link. The multiplexed local programs are inserted in a gateway to create local PLP: As it is no more possible to re-multiplex local and national PLP, without destroying T2 national signal, T2-Edge equipment capable to combine PLPs, combining gateways is needed.

The T2-Edge creates at its output, a T2-MI signal aggregating local and national PLPs. The resulting T2-MI signal is then connected to a DVB-T2 transmitter to be processed and broadcasted over local area.

At the same time, this T2-MI signal is transmitted via microwave to ensure linkage between neighboring DVB-T2 transmitters, inside the region's territory.

Ancillary data, such NIT (Network Information Table), SI (Service Information) PSI (Program Service Information), CAS (Conditional Access Service) etc. are inserted in the multiplexer to enable receiver to process accurately received data.

II.3.3. Conclusion:

Designing the architecture of DVB-T2 platform for national head-end as well as for regional head-end enables to establish the transmission channels to combine for data transmission purpose. In this manner, it gives an idea of programs transmission cost to be planned as well as the required data rate.

As the final aim is to enable interactive programs, streaming servers are planned to enable downloaded programs throughout IP. In the other hand, live interactive programs from satellite content providers, are inserted in the multiplex together with playout programs, incoming from terrestrial content providers' Master Control Rooms.

CHAPTER III: HbbTV as Interactive Television solution

The challenge to overcome here in designing this project is to enable dwellers to benefit interactive programs whatever they do not have broadband coverage. The aim of doing that is linked to that: social and economic development is related to infrastructures' development as well as communications' means, regardless the environment conditions.

The advent of Digital Terrestrial Television (DTT) must be considered for us, as an opportunity to catch up to extend and to develop electronic communication means in poorest and distant areas. Thus Hybrid broadcast broadband Television (HbbTV) proposal as interactive application solution in DTT. This application works if and only if there is internet connection to a receiver device (DTV receiver or DTT set top Box) moreover that needs a broadband internet connection.

To perform HbbTV in DVB-T2 platform it first makes sense to analyze the broadband coverage over national territory.

III.1. Broadband network and population access

The data contained in table IV-1 below recall some important data related to broadband subscribers evolution in year time, mobile phone subscribers and population access to electricity.

Table III-1: population subscription to broadband and electricity

INDICATORS	2010	2011	2012	2013	2014
Internet and fixe Broadband Subscribers	13 705,00	14 063,00	14 328,00	12 962,00	5 381,00
Mobile phone subscribers	5 707 850,00	7 682 100,00	9 976 105,00	11 240 886,00	12 496 391,00
Population access to electricity in %	13.10	13.10	---- -	---	---

The mentioned data in this table are from World Bank web-site as said in previous discussed sections.

The analyses of data in this table indicate the growth of fixed internet subscribers between 2010 and 2012, from 2013 the fixed internet subscription is decreasing, this situation can be explained by the fact that most of people now prefer wireless internet connection than fixed one: Thus the rapid increases of mobile phone subscribers: wireless internet connection does not concern only mobile phone owners' it concern also all connected electronic devices such as smart phones, tablets , IPad, PCs etc.

Regarding the increases of subscribers to mobile devices it makes sense to extend internet access to dwellers in order to add the list of connected electronic devices, DTV sets or DTT set-top-boxes.

Another important parameter to take in account designing this project is population access to electricity, indeed it doesn't make sense to spend lot money to provide new technology equipment while there is no electricity to fit with.

The above table shows 13.10% of the population in 2011 who have access to electricity energy that represents 1.965.000 habitants for a total population of 15.000.000 in 2010[34] .

From 2010 data, we can extrapolate those data according to government national economic and social development program (SCADD-PNDES) [35]: one of the priorities of this program is to bring electricity to rural areas by 2020 year started in 2010. In 2016, the population access to electricity can be estimated at more than 6.000.000 habitants. Solar energy project is also being running across the country enabling rural dwellers to access to renewable energy. The combination of primary and renewable energy will boost the access to electronic devices uses in those areas: Thus the reason to bring them high-tech devices by HbbTV in DVB-T2 platform.

III.2. Interactive HbbTV platform design

III.2.1. Definition

HbbTV[36] stands for Hybrid broadcast broadband Television that enables consumer, in addition to linear broadcasted TV content via satellite, terrestrial TV, cable or IPTV, to accede to non-linear streaming services from internet applications such as VoD, catch up TV, games, voting, sports events statistics and scores, asking question to a guest or host during a show, etc.

The main difference and advantage is that, the broadcasted content is controlled by broadcasting operators and cannot be overlapped by any other user in contrast to pure internet television interactive applications.

The advent of digital television raised consumers demanding; more than pictures and sound quality requirements the pressure put on digital television consumer, led respectively from portable reception to digital mobile reception.

Consumers' have introduced a new requirement that is digital device interactivity. Indeed, digital television receivers nowadays are technologically designed to be interactive: for that purpose, various interactive technologies are available on consumers market. Among them, digital television receivers named Hybrid broadcast broadband television, In short HbbTV.

Those types of receivers are interactive technology capability: Through that technology, consumer is no longer a passive TV program viewer but he has the opportunity to interact with the program throughout the screen.

Moreover, he is no longer constraint to be on time to watch live broadcasting TV program; he can schedule his own TV program at any time at his convenience with more details added to linear live broadcasted program.

III.2.2. HbbTV concept

The consumers market of internet connected devices is unlimited increasing, as it is noticeable, the technology of smartphones, games consoles, Blu-rays, STB, smart TV etc. enables users' interactivity via internet path. The estimated number of connected devices is about one (1) billion[37] in 2015 across the world (12 496 391,00 in Burkina), inducing an open competition within device vendors, as well as manufactures that are expanding their technology to support a wide range of non-linear services and interactive capability, thus HbbTV.

HbbTV has MPEG-DASH function, which is a Dynamic Adaptive Streaming over HTTP standard, enabling dynamically to optimize the picture quality in accordance to the program content that rate is not constant due to allocated bandwidth and program nature (sport, show or movie etc.). During a sport program the needed data rate must be higher than for example a talk show.

Program where pictures movement is few: the MPEG-DASH application will optimize compensating the data rate in between programs to enhance services. Moreover it is possible to access to DVB Event Information Table (DVB EIT) and schedule an Electronic Program Guide of seven days duration. The option of HbbTV that can perform above described functions was standardized as HbbTV V1.5.

The latest version of HbbTV is the version **HbbTV V2.0**. This is the recent version released in February [38] 2015 , it was focused on features enhancement especially on video and audio compression processing improvement, HbbTV V2.0 henceforth support HTML5 (new application language) and HEVC video compression in order to improve spectral efficiency and save transmission bandwidth. Obviously, this leads to transmission costs reduction, as we know, usually transmission is charged according to occupied bandwidth in MHz, few bandwidth occupied means low costs.

Broadcasted programs and broadband (internet path) services follow different ways to reach destination (TV set), HbbTV V2.0 enable synchronization of broadcast and broadband content.

Using HbbTV V2.0, possibility is given to synchronized content to be split over one screen or several screens. The version V.2 was improved to support MPEG DASH and Push VoD.

III.2.3. HbbTV and DVB-T2 platforms architecture designing

The target here is to combine two digital television technologies in order to deliver linear and non-linear broadcasted contents. Non-linear content is the one delivered on broadband path and linear content is that delivered on conventional television broadcasting way. So, the architecture of combined platforms must be designed in a manner to implement both linear and non-linear contents. The insertion of HbbTV services in DVB-T2 unfortunately will increase bandwidth consumption for about 15% of the total DVB-T2 multiplex bandwidth requirement. This can be explained by the insertion of applications such as: Interactive applications Table (IAT), Triggers and signaling. Designing the architecture implies estimation of total required data rate.

Knowing the total data rate, it makes sense to achieve our goal to provide not only linear content all over the country territory and abroad but to deliver in addition non-linear content in areas with no or few broadband coverage.

If the challenge to deliver linear content through conventional digital terrestrial television broadcasting can be successful achieved regarding the previous network deployment strategy, it is not the case for broadband content broadcasting.

As broadband does not exist anywhere in the country, a solution must be found to ensure the demand.

Analyzing multiplex signal transportation means, it is shown that satellite will contribute providing signal to distant transmitters in regions: This opportunity must be taken to optimize satellite capacity that means, evaluate satellite bandwidth occupation in terms of multiplex bandwidth + auxiliary data + Interactive data. Then add internet broadband capacity and HbbTV service data rate.

In this manner it makes sense to estimate to total bandwidth to lease on the satellite taking in account HbbTV bandwidth requirement: Indeed the first version of HbbTV was given up and replaced by HEVC [39] codec more efficient than H264 codec. The introduction of this new encoding technology in HbbTV features was published in

2013 by ETSI [40]. HEVC codec can process the same quality of pictures and sound at half [41] the bitrate of the H.264/AVC.

Due to HEVC codec performance, the required bandwidth of HbbTV services in area with broadband coverage can vary only between 100-300 Kbps.

In contrast, in regions with no broadband coverage HbbTV services bandwidth r equirement can attain (2-3) [42] Mbps.

Base on the above information's, the total bandwidth required for DVB-T2 platform and HbbTV service and broadband data to provide to no coverage areas is assessed as follow:

- Multiplex signal and data bandwidth:

According to the previous discussed multiplex characteristics, 18 TV programs are planned, 5 of them will be broadcasted locally and 13 others will be broadcasted across the national territory: To provide national multiplex signal to DVB-T2 transmitters all over the territory, satellite will be used as transmission channel.

Thus the estimation of satellite capacity to lease.

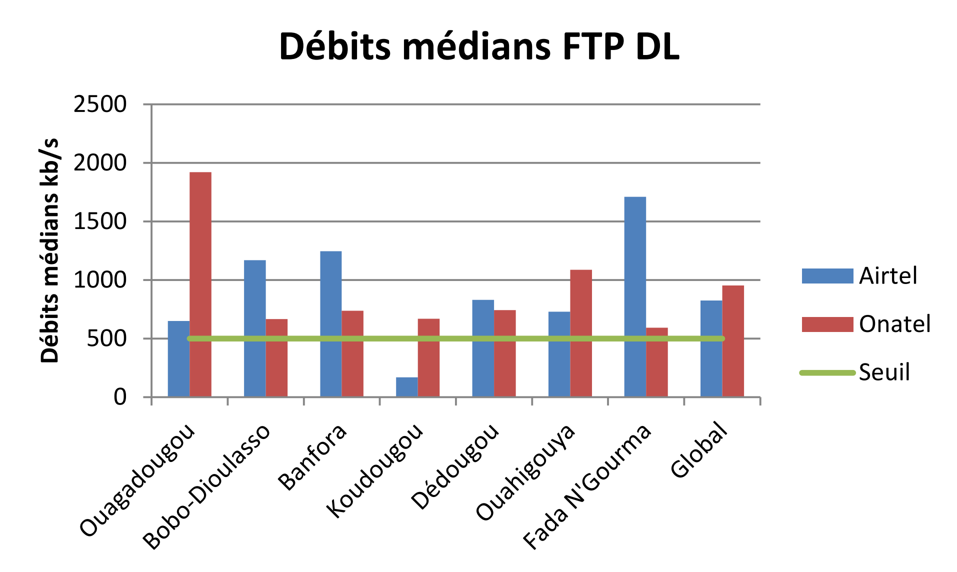
- SD TV programs to transmit over satellite: **12**
- SD programs required data rate: 12 x 2, 42 Mbps = 29, 04 Mbps
- HD TV program to transmit over satellite: **01**
- HD program required data rate: 01 x 4,912= 4,912 Mbps
- **02** Radio programs data rate: 02 x 0, 912= 1, 84 Mbps
- Linear broadcasted Interactive services (CA):750 Kbps
- HbbTV services data requirement: 3 Mbps

Total required data rate on the satellite = 39,542 Mbps

With regard to the estimated data rate to transmit over satellite, it is suitable to analyze broadband data that is required to provide to non-coverage areas.

Three mobiles telephone operators are providing both voice and internet broadband connection in Burkina. The delivered broadband service is '3 G 'and represents 58% of the territory coverage. The data rate for provided services to users' is estimated at 500[41] Kbps in download as indicated in table IV-1.

Table III-2: 3 G performances in Burkina main cities [43]



Hence, for interactive services; the required data rate will be estimated at 5 Mbps in upload and 500 Kbps in download. That mean, the total data to transmit over satellite will be: 39,542+ 5,500 = **45, 042 Mbps**. With regard to the total required data rate, it makes sense to lease the corresponding bandwidth on the satellite to transmit both DTT multiplexed data and broadband data.

Assuming that 1Mbps cost 2800, 00 [44] USD per month, the required data rate will cost: 45, 042 x 2800 = 126 117, 6 USD per month that means: 1 513 411, 2 USD per year, to be charged by DTT broadcasting operator.

The combination of HbbTV and DVB-T2 simplified platforms is shown hereafter:

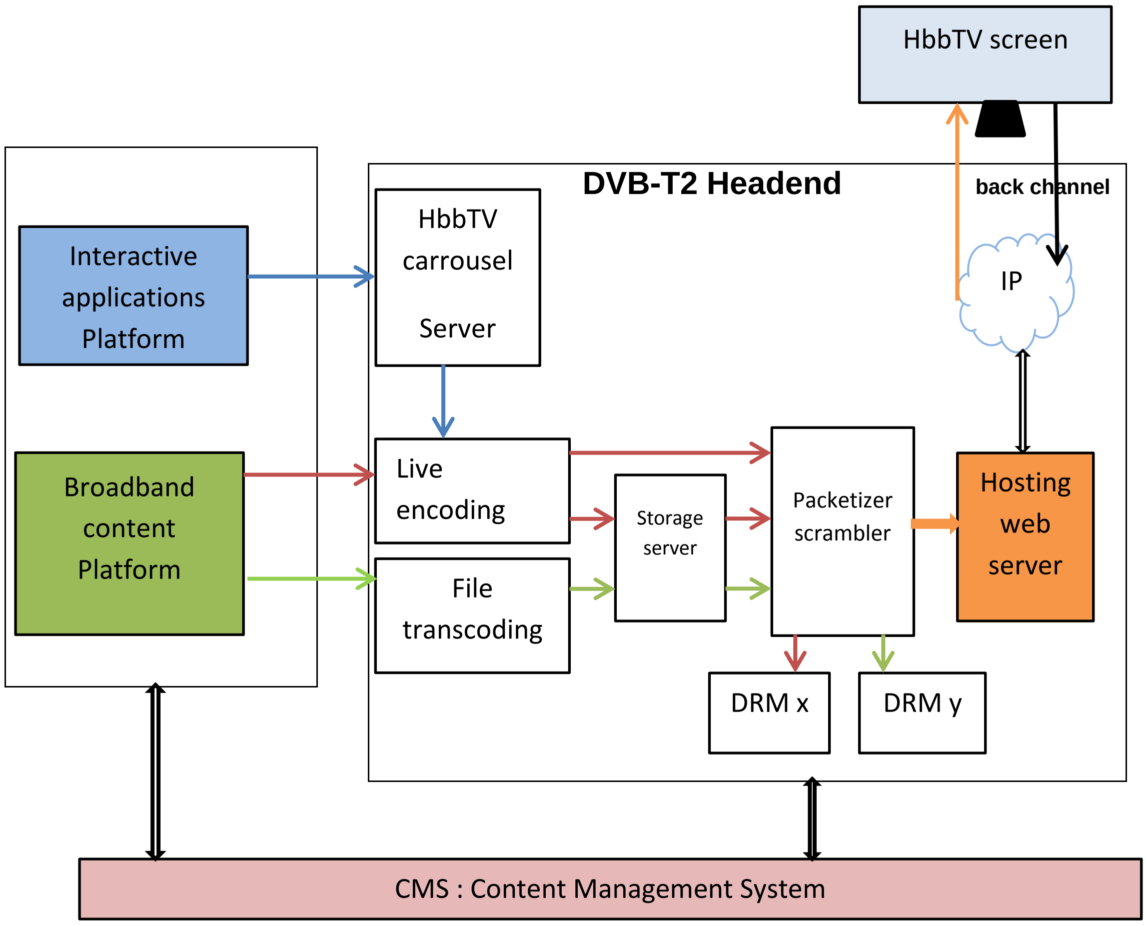


Fig. III-1: HbbTV and DVB-T2 platforms simplified architecture

DVB-T2 platform was designed in previous chapter, the matter here is to insert HbbTV platform at the headend of DVB-T2 designed platform: The target aim is to enable linear streaming content as well as non- linear content.

Broadband content are added-values content, for that the content is controlled through a Content Management System (CMS) then scrambled for both linear and non-linear content , thus a Digital Right management on linear (x) and on non-linear (y). HbbTV Interactive applications are added from a carousel server to enable HbbTV set or DTT Set Top Box (STB) through Internet Protocol (IP) to accede to those applications and inter-act via return channel.

The delivery range of HbbTV applications are as follow:

- EPG linked and program preview
- Backward EPG for Catch-up
- Voting, asking question for live transmission program

- Enhanced teletext
- VOD, DVR
- News (text & graphics), Linear streaming TV and Radio channels
- Advertising on the portal pages
- Pre rolls Advertising on start over and catch up
- Social TV Shopping portal and social networks (Facebook, YouTube, Twitter etc.)

-Games, weather information's, e-government, etc.

An example of capture HbbTV screen is given in the pictures below.

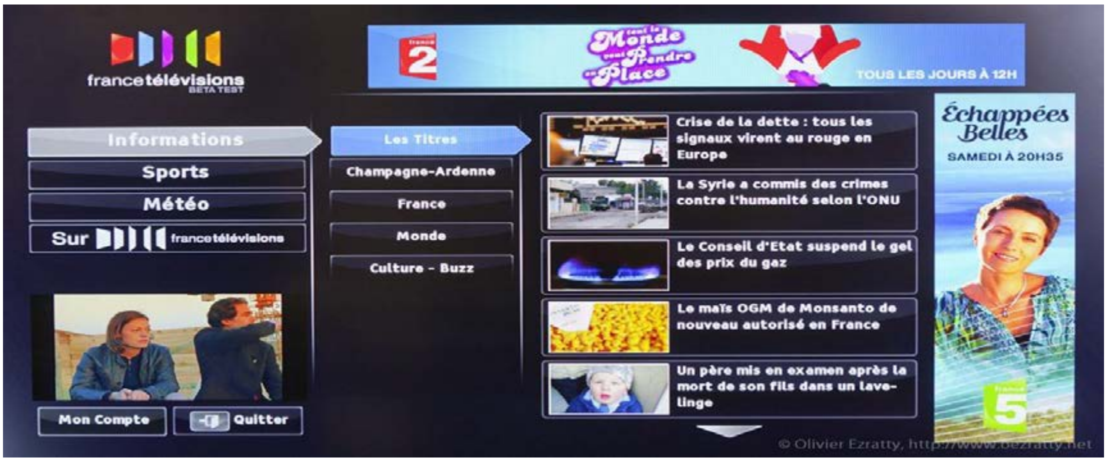


Fig.III-2: Picture screen A (Source: Thomson catalog on HbbTV solution 2014)

Fig.III- 2: Picture screen B (Source: Thomson catalog on HbbTV solution 2014)

In the pictures above, is noticeable in screen A, EPG with programs preview, while washing in a window, another program from another channel. In screen B multi-screen are selected to display different programs while possibility is given to browse Facebook or shopping advertising.

Note: Linear content are encoded by MPEG-DASH (Dynamic Adaptive Streaming over HTTP) compression system, which mean dynamic compression is achieved to adapt data rate to real content, enabling live streaming reception. Non-linear content are downloaded from hosting server.

In the other hand HbbTV interactive applications and can be also delivered through satellite transmission: Indeed multiplexed programs will be delivered not only to regional headend places but also to households via DTH (Direct To Home). The satellite segment that will achieved this service is Ku band carrying out multiplexed programs and broadband data for HbbTV service as previously discussed for required data rate paragraph.

In this manner, DTH will deliver both multiplex and broadband signal to subscribers with no broadband coverage in regions'. The conception of this architecture can be designed as shown in figure below:

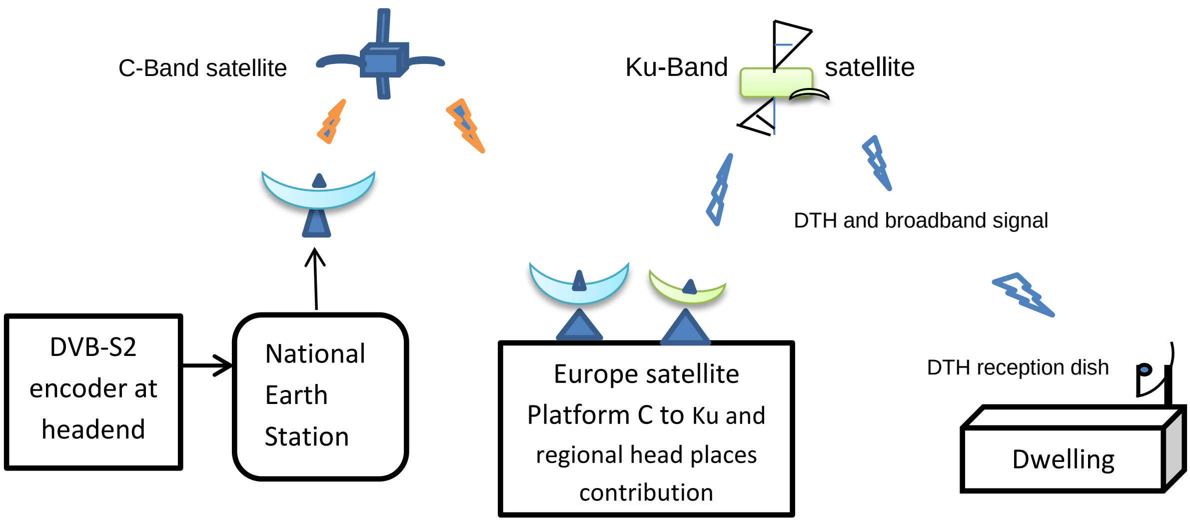


Fig.III-3: DTH and broadband delivery architecture

The T2-MI signal at the output of gateway device in the national DVB-T2 headend platform as seen in chapter III is encoded via a DVB-S2 encoder at the national headend; The DVB-S2 signal is transmitted through C-Band earth station to a dedicated satellite: In Europe, particularly in France, this signal will be downlinked in a satellite platform capable to convert C-Band signal onto Ku: then re-uplink Ku signal to a dedicated Ku satellite for DTH coverage focused in Burkina territory and west Africa area. This solution is proposed to deliver both multiplexed signal and broadband data to dwellers with no broadband coverage.

Meanwhile the achievement of national Backbone project that will provide fiber optic across the territory and broadband delivery mostly everywhere, the proposed solution will be helpful to distance dwellers.

Indeed the ongoing backbone project network can be shown in the map below:

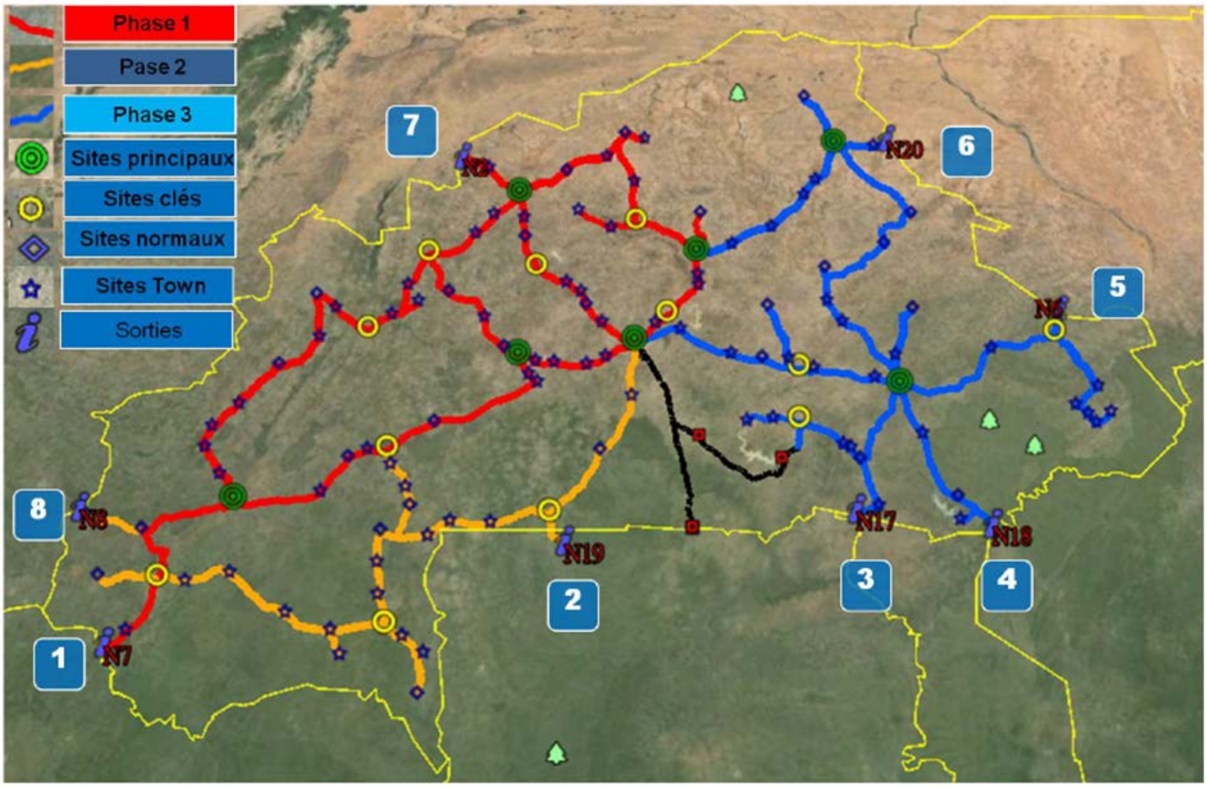


Fig. III-4: Backbone project map in Burkina

(source: Burkina Ministry of digital economic and postage development 2015)

In this map is shown the fiber optic network projected to cover the territory:

Three Phases are planned to achieve the deployment of this project: Red lines, Blue lines and Yellow lines represent the different steps of deployment. The funding of the project is estimated at **182 million USD** to be charged by government budget which has a number of social development priorities.

With regard to the huge amount to mobilize, the project dragged to start. However, the objectives targeted by this project are great: provide broadband data all over the country develop e-administration and education; enhance broadband services, in one word: make available an efficient and reliable infrastructure for ICTs users.

III.3. DTT receivers' technical specifications

After designing HbbTV and DVB-T2 platforms to deliver both linear and not linear content with interactive applications, it makes sense to propose suitable DTT receivers to consumers, enabling them to accede to these interactive contents.

For that, DTT receivers, capable to operate with internet connection or WIFI are required. Moreover, they shall be compatible to HbbTV version 2.0 and support both SD and HD content as well as multimedia content. Some relevant specifications of recommended DTT receivers are given in the table below:

<u>Table III-3:</u> Some relevant DTT receiver's technical specifications

Interface	**Connector**	**Requirement**
USB 2.0, USB 3.0	USB Standard A Socket	Minimum 1
HDMI	Standard socket	Minimum 1
Ethernet port supporting 10BASE-T and 100BASE-	RJ45	Required
Wi-Fi Adapter for Internet connectivity,	Standard socket	required
SM/3G Adapter for Internet connectivity,	USB flash	required
Mass storage device for user recordings and Push	Standard socket	required
Storage capacity	**DTT STBs**	**iDTV**
128 Mbytes Flash memory	required	required
256 Mbytes RAM	required	required
CA Card Slot	required	required
Protocol	**Containers**	**requirement**
HTTP Progressive Download	MPEG-2 SPTS, MP4	mandatory
Adaptive bitrate streaming over HTTP	MPEG-2 SPTS, MP4	mandatory
Live streaming using UDP	MPEG-2 SPTS	mandatory
Audio video standards	**DTT STBs**	**iDTV**
MPEG-4 AVC HP@L3 SDTV / MPEG-H part 2,	Mandatory	Mandatory
MPEG-4 AVC HP@L4 HDTV / MPEG-H part 2 ,	**Mandatory**	Mandatory
MPEG-1 Layer II (Musicam) / MPEG-1 Layer III	Mandatory	Mandatory
HE-AAC V2 Level 4 down mix to stereo	Mandatory	Mandatory
DVB HbbTV 2.0 / DVB-T2	Mandatory	Mandatory

III.3.1. Conclusion:

The discussed HbbTV and DVB-T2 platforms had shown the required equipment to perform linear and non-linear content delivery in areas with no broadband coverage, including the means that are essential to achieve the goal. Satellite as linkage mean, will carry out from Europe broadband data as well as DTT multiplexed programs. The backbone project discussed may takes some years further before getting a reality. Delivering HbbTV applications content to dwellers implies making available suitable DTT receivers at affordable prices. Thus the necessity for the government to subsidize DTT receivers.

CHAPTER V: General conclusion

In relation to frequency spectrum scarcity, ITU Regional Radio Communication Conference (ITU/ RRCC) held from 15th of May to 16th of June in Geneva had concluded an agreement to migrate from analog radiobroadcasting to digital radiobroadcasting by 17th of June 2015. In this regard, a number of digital television broadcasting standards were adopted and released by ITU and other regional standards organization such ETSI, IEEE.

In Europe, DVB consortium firstly developed and released DVB-T as Digital Television Terrestrial Standard. Because of consumers' requirement, DVB-T has been improved to DVB-T2, taking into consideration the impairments of DVB-T. Therefore, DVB-T2 is widely used in Europe as well as in African countries due to its high performances in terms of technology reliability, spectrum efficiency, mobility and interactivity. Consequently consumers henceforth have the possibility to interact with television programs and accede to social network applications.

For specific case of Burkina Faso It was therefore essential to study a DVB-T2 platform then match it with HbbTV platform to enable interactive delivery by linear and non-linear broadcasting content. HbbTV as one of interactive solution requires broadband availability to operate fluently; however broadband coverage in Burkina Faso is less than 15% of population: It was propose a solution of delivering broadband data, in areas where they do not exist. This technical solution is designed and will be implemented to enable youth in rural areas to benefit to advantages of Information and communication technology applications.

The advent of DTT will be an opportunity for Burkina Faso to cover the maximum of the territory with terrestrial television service, indeed the strategy of network designing forecast 98% of territory coverage, while the DTH delivery will cover 100%.

It was shown that DTT will provide a clearer picture and superior sound quality when compared to analog TV, with less interference. DTT offers far more channels: especially DVB-T2 standard uses advanced technology of compression and multiplexing, thus providing the viewer greater variety of programs to choose from and interact with. This obviously requires suitable and accessible DTT receivers by consumers.

It is known that the implementation of DTT in most of developing countries is still starting: funding to mobilize for investment stay out of national budget with a lot of priorities. It therefore makes sense to state a reflexion on business model that can in short time recover the investments. In the other hand, the speed of technology changes today may pass over the chosen technology of DVB-T2 for a new one. In the same time Digital Radiobroadcasting Terrestrial (DRT) that is not yet implemented in Africa, stays a challenge to schedule for the incoming years: thus the neediness now to take in account designing DVB-T2 architecture, DRT network architecture. DTT and HbbTV network designing shown huge investments neediness, it is therefore recommended to find out a convenient business model that can make financial revenue.

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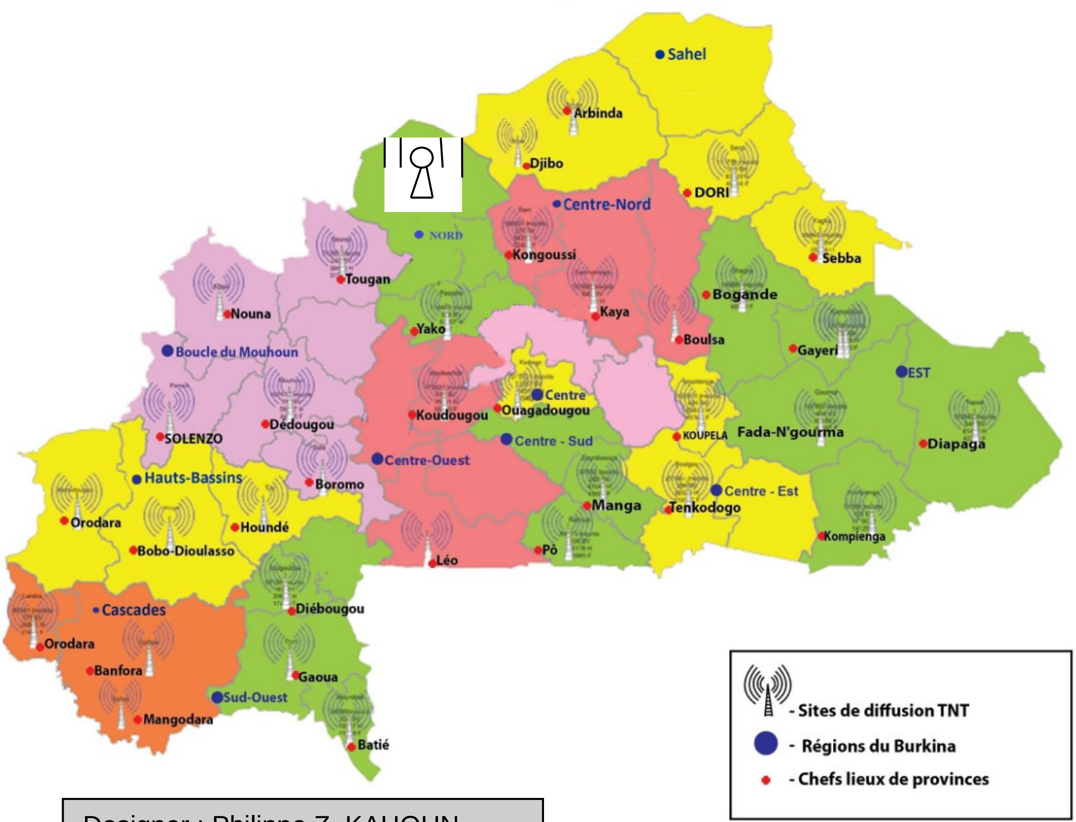
ANNEX

BURKINA FASO DTT TRANSMITTERS NETWORK MAP



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