République Algérienne Démocratique et Populaire
Ministère de l'Enseignement Supérieur et de la Recherche
Scientifique

Université Amar Telidji- Laghouat

FACULTÉ : Technologie

DÉPARTEMENT : électronique

MÉMOIRE DE MASTER

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DOMAINE : SCIENCE ET TECHNIQUE

FILIÈRE : TÉLÉCOMMUNICATIONS

OPTION : SYSTÈME DES TÉLÉCOMMUNICATIONS

Thème

Radio channel in QAM based FBMC for 5G

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Dedication:

I dedicate this brief to:

My mother, who has worked for our success, thanks to her love, her support, all the sacrifices she has made, and her precious advices.

My father, who can be proud and find here the result of long years of sacrifices and privations to help me advance in life. Thank you for the noble values, education and ongoing support coming from you.

My brothers and sisters who have never ceased to be examples of perseverance, courage and generosity.
To my uncles, aunts, cousins.

To all the friends we were lucky enough to have and all those in the faculty of technology, Of the Department of Electronics of the University of AMMAR Thelidji.

My telecommunication professors who must see in this work the pride of knowledge well acquired.

For All The People I Love.

Thanks and Appreciation:

Praise be to Allah who enlightened me the path of science and knowledge and helped me to perform this duty and agreed to complete this work I extend my thanks and gratitude to all those who helped me from near or far to accomplish this work and to overcome the difficulties I encountered.

I am especially the honorable Doctor Ramdani Saadi who did not spare me the valuable guidance and advice that helped me in completing this research.

I would like to thank all the communications workers and faculty for their invaluable assistance in implementing the internship. I would also like to thank the head of the center who advised me during my training.

Thanks Everyone.

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Abbreviations

Chapter I:

AP : Auxiliary Pilot
ARQ : Automatic Repeat Request
DAB : Digital Audio Broadcasting
EM : Electro magnetic
FES : Forward Error Correction
HF : High Frequency
ICI : Inter-Carrier Interference
LOS : Line Of Sight
LMMSE : Least Minimum Mean Square Error Estimation
LS : Least Square
MMSE : Minimum Mean Square Error Estimation
ML : Maximum-Likelihood
NLOS : None Line Of Sight
SHF : Super High Frequency
UHF : Ultra High Frequency

Chapter II:

AFB : Analysis Filter Bank
CP : Cyclic Prefix
CS : Cyclic Suffix
FBMC : Filter Bank Multi Carrier
FDMA : Frequency Division Multiple Access
FFT : Fast Fourier Transform
FMT : Filtered Multi tone
IFFT : Invers Fast Fourier Transform
ISI : Inter-Symbol Interference
MLSD : Maximum-Likelihood Sequence Detector
OFDM : Orthogonal Frequency Division Multiplexing
OQAM : Offset Quadrature Amplitude Modulation
PAR : Peak to Average Ratio
PSK : Phase-Shift Keying
QAM : Quadrature Amplitude Modulation
RF : Radio Frequency
SFB : Synthesis Filter Bank
ZP : Zero Padding

Chapter III:

AWGN : Additive White Gaussian Noise
BER : Bit Error Rate
BPSK : Binary Phase Shift Keying
PSD : Power Spectral Density
PSK : Phase Shift Keying
QAM : Quadrature Amplitude Modulation
QPSK : Quadrature Phase Shift Keying
SNR : Signal to Noise Ratio

Abstract

Multi-carriers modulation are very seductive techniques for the event of modern wireless communication systems such as the 5th generation (5G). In defiance of their many benefits, (OFDM) system suffers from some drawback.

Due to the abovementioned drawbacks in OFDM systems, the filter bank multicarrier with an offset quadrature amplitude modulation (FBMC/OQAM) system has recently drawn increasing attention from many researchers. FBMC/OQAM well utilizes time frequency localization (TFL).

Regardless of the higher complexity compared to OFDM, FBMC/OQAM can provide remarkably reduced out of band emissions, robustness against carrier frequency offset, and better spectral efficiency as CP is not required.

The purpose of our work is to compare the proposed 5G modulation techniques FBMC against OFDM, which is the modulation technique, used in 4G communications.

In this work, we compare Spectral Density, Spectral Efficiency, BER of FBMC and OFDM modulation techniques to analyze the merits of them.

The simulation results show that FBMC has the lower BER, greater Spectral Efficiency (if the burst is larger) and better performance of Spectral Density compared to OFDM modulation.

The diverse results were simulated using MATLAB software where we can see that the FBMC can do the job better than OFDM technique in several cases…

Key words: 5G, OFDM, FBMC, OQAM.

ملخص

تعد تقنيات نقل المعلومة التي تعتمد على تعديل الحامل المتعدد (MCM) من بين أحسن التقنيات المعتمدة في أنظمة الاتصالات، ومن بينها تقنية (OFDM) المستعملة على نطاق واسع، وعلى الرغم من المزايا المتعددة التي تتميز بها على الأنظمة السابقة إلا أنها تعاني من بعض العيوب مثل خسائر الكفاءة الطيفية إذا قورنت بالنتائج النظرية وذلك بسبب البادئات الدورية (CP)، وعليه فقد ظهرت تقنية أخرى وهي بنك التصفية متعدد الناقلات (FBMC) من أجل تحسين عيوب (OFDM) والتي تعتبر من أنجع الحلول.

والهدف من هذا العمل هو مقارنة تقنيات نقل المعلومة المقترحة للجيل الخامس (FBMC) و (OFDM) مع العلم أن هذه الأخيرة هي أيضا مستعملة في الجيل الرابع.

في هذا البحث سنعتمد على مقارنة الكثافة الطيفية والكفاءة الطيفية و(BER) بين تقنية (OFDM) و(FBMC) لتحليل مزاياهما.

ولقد أظهرت لنا نتائج المحاكاة أن تقنية (FBMC) تمتلك معدل (BER) أقل من (BER) في تقنية (OFDM) وكفاءة طيفية أكبر في حالة الاندفاع الأكبر وأداء أفضل للكثافة الطيفية مقارنة بتشكيل (OFDM).

وتمت محاكاة النتائج المختلفة باستخدام البرنامج (Matlab).

Résumé

Parmi les meilleures technologies adoptées dans les systèmes de communication est la technologie OFDM qui est largement utilisée, et malgré les multiples avantages qu'elle présente par rapport aux systèmes précédents, elle contient certains inconvénients. Telles que les pertes d'efficacité spectrale par rapport aux résultats théoriques dus aux préfixes périodiques (CP), et en conséquence, une autre technique a émergé, qui est le banc de filtres multi-porteuse (FBMC) afin d'améliorer les défauts de (OFDM), qui est l'une des solutions les plus efficaces.

Le but de notre travail est de comparer les techniques de modulation FBMC avec OFDM cette dernière est la technique de modulation, utilisée dans les communications 4G. Sachant que ces 2 techniques sont proposées dans la 5G.

Dans ce travail, nous nous appuierons sur une comparaison de la densité spectrale, de l'efficacité spectrale et (BER) entre la technologie OFDM et le FBMC pour analyser leurs avantages.

Les résultats de la simulation ont montré que le FBMC a un BER plus faible que le BER dans l'OFDM, une plus grande efficacité spectrale dans le cas d'une impulsivité plus grande et de meilleures performances de densité spectrale par rapport à la modulation OFDM.

Les différents résultats sont simulés à l'aide du logiciel (Matlab).

General introduction

In the telecom industry in the mobile industry we have a new generation roughly every 10 years, the first G brought us the very first cell phone, 2G let us text for the first time, 3G brought us online and 4G delivered the speed that we enjoy today.

Now the 5G the new generation of wireless, 5G has a very essential part it is high data rate it deliver peak data rates up to 20 Gbps, and guide for each object. every day several hundred megabits that will be available it will allow to consider many new uses even the most greedy, and above all it will allow us to respond to a capacity challenge.

After it is also a promise of quality of service management we will be able to have what we called network slicing. Then low latency we speak of two milliseconds, which will allow to consider new uses for the industry, and for the connected car, for multimedia and for all other fields of applications, which will require specific service quality.

The 5 G will enhance the IOT due to the difference characteristics as latency and data rate; it will allow us to connect many more objects and connect them better.

Until now the 5G uses different forms of OFDM which is certainly one of the most famous and accepted multicarrier technologies among the mainly wireless communication systems. However, the use of cyclic prefixes and guard bands to avoid too big side lobes, results in losses of spectral efficiency compared to theoretical performance. Due to the abovementioned drawbacks in OFDM systems, the filter bank multicarrier with an offset quadrature amplitude modulation (FBMC/OQAM) system has recently drawn increasing attention from many researchers. FBMC/OQAM well utilizes time frequency localization (TFL).

Regardless of the higher complexity compared to OFDM, FBMC/OQAM can provide remarkably reduced out of band emissions, robustness against carrier frequency offset, and better spectral efficiency as CP is not required.

In this work, we compare FBMC modulations technique against OFDM technique by comparing: Spectral density, Spectral efficiency and response magnitude of FBMC and OFDM modulation techniques to analyze the merits of them.

The memory consists of three chapters as follows: In the first chapter, we will describe a complete presentation on the main concepts relating to digital communication systems and their various characteristics. therafter, we will explain how a digital transmission chain works; finally we will introduce the methods of channel estimation.

In the second chapter, we will introduce single carrier transmission and multi carrier transmission then we will focus on the study of OFDM modulation. We will talk about OFDM modulation, starting with the operating principles and their digital implementation. Then introduced the different FBMC modulation characteristics go through the basics of the multi rate filter bank and then the filter bank modulation, finally the prototype filter implementation.

In the last chapter, we ran simulations on MATLAB on the different blocks constituting the second preceding chapter, a simulation for the OFDM modulation and another for FBMC modulation giving results with different input modulations and also the BER for each modulation.

Chapter I

Radio Link Environment

I. Radio link environment:

I.1. Introduction:

The objective of this chapter is to present the principles and some concepts relating to wave radio propagation and to digital communication systems by a description of digital transmission systems and describe the functioning of a transmission chain from source to receiver. Then, we end with channel estimation and its techniques.

I.2. Wave radio propagation:

I.2.1. Electromagnetic Wave:

- **Definition** :

Electromagnetic waves or EM waves are waves that are created as a result of vibrations between an electric field and a magnetic field. In other words, EM waves are composed of oscillating magnetic and electric fields. An electromagnetic wave can travel through anything - be it air, a solid material or vacuum. It does not need a medium to propagate or travel from one place to another they are measured by their amplitude and wavelength λ of this wave is given by λ = c/f.

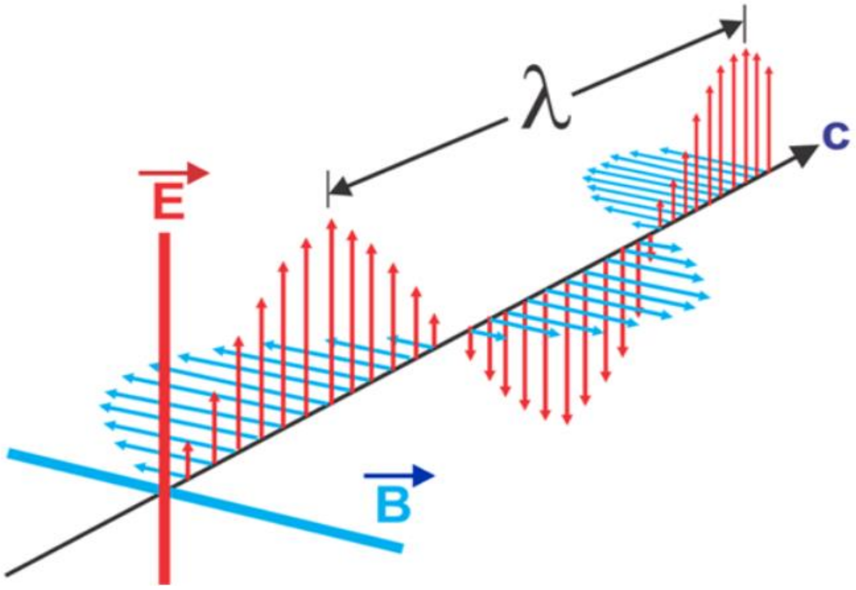


Figure 1 electromagnetic wave

- **Characteristics**

The electromagnetic waves are transverse waves, all waves are characterized by:

- Wavelength, frequency, and amplitude.
  - No need for a carrier medium.

- Examples: light, X-rays and radio waves.

Amplitude:

The amplitude of electromagnetic waves relates to its intensity or brightness (as in the case of visiblelight).

With visible light, the brightness is usually measured in lumens. With other wavelengths the intensity of the radiation, which is power per unit area or watts per square meter is used. The square of the amplitude of a wave is the intensity.

Wavelength:

The wavelengths of electromagnetic waves go from incredibly long to exceptionally short and everything within the center. The wavelengths determine how matter responds to the electromagnetic wave, and those characteristics determine the name we give to that particular group of wavelengths.

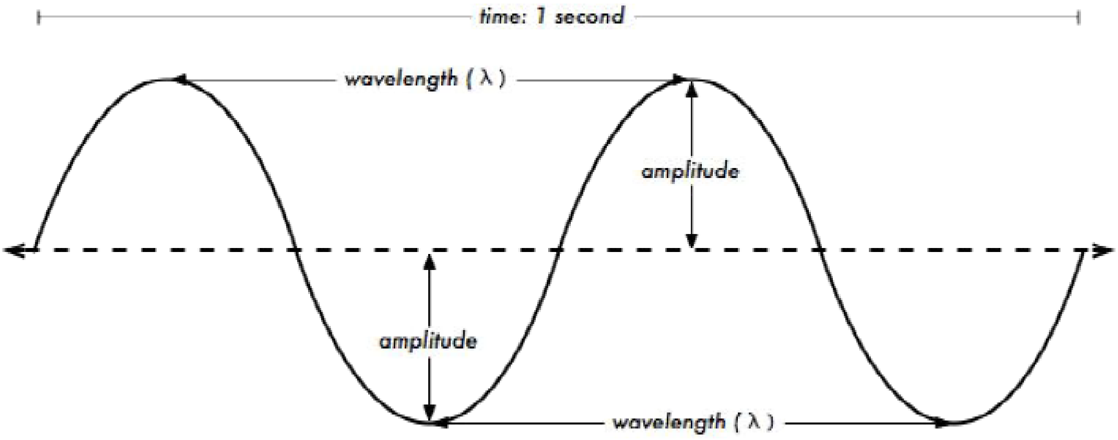


Figure2 amplitude of wave

➢ Behavior of radio waves :

There are few simple rules of basic dependable guidelines that can be incredibly helpful when making first plans for a wireless network:

The **longer** the wavelength, the further it goes.
The **longer** the wavelength, the better it travels through and around things.
The **shorter** the wavelength, the more data it can transmit. [1]

➢ Properties :

In Wireless communication, radio propagation alludes to the conduct of radio waves when they are propagated from transmitter to receiver. During the propagation, radio waves are fundamentally influenced by several different modes of physical phenomena: [1] [2]

- Absorption
- Reflection
- Diffraction
- Refraction
- Scattering

- **Absorption :**

When electromagnetic waves propagated through some material, they generally get weakened or dampened, Materials that absorb energy include:

- **Metal** electrons can move freely in metals, and they are readily able to swing and thus absorb the energy of a passing wave.
- **Water** molecules jostle around in the presence of radio waves, while absorbing some energy.
- **Trees** and **Wood** absorb radio energy proportionally to the amount of water contained in them.
- **Humans** are mostly water, we absorb radio energy. [1]

- **Reflection :**

Reflection is the physical phenomenon that happens when a propagating electromagnetic wave encroaches upon an object with very large dimensions compared to the wavelength. The surface of the earth or a building for example. It forces the transmit signal power to be reflected back to its origin rather than being passed along the path to the receiver[2].

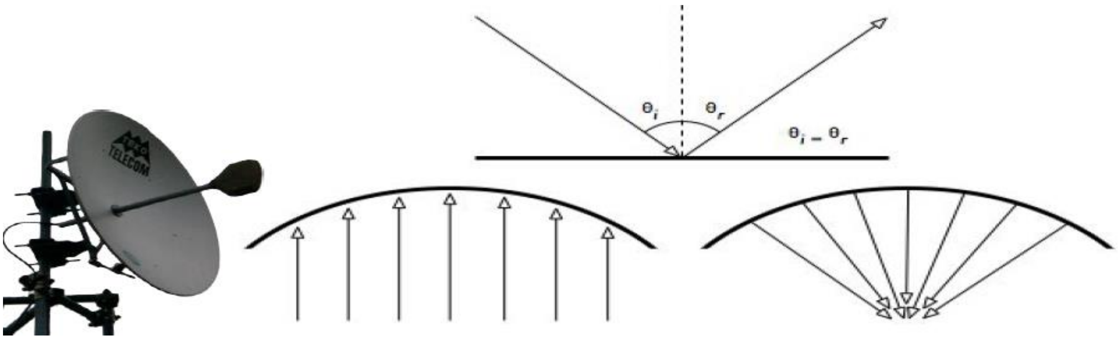


Figure 3 Reflection of wave [1]

The rules for reflection are quite simple: the angle at which a wave hits a surface is the same angle at which it gets deflected.

and **Water** are excellent reflectors of radio waves. [1]

- **Diffraction :**

Diffraction refers to different processes that arise when a surface with sharp irregularities or minor gaps obstructs the radio course between both the transmitter and the receiver. It shows up as a twisting of waves around the little obstacles and expanding through tiny gaps[2].

The secondary waves produced by diffraction are useful for setting a path between the transmitter and the receiver, even when there is no line-of-sight direction. Due to the influence of diffraction waves, it can "bend" around corners, or through a barrier gap.[1]

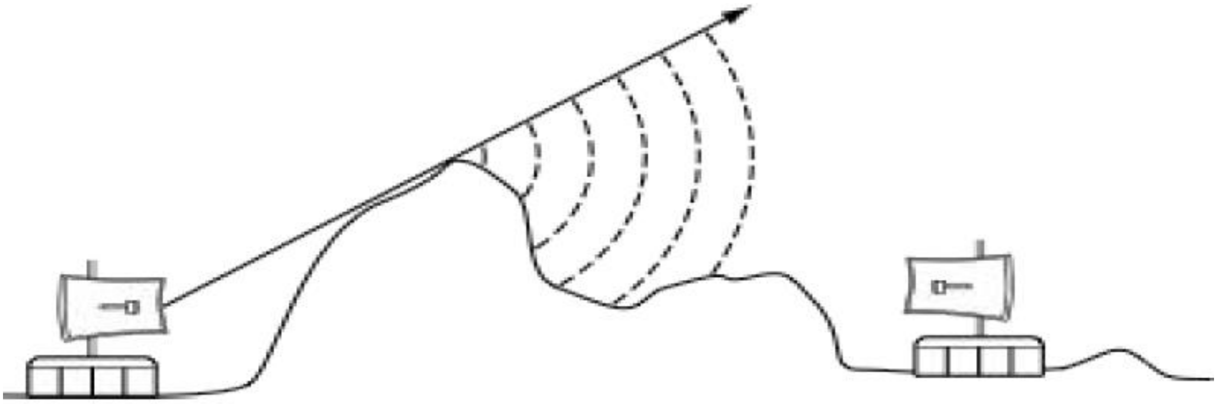


Figure 4 diffraction of wave [2]

- **Refraction**

Refraction is the visible "bending" of waves as they come across various characteristics of a substance. When a wave travels from one medium to another, velocity and direction change when reaching the new medium. [2]

- **Scattering**

Scattering is the scientific principle that obliges the radiation of an electromagnetic wave to detract by one or more nearby obstacles from a straight line, with small dimensions relative to the wavelength. Those obstacles that induce scattering, such as foliage, street signs, and lamp posts, are referred to as the scatters. In other words, the propagation of a radio wave is a complex, less straightforward mechanism controlled by reflection, diffraction, and scattering, whose intensity at different instances differs with different environments.[2]

- **Other important wave properties :**

These properties are also important to consider when using electromagnetic waves for communications:

- Phase
- Polarization
- Fresnel Zone [1]
- **Phase :**

The phase of a wave is a fraction of a cycle that the wave is offset from a reference point. It is a relative measurement which can represent itself through various ways (radians, cycles, degrees, percentage).

Two waves that share similar frequencies and different phases have a **phase difference**, and the waves are considered out of phase with one another.[1]

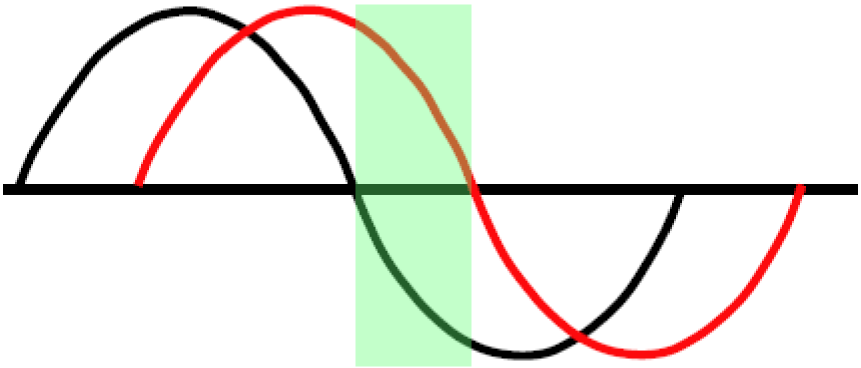


Figure 5 phase of wave [1]

When two waves of the same frequency amplitude and phase meet the result is **constructive interference**, the amplitude doubles.

When two waves of the same frequency and amplitude and **opposite phase** meet the result is destructive interference.[1]

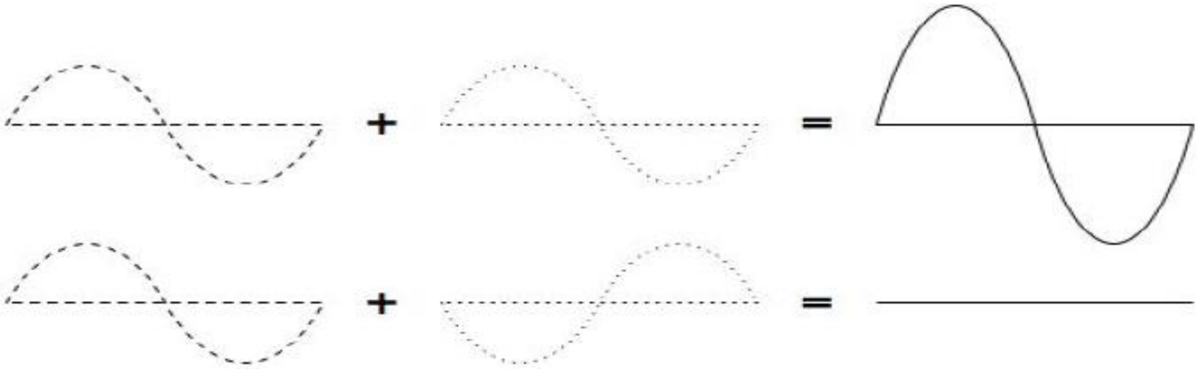


Figure 6 constructive interference and opposite phase[1]

- **Fresnel zone :**

The Fresnel zone is made up of many zones, zone 1 has the strongest signal and following zones (Zone 2, and Zone 3) have weaker signals.

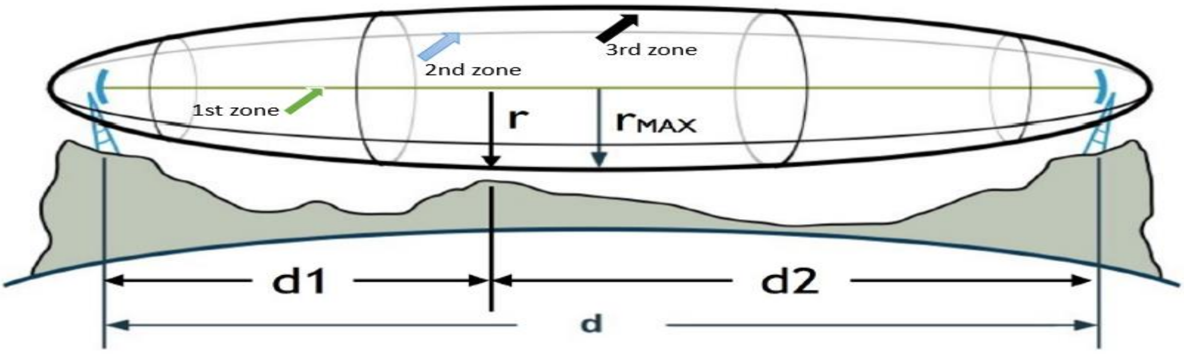


Figure 7 Fresnel zone

$$r = sqrt\left(\partial \times d1 \times \frac{d2}{d}\right) \qquad (I.1)$$

$$r\ max = \frac{1}{2} \times sqrt\ (\partial \times d) \qquad (I.2)$$

Where all the dimensions are in meter

I.2.2. Power attenuation:

Attenuation is the drop in the signal power when transmitting from one point to another. The power of the received signal generally decreases on average as a function of the distance ‘d’ traveled by electromagnetic wave with a given attenuation in the form d_n where n is a positive real which depends on the link between the base station and the mobile. When the base station and the mobile are in direct view (« Line Of Sight LOS», the average signal power decreases in power by 2 as a function of the distance (n = 2). Power attenuation :

$$Att(d) = (2\pi d / \partial)^2 \quad \text{(I.3)}$$

Where ∂ is the wavelength of the transmitted signal.

Generally it caused by path length, obstructions in the signal path, and multipath effects Figure 8 shows some of the radio propagation effects that cause attenuation.

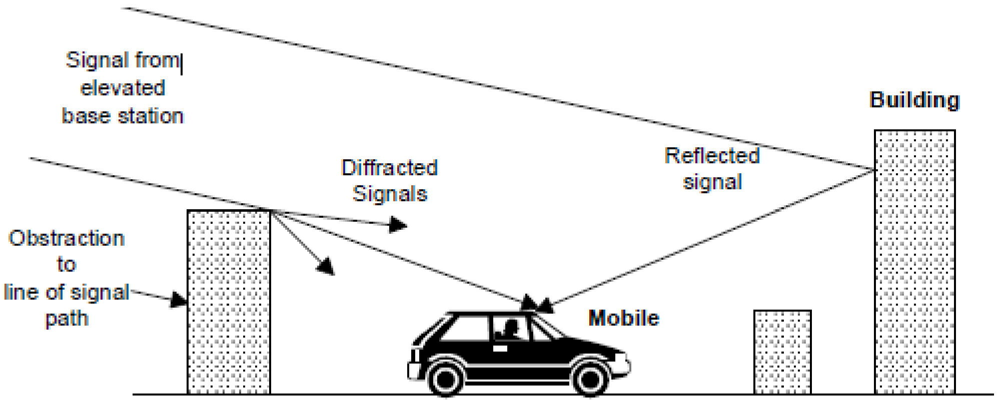


Figure 8 Radio propagation effects [3]

I.2.2.1. Shadowing:

It is generally caused by obstruction of waves by obstacles (buildings, forests, hills, etc.) which results in more or less pronounced attenuation. Unlike fluctuations due to multipath, this type of fluctuation is called long-term fluctuation compared to wavelength. Many studies model this fading as a random variable of log-normal distribution which brings a certain uncertainty to the attenuation.

Typical amounts of variation in attenuation due to shadowing are shown in Table 1 Shadowed areas tend to be large, resulting in the rate of change of the signal

Description	**Typical Attenuation due to shadowing**
Heavily built-up urban center	20dB variation from street to street
Sub-urban area (fewer large buildings	10dB greater signal power than built-up urban center
Open rural area	20dB greater signal power than sub-urban areas
Terrain irregularities and tree foliage	3-12dB signal power variation

Table 1: Typical Attenuation in a radio channel (values from [3])

I.2.2.2. Fading phenomenon:

➢ Definition:

One of the characteristics of a wireless channel is a phenomenon called fading, it is the variation of the amplitude of a signal in time and frequency, fading is the source of signal non-additive signal degradation in the wireless channel. It can be caused by multipath propagation or by shadowing. The fading phenomenon in the wireless communication channel was initially modeled for HF (High Frequency, 3_30MHz), UHF (Ultra HF, 300_3000 GHz), and SHF (Super HF, 3_30 GHz) bands in the 1950s and 1960s. Currently, the most popular wireless channel models have been established for 800MHz to 2.5 GHz by extensive channel measurements in the field.[4]

➢ Classification of fading channel:

Fading can be classified into 2 types: large scale fading and small scale fading as shown in figure

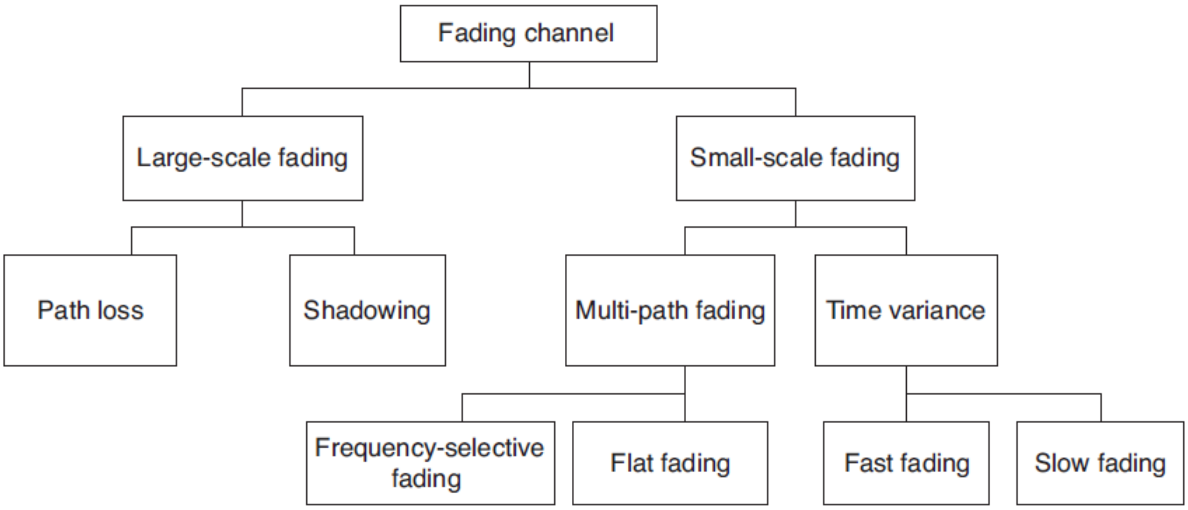


Figure 9 classification of fading channel

Large-scale fading:

occurs when the mobile move over a very long distance (several hundreds or thousands of meters) order of cell size or maybe more, it represented the average signal power attenuation or path loss of signal as a function of distance and shadowing in general path loss model. We use the propagation model for predicting the received signal where there is no obstacle between the transmitter and the receiver. It is also adopted for the satellite communication systems. By using nonisotropic antennas with a transmitting gain of Gt and a receiving gain of Gr, the power obtained at distance d, Pr(d), is represented by the well-known Friis equation [4], given as**:**

$$\Pr(d) = \frac{PtGtGr\partial^2}{(4\pi)^2\, d^2 L} \qquad (I.4)$$

Where Pt specifies the transmit power (watts), l is the radiation wavelength (m), and L is the system loss factor failure which is independent of the propagation environment. The factor of system loss reflects total attenuation or failure in the actual hardware including transmission line, filter and antennas. In general, L > 1, but L 1⁄4 1 if we presume that device hardware does not compromise. From Equation (1.1) it is clear that the power obtained attenuates exponentially with the distance d. The free-space path loss, PLF(d), can be generated directly from Equation (I.4) without any device loss with L ¼ 1 as[4]

$$PLf\ (d)[dB] = 10\log\left(\frac{Pt}{Pr}\right) = -10\log\left(\frac{GtGr\partial^2}{(4\pi)^2 d^2}\right) \quad \text{(I.5)}$$

Without antenna gains (e.i. Gt =Gr=1) equation (I.5) is reduced to:

$$PLf\ (d)[dB] = 10\log\left(\frac{Pt}{Pr}\right) = 20\log\left(\frac{4\pi d}{\sigma}\right) \quad \text{(I.6)}$$

Small-scale fading

Represent the rapid changes of the amplitude and phase of a radio signal when the mobile station moves a short distance or for a short period of time , it’s due to the effect of multiple signal paths. Depending on the relative extent of a multi-path, frequency selectivity of a channel is characterized (e.g., by frequency-selective or frequency flat) for small-scaling fading and also it’s due to the mobile speed depending on time variation in it in other words small-scale fading is attributed to multi-path propagation, mobile speed, speed of surrounding objects, and transmission bandwidth of signal. [4]

➢ **Types of fading** :

- **Frequency selective fading**: is when different parts of the transmitted signal spectrum are attenuated differently due to the multipath effects, it provokes a cancellation of some frequencies in the receptor because of reflections.
- **Frequency Non-Selective fading:** when all the frequencies to the signal undergo kind of the same degree of fading this kind of channel are classified as frequency non- selective.
- **Slow fading:** Slow fading is a long-term, vanishing effect that changes the average value of the signal received. Usually slow fading is strongly correlated with shifting away from the transmitter and experiencing the estimated decrease in signal strength. Slow fading could be caused by situations like shadow, in which a large obstacle such as a mountain or a huge building obstructs the main signal path between both the transmitter and the receiver.
- **Fast fading:** Fast fading is the component associated with multi-path propagation in the short term. It's impacted by the mobile terminal velocity and signal transmission bandwidth.

I.2.2.3. Multipath channel:

➢ Definition :

Means that there are many paths, one direct path and many reflected paths and generally the reflected paths are more long than direct path means they arrive later than the direct path, so at the receiver all signals arrived with deferent delay

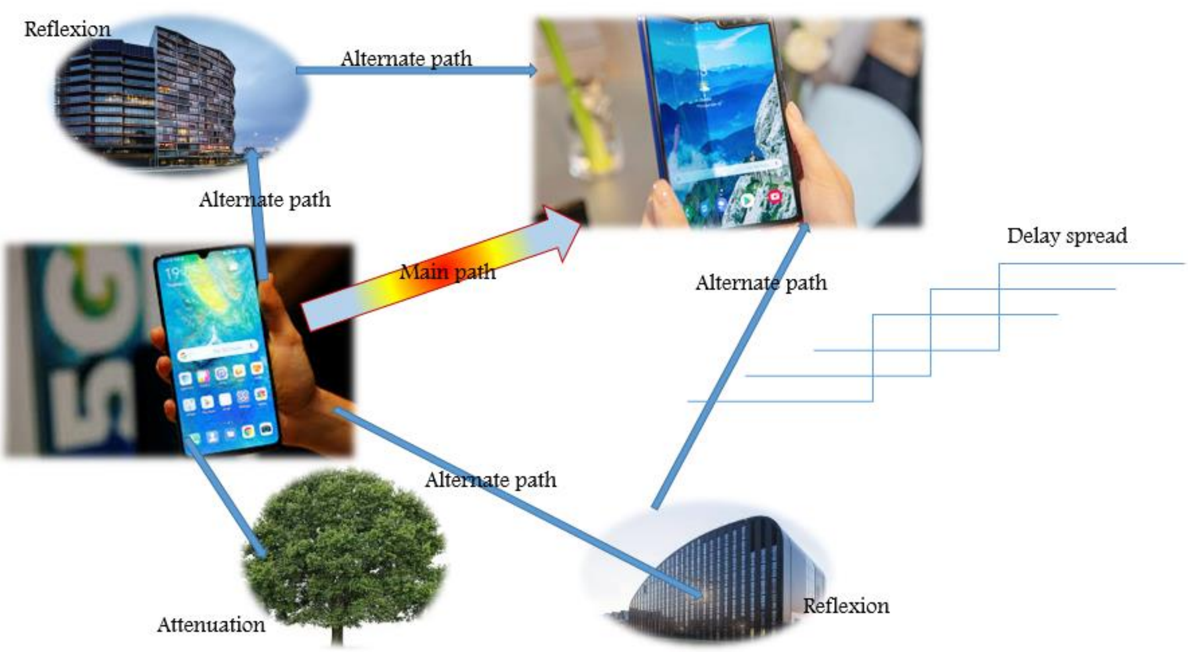


Figure 10 Multipath channel

Generally we have two situation of propagation:

- **NLOS (None Line Of Sight)**
  When there is no direct visibility between transmitter and receiver in this case the density of probability of amplitude follows the distribution of Rayleigh. [5]

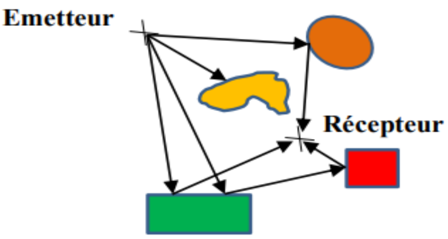


Figure 11 NLOS

- **LOS (Line Of Sight):**

When there is direct visibility between transmitter and receiver in this case (there is a direct path) the density of probability of amplitude follows the distribution of Rice.[5]

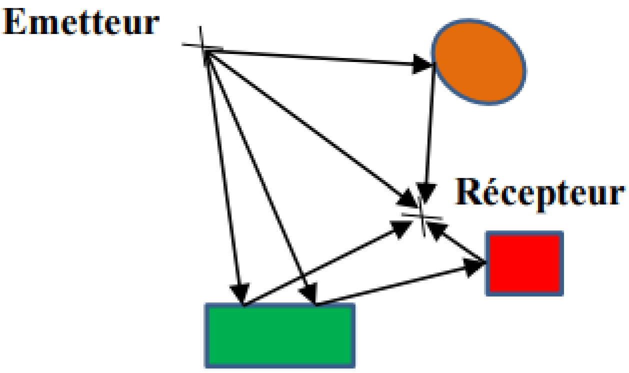


Figure 12 LOS

The multipath propagation problem is usually due to the reflections and refractions that occur during wireless transmission. Multipath fading affects most forms of radio communications links in one form or another which decreases the quality of the communication, causing interference and call drop, which has side effects as a waste of spectrum resources and a large side lobe.

➢ Characteristic of multipath channel

Let us consider a simple model of the channel at the end of which the receiver receives the sum of the transmitted signal and of the signals having undergone echoes therefore delayed by amplitude hi the impulse response is written:

$$h(t) = \sum_i hi\delta(t-\tau i) \qquad \text{(I.7)}$$

It transfer function is :

$$H(f) = \sum_i hie^{-2j\pi f\tau i} \qquad \text{(I.8)}$$

We take an easy exemple of two traject the direct is delay with τi

$$h(t) = 1 + \partial\delta(t-\tau) \quad \text{and} \quad H(f^2) = 1 + \partial^2 + 2\partial\cos 2\pi f\tau \qquad \text{(I.9)}$$

The transfer function looks like this:

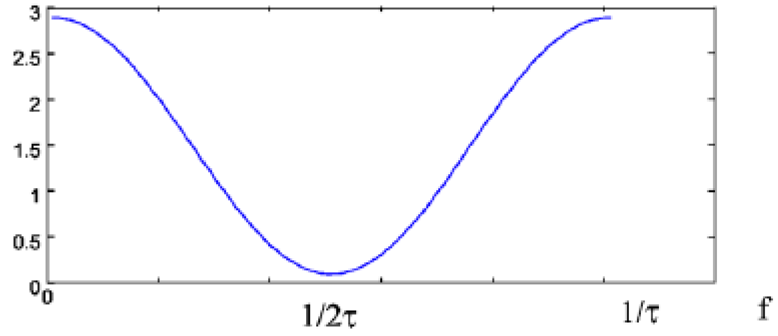


Figure 13.transfer function of channel with a delayed

We remarque that the transfer function with some areas whene the signal is amplified (H(f)>1and some area when the signal is so weakened (H(f) <1) area of fading. It period of variation is around $\frac{1}{\tau}$, τ is the spread of delays.

Depending on the value of the band occupied B by the signal, two cases may arise:

- B<<1/ tau , H(f) we can considerd constant in the bandwith B don't udergo a distorsion, but it can be so weakened if the frequence of modulation is near of 1/2tau (the signals from direct path and path delayed are opposition of phase)
  But it can be also amplified
- B>>1/tau , H(f) it isn't constant sur in the band of frequence and the signal don't undergo distorsion that can be corrected with an equalizer [6]

Propagation model

The propagation model is the main tool, its daily use is designing planning and analyzing wireless communication networks.

It is important to point out that there is no general method or algorithm that is universally accepted as the best propagation model, each model can be useful for some specific environment and the accuracy of any particular technique of algorithm depends on the fit between the parameters available for the area concerned, and the parameters required by the model, here we (are interested) in proparagation model of the multipath channel.

We suppose that the transmitter signal can be written:

$$s(t) = Re[S1(t)e^{2j\pi fct}] \qquad (I.10)$$

fc frequence of carrier.

the receiver signal is sum of signals from deferent paths:

$$\boldsymbol{x(t) = \sum \partial n(t) s(t - \tau n(t))} \qquad \text{(I.11)}$$

Each path causes an attenuation ∂n and delay τn which varies in function of time

The signal x(t) can also be written :

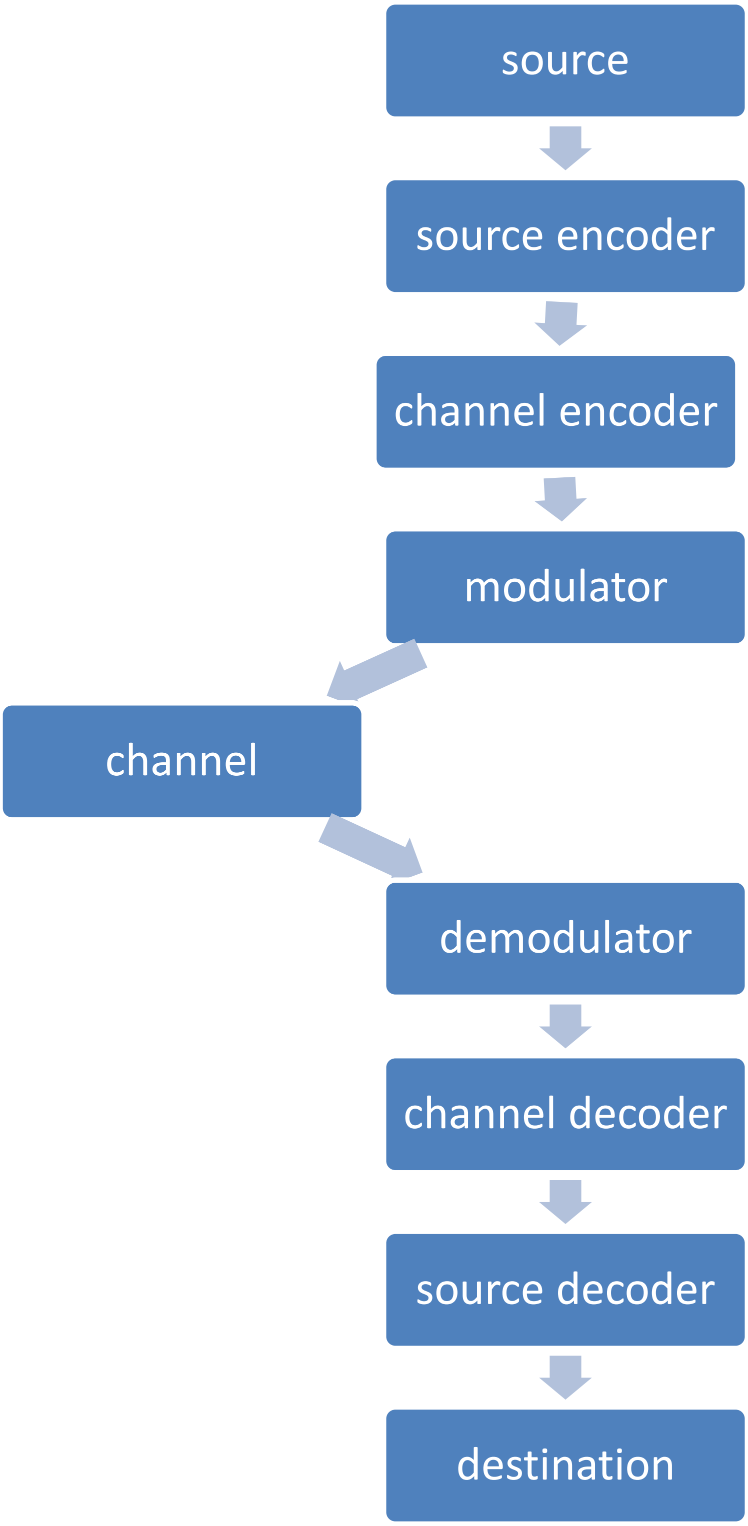
$$\boldsymbol{x(t) = Re\left\{\sum[\partial n(t) e^{-2j\pi f c\tau n(t)} S1(t - \tau n(t))] e^{2j\pi f ct}\right\}} \qquad \text{(I.12)}$$

The expression in square brackets shows that the baseband signal is attenuated.

∂n and delays τn and phase shifts$2j\pi f c\tau n$ that all depend in time from this formula can define the baseband transfer function:

$$\boldsymbol{c(t,\tau) = \sum \alpha n\ (t) e^{-2j\pi f c\tau n(t)}\ \delta(t - \tau n(t))} \qquad \text{(I.13)} \qquad [7]$$

II. Radio channel chain :



II.1. Description of radio channel chain:

- The Sender: Has the objective of adapting the information of the source before transmitting through a propagation channel. Indeed, from the received signal, the receiver makes the operation to retrieve information from the source.

- Source Encoder: is designed to effectively specify the digital or analog data by bits. To increase the required transmission rate and optimize usage system resources, the sequence transmitted by the source must be as short as possible. The source encoder is intended to compress information by eliminating bits of insignificance. Shannon published the principle of source coding.

- Channel Coding: When passing through the transmission channel, the signal is subject to various interferences, introducing reception errors. To increase transmission reliability, channel coding introduces redundancy into the information sequence. The receiver knows the coding law used and is, therefore, able to detect and correct erroneous binary data.

- Digital modulation: Its main purpose is to modulate or transform the binary signals in signal waveforms (amplitude and phase) because the actual channels allow only the transmission of electrical signals of this type [4]. The sequence binary information goes through a digital modulator which acts as an interface with the transmission channel by giving the signal a physical envelope. Each element or group of binary elements is associated with a waveform according to a modulation law, this waveform is generated by the converter binary. Each waveform associated with a group of bits is called a "symbol", everything then forming a signal capable of being sent into the channel by a frequency carrier [3].

- Communication channel: A communication channel is a physical environment that is used to transmit a signal from a transmitter to the receiver and during transmission noise effect is considered a random disturbance which comes from outside and inside the receiver [2].

- Digital demodulation: Demodulation makes it possible to recover information sent by the transmitter. This is the basic function of the receiver. Its location obviously depends on the modulation used.

- Channel decoding: Channel decoding consists first of all in detecting the presence of errors in the information and then in a second step correct. From these two actions arise three main strategies: ARQ (Automatic Repeat Request) strategies which are limited to detecting the presence of any errors, the correction is made by retransmission of erroneous blocks, the FES (Forward Error Correction) strategies implementing the codes allowing the detection and correction of errors without any retransmission. Finally, hybrid systems combine between the two techniques.

- Source decoding: Source decoding consists in reconstructing, by the application of the source decoding algorithm "decompression for example", the information original substitution sequence

- The receiver: Has the objective of reconstructing as best as possible, the message sent from the signal received. It includes amplification circuits for frequency change of demodulation and sampling. Finally, a decision device: identifies the value binary symbols transmitted [5].

III. Channel estimation:

III.1. Definition:

Any transmission is affected in one way or another by the transmission channel, the estimation of the channel is an essential task in a coherent detection process on reception before any transmission of information [8]. Any error in channel estimation suggests a higher detection error rate. Depending on the nature of the channel and the system, an adequate method is required for greater precision in the channel estimation and that based on maximum likelihood:

$$\boldsymbol{\hat{h}} = \mathbf{arg}\,\boldsymbol{h}\,\mathbf{min}\|\boldsymbol{y} - \boldsymbol{Sh}\| \quad \text{(I.14)}$$

Where y, S and H are the matrix expressions of the received signal, the training sequence and the channel respectively. [9]

This method is complex and requires a long computation time. For this reason, suboptimal methods are used. In this section we will discuss suboptimal techniques for channel estimation.

Channel estimation for multiple modulations depends on the demodulation processes implemented, two main techniques are used differential and coherent. [9]Differential demodulation: when using differential demodulation there is no need for a channel estimate [8] supposes a quasi-invariance of the channel over a period of 2 consecutive symbols this method is based on the coding of the transition from one symbol to another according to the time and frequency axes. It therefore does not require the estimation of the channel at any point; however, it only applies to phase modulations. [10] This is a common technique in wireless communication system, which, since no channel estimates is needed, reduces the complexity of the receiver. Differential modulation is used in European DAB standard [11].

The drawbacks are about a 3 dB noise enhancement [12], and inability to use efficient multi amplitude constellations. An interesting alternative of DPSK is differential amplitude phase shift keying [13], where a spectral efficiency greater than DPSK is achieved by using a differential coding of amplitude as well.

Obviously, this requires a non-uniform amplitude distribution. However, in wired systems, where channel is not changing with time, coherent modulation is an obvious choice. But, in wireless systems, the efficiency of coherent modulation makes it an ideal choice when the bit error rate is high, such as in DVB [4].

- Differential demodulation, supposes a quasi-invariance of the channel over a duration of two consecutive symbols. This method is based on the coding of the transition from one symbol to another according to the time and frequency axes. It therefore does not require channel estimation at any point; however, it only applies to phase modulations.

- Coherent demodulation: is not limited to the chosen modulation. However, it requires the estimation of the frequency response of the channel on all the subcarriers. It is based on the interpolation of the channel coefficients at the defined positions of pilot carriers; the possibility of adapting the representation pattern of these pilot carriers in the time-frequency domain makes it an interesting solution with respect to mobile radio channels.

In the case of coherent demodulation, several schemes for inserting pilot carriers can be considered depending on the characteristics of the propagation channel.

There are mainly two problems in the design of channel estimators for the wireless systems. The first problem is concerned with the choice of how the pilot information should be transmitted. Pilot symbols along with the data symbols can be transmitted in a number of ways, and different patterns yields different performances [8]. The second problem is the design of an interpolation filter with both low complexity and good performance. These two problems are interconnected, since the performance of the interpolator depends on how pilot information is transmitted.

The estimation channel is mostly done by inserting pilot symbols into all of the subcarriers. Channel estimates are often achieved by multiplexing known symbols [14], so we can say that there's many methods of channel estimation.

III.2. Methods of channel estimation:

1. Comb type :

The comb type based channel estimation in this type pilot symbols are transmitted on some of the subcarriers of each symbol. This method usually uses different interpolation schemes such as linear, low-pass, spline cubic, and interpolation polynomial

This method relies upon the insertion of known phasors into the stream of useful information symbols for the purpose of channel sounding, these pilot symbols allow the receiver to extract channel attenuations and phase rotation estimates for each received symbol.

It consists on every symbol has pilot tones at the periodically located subcarriers which are used for a frequency domain interpolation.

Let Tf be the period of pilot tones in frequency, the pilot symbols must be placed as coherent bandwidth is determined by an inverse of the maximum delay spread $\partial\tau$ max the pilot symbol period must satisfy the following inequality:

$$\boldsymbol{Tf \leq 1/\sigma\tau max} \qquad \text{(I.15)}$$

An example of this shown if figure... Which shows both scattered and continual pilot symbols. [14]

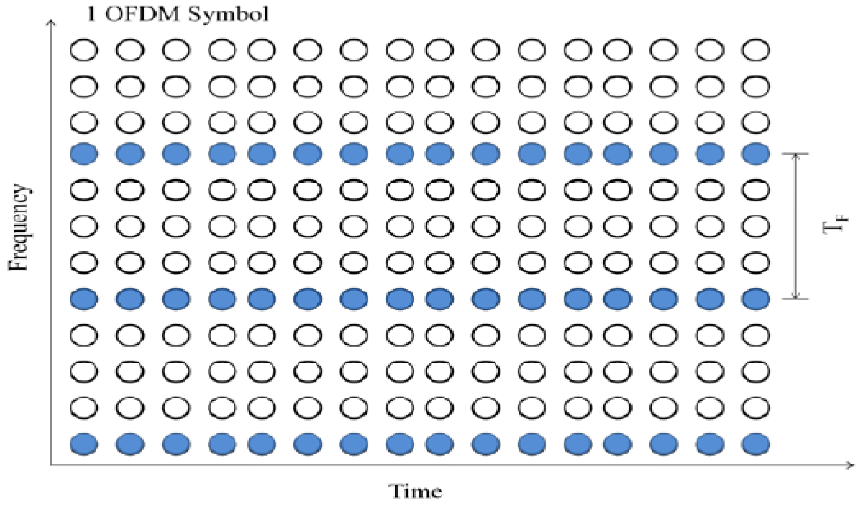


Figure 14 Comb type pilot arrangement.

2. Lattice type :

In this method we insert pilot tones along both of time and frequency axes with given periods, the pilot tones scatted in both time and frequency axes facilitate time/frequency domain interpolations for channel estimation.

Consider **Tp** and **Tf** are periods of pilot symbols in time and frequency respectively. The pilot symbol arrangement must satisfy both of the following Equation:

TP=St and TF = Sf (I.16)

$$\boldsymbol{Tp \le \frac{1}{fDoppler}} \quad \text{(I.17)} \qquad \textbf{and} \qquad \boldsymbol{Tf \le 1/\sigma\tau max} \ \text{(I.18)}$$

Where Fdoppler is Doppler spreading and max is maximum delay spread.

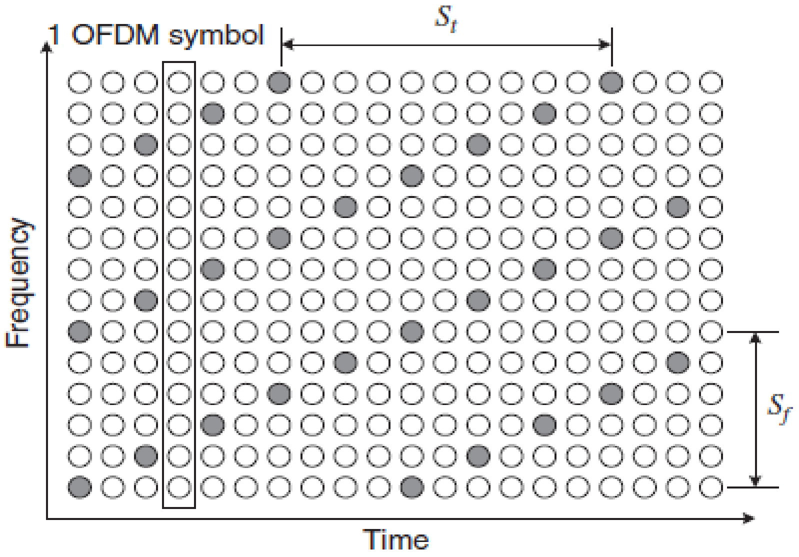


Figure 15 Lattice type pilot arrangement [4]

3. Training symbol based channel estimation :

In channel estimation we can use a training symbol, cause they provide a good performance, however due to the required of training symbols such as preamble or pilot tones that are transmitted in addition to the data symbols, their transmission efficiencies reduced.

In training symbol the least square and minimum square error techniques are widely used for channel estimation.

We can represent training symbols with N subcarrier by the diagonal matrix X, considering that all subcarriers are orthogonal.

$$\boldsymbol{X} = \begin{bmatrix} X[0] & 0 & & \cdots & 0 \\ 0 & X[1] & & & \vdots \\ \vdots & & & \ddots & 0 \\ 0 & \cdots & & 0 & X[N-1] \end{bmatrix}$$

Considering that X[k] denotes a pilot tone at the K-th subcarrier, H[k] denotes the channel gain for each subcarrier k, W is a noise vector and the received training signal Y[k] can be represented as:

$$\boldsymbol{Y} \equiv \begin{bmatrix} Y[0] \\ Y[1] \\ \vdots \\ Y[N-1] \end{bmatrix} = \begin{bmatrix} X[0] & 0 & & \cdots & 0 \\ 0 & X[1] & & & \vdots \\ \vdots & & & \ddots & 0 \\ 0 & \cdots & & 0 & X[N-1] \end{bmatrix} \begin{bmatrix} H[0] \\ H[1] \\ \vdots \\ H[N-1] \end{bmatrix} + \begin{bmatrix} W[0] \\ W[1] \\ \vdots \\ W[N-1] \end{bmatrix}$$

$$\mathbf{Y = XH + W}$$

Let $\widehat{H}$ the estimate of channel H.

The following figure represents the general structure of channel estimator.

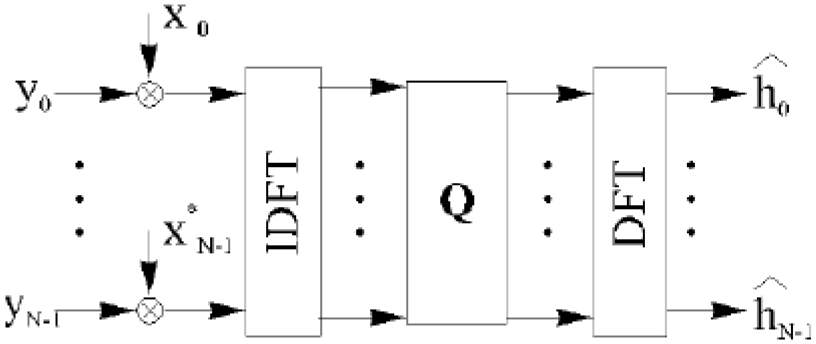


Figure 16 General estimation structure [4]

The input to the estimator is the vector Y considered as received vector. X are pilot carriers placed as shaded symbols, these pilots are available at receiver and on base of received Y and X, channel is estimated. [4]

3.1. (LS) Least Square Estimation :

The idea behind LS channel estimation method is to find channel estimate $\widetilde{H}$ in such a way that weighted errors between the measurements and the model are minimized, The LS estimate of the channel given the received data Y and the transmitted symbol X is

$$J(\widetilde{H}) = \|Y - X\widetilde{H}\|^2 \tag{I.19}$$

$$= (Y - X\widetilde{H})^H (Y - X\widetilde{H}) \tag{I.20}$$

$$= Y^H Y - Y^H X\widetilde{H} - Y^H X^H Y + \widetilde{H}^H X^H X\widetilde{H} \tag{I.21}$$

By fixed the derivation of the function compared to $\widetilde{H}$ to zero to the LS channel estimation as:

$$\widetilde{H} = (X^H X^{-1} X^H Y) = X^{-1} Y \tag{I.22}$$

The LS estimate of the channel H is susceptible to Gaussian noise and inter-carrier interference (ICI). Because the channel responses of data subcarriers are obtained by interpolating the channel responses of pilot subcarriers [14].

3.2. (MMSE) Minimum Mean-Square Error Estimation

In this method we use a new matrix which is weight matrix ω we define:

$$\widetilde{H} \equiv \omega \widetilde{H} \tag{I.23}$$

corresponding to the MMSE estimate, the MMSEchannel estimation method finds a better (linear) estimate in terms of ω.

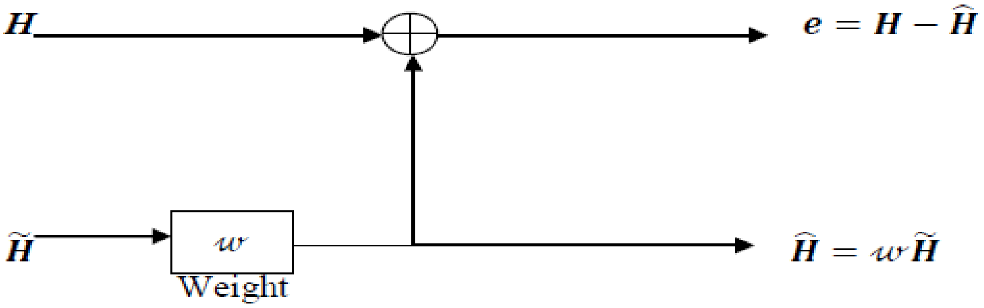


Figure 17 MMSE channel estimation [4]

The principle of orthogonality states that the estimation error $e = H - \widehat{H}$ vector is orthogonal to $\widetilde{H}$such that:

$$E\{e\widetilde{H}^H\} = E\{(H - \widehat{H})\widetilde{H}^H\} \tag{I.24}$$

$$= E\{(H - \omega\widetilde{H})\widetilde{H}^H\} \tag{I.25}$$

$$= E\{H\,\widetilde{H}^H\} - \omega E\{\widetilde{H}\widetilde{H}^H\} \tag{I.26}$$

$$= R\,H\widehat{H} - \omega R\,H\widehat{H} = 0 \tag{I.27}$$

R AB denote the cross correlation matrix of NxN matrices A and B

$$(\boldsymbol{i.e.}, \boldsymbol{R\ AB}\ = \boldsymbol{EAB^{H}}\ ) \qquad \text{(I.28)}$$

and $\widehat{H}$ is the LS channel estimation given as:

$$\widetilde{\boldsymbol{H}}\ =\ \boldsymbol{X^{-1}\,Y}\ = \boldsymbol{H} +\ \boldsymbol{X^{-1}\,W} \qquad \text{(I.29)}$$

When we solve equation, we find w.

$$\boldsymbol{\omega} =\ \boldsymbol{R_{H\widetilde{H}}\,R^{-1}_{\widetilde{H}\widetilde{H}}} \qquad \textbf{(}\text{I.30}\textbf{)}$$

Where $R_{H\widetilde{H}}$ is the autocorrelation matrix of $\widetilde{H}$ and can be derived as:

$$\boldsymbol{R_{H\,\widetilde{H}}}\ = \boldsymbol{E\{HH^{H}\}} +\ \boldsymbol{\sigma^{2}\,\frac{W}{\sigma^{2}X}\ \ I} \qquad \text{(I.31)}$$

And $R_{H\,\widetilde{H}}$ is the cross correlation matrix between the true channel vector and temporary channel estimate vector in time domain.[4]

3.3. LMMSE :

The LMMSE estimate has been shown to be better than the LS estimate; it aims to minimize the MMSE estimate of the channel response given as:

$$\boldsymbol{H^{LMMSE}} = \boldsymbol{R_{H\,\widetilde{H}}}(\ \boldsymbol{R_{HH}} +\ \boldsymbol{\sigma\mu^{2}}((\boldsymbol{XX^{H}}))^{-1})\widetilde{\boldsymbol{H}} \qquad \text{(I.32)}$$

H^{LMMSE} denote the LMMSE channel estimate and $\sigma\mu^{2}$ is the noise power.

The problem behind the LMMSE estimator is to:

$$(\boldsymbol{XX^{H}})^{-1} = \boldsymbol{E}\ [\ (\boldsymbol{XX^{H}})^{-1}] \qquad \text{(I.33)}$$

$$\boldsymbol{H^{LMMSE}} = \boldsymbol{R_{H\,\widetilde{H}}}\left(\ \boldsymbol{R_{HH}} + \frac{\boldsymbol{\beta}}{\boldsymbol{SNR}}\boldsymbol{IP}\right)^{-1}\ \widetilde{\boldsymbol{H}} \qquad \text{(I.34)}$$

β is the scaling factor depending upon the constellation. [14]

4. Two dimensional Estimation :

Two-dimensional estimation methods exist, we will present a method by way of example, and this method has the particularity of combining two methods of one-dimensional channel estimation, hence the term two-dimensional.

The two-dimensional approach which seems to us the simplest is that which combines channel estimation both in the time domain and in the frequency domain as follows:

S denote the cyclic prefix and η indicates the order of the OFDM symbol.

$$\boldsymbol{y} = \boldsymbol{Sh} + \boldsymbol{\eta} \qquad \text{(I.35)}$$

With:

$$\boldsymbol{S} = \begin{matrix} \boldsymbol{S1} & \boldsymbol{S_L^p} & \boldsymbol{S_1^p} \\ \boldsymbol{S2} & \boldsymbol{S1} & \boldsymbol{S_2^p} \\ \boldsymbol{S_L} & \boldsymbol{S_{L-1}} & \boldsymbol{S_l^p} \end{matrix} \qquad \text{(I.36)}$$

when the upper index p designates the cyclic prefix elements of the preceding block. At the initialization of the transmission, they will be considered null but thereafter, they may be replaced by the estimated value in combinations with the training sequence.

$$\boldsymbol{Y} = (\boldsymbol{X})\boldsymbol{h} + \boldsymbol{\eta} \qquad \text{(I.37)}$$

$$\boldsymbol{Y} = (\boldsymbol{x})\boldsymbol{Q_L}\,\boldsymbol{h} + \boldsymbol{\eta} \qquad \text{(I.38)}$$

Or Y is the recieved signal and h is the channel, Q_L is the matrix reduced to the first L columns of the square matrix Q of size N.

Defined by:

$$\boldsymbol{Q} = \left[\boldsymbol{e}^{\boldsymbol{j\frac{2\pi}{N}lm}}\right] \qquad \boldsymbol{0 \leq l \leq N-1 \text{ and } 0 \leq m \leq N-1} \qquad \text{(I.39)}$$

We can therefore combine equations (1 e) and (2 e) to obtain the impulse response of the channel as follows:

$$\begin{bmatrix} Y \\ \underline{y} \end{bmatrix} = \begin{bmatrix} diag(X)Q_L^* \\ \underline{S} \end{bmatrix} h + \begin{bmatrix} n \\ \underline{\eta} \end{bmatrix} \qquad \text{(I.40)} \quad [14]$$

5. Auxiliary Pilot

In recent years, many researchers have studied pilot based channel estimation schemes, and they proposed a new methods such as: Auxiliary pilot AP.

Auxiliary pilot AP : in this method we insert pilots in the interval of auxiliary pilots of comb pilots, [15] we use this method to reduce the interference on pilot symbols

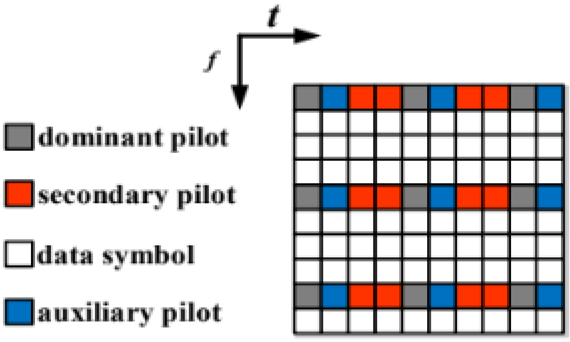


Figure 18 a comb pilot based on auxiliary pilot [15]

Figure [19] show as that, the pilot with the auxiliary pilot is called the dominant pilot to help; the secondary pilot does not have the auxiliary pilot it estimate.

The equation of AP can be done as following:

$$a_{i8} = -\sum_{k=1}^{7} \frac{a_{ik}\gamma_{ik}}{\gamma_{i8}} \qquad \text{(I.41)}$$

Where γ_{ik} is the coefficient interference, in order to have more precision in channel information, we have to get the interference information of the secondary pilots. Because the **AP** value just relate to the data interference coefficient.

The proposed feedback interference estimation algorithm is shown in figure:

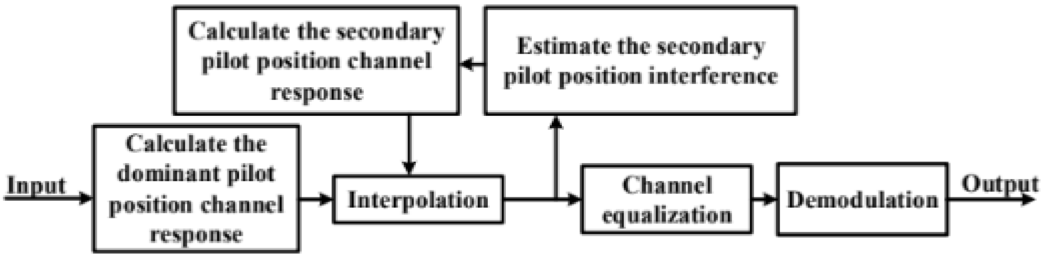


Figure 19 Feedback interference estimation algorithm [16]

In the first place of feedback interference estimation algorithm, all channel information is gotten by **AP** algorithm and interpolation, then we obtain the channel response of the secondary pilots by interpolation of the dominant pilots.

In secondary pilot position, we use the received signal and the estimated channel information to obtain the corresponding transmission signal and the estimation of the interference information. It can be expressed as following:

$$\hat{\boldsymbol{d}}_{k,n} + j\tilde{\boldsymbol{u}}_{k,n} = \frac{y_{k,n}}{\hat{h}_{k,n}} \qquad \text{(a)}$$

Where $\hat{d}_{k,n}$ represents the auxiliary pilot at the position **k**, **n** (**k**, **n** are the time domain and frequency domain) and $\tilde{u}_{k,n}$ represents the estimate of the imaginary interference $u_{k,n}$The estimation of the transmission signal obtained in (a) is corrected by the pilot signal Cause $d_{k,n}$ it's known. Then to correct the estimated channel information the feedback to the received signal is used. Finally, the channel information acquired by the second estimation is interpolated to obtain the response of the whole channel estimation, with equalization and demodulation followed [16]

The proposed feedback interference calculation algorithm is shown in figure:

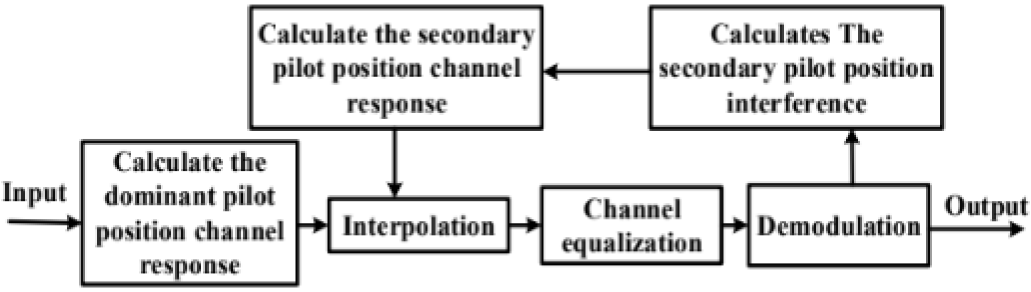


Figure 20 Feedback interference estimation algorithm [16]

In feedback interference calculation algorithm, we first obtain all the channel information or the entire channel information by **AP** algorithm and interpolation, and we obtain the channel response of the secondary pilots by interpolation the channel response of the dominant pilots. Then we calculated the interference information of the secondary pilots by the estimated transmission signal after equalization and demodulation as shown in equation (**c**) so we get:

$$\boldsymbol{d_{k,n} + \hat{u}_{k,n} =}$$

$$\boldsymbol{\hat{u}_{k,n} = \sum \hat{d}_{k,n}\ \hat{t}_{k-k',}\ n - n'} \qquad \text{(I.42)}$$

$$\boldsymbol{(k', n') \epsilon \Omega_{k,n}}$$

$$\boldsymbol{(k', n') \neq (k, n)}$$

$\hat{t}_{k-k',}\ n - n'$ denote the interference coefficient and $\hat{d}_{k,n}$ is the symbol after demodulation and $\Omega_{k,n}$ represents the interference calculation window around the secondary pilot and $\hat{u}_{k,n}$ represents the estimation of the interference information of the secondary pilot symbol.

When we get $d_{k,n} + \hat{u}_{k,n}$, we estimate again the channel response by using (**b**). At each moment we obtain the channel response of the pilot subcarrier, and then the channel information is acquired by re-interpolation with the equalization and demodulation are carried out again.

There's also another method of estimation which is data spreading this method consist on imposing the zero forcing condition on a certain number of neighboring symbols. This concept is used to avoid the use of the auxiliary pilot as proposed in [16]. instead in AP wish uses one symbol to combat interference on the pilot symbol.

IV. Channel interpolation:

The LS estimation makes it possible to obtain the frequency response (noisy) on the pilot carriers. In a large number of cases, it is then necessary to perform an interpolation to estimate the channel over the entire time-frequency network [17]. We have seen that LMMSE could serve as an interpolator filter. However, its complexity means that we often prefer to use simpler interpolations, such as those presented in this part. These have the particularity of being based only on interpolating polynomials, and do not need any characteristic of the channel or signal.

IV.1. Linear interpolation :

Linear interpolation is relatively simple because it is based on a degree one interpolating polynomial [17]. For a value $f \in [f_p + \delta_f, f_p + \delta_f]$, the estimated channel $\hat{H}\ (f)$ is the average between $\hat{H}\ (f_p)$ and $\hat{H}(\ f_p + \delta_f)$, weighted by the distance$f_p + \delta_f$. So, we get:

$$\hat{\boldsymbol{H}}\ (\boldsymbol{f}) = \hat{\boldsymbol{H}}(\boldsymbol{f_p}) + (\boldsymbol{f} - \boldsymbol{f_p})\,\hat{\boldsymbol{H}}(\boldsymbol{f_p} + \boldsymbol{\delta_p}) - \frac{\hat{\boldsymbol{H}}(\boldsymbol{f_p})}{\boldsymbol{f_p}+\boldsymbol{\delta_f}-\boldsymbol{f_p}} \qquad \text{(I.43)}$$

Fig. (b) illustrates the principle of linear interpolation between two frequency positions f_p and $f_p + \delta_f$ drivers. Although more precise than NN interpolation, linear interpolation has poor results when the channels are very selective.

The linear channel interpolation can be implemented by using digital filtering such as Farrow-structure [18]

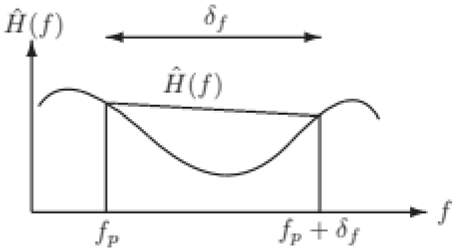


Figure 21 Illustration of the principle of linear interpolation. [18]

IV.2. Polynomial interpolation:

The principle of polynomial interpolation is to approximate Hf by a polynomial of degree P - 1, where P is the number of pilots per symbol. Using Lagrange polynomials {L0, L1, ..., LP −1} as a basis, we obtain χ (f) the interpolating polynomial:

$$\boldsymbol{x(f)} = \sum_{\boldsymbol{p=0}}^{\boldsymbol{P-1}} \boldsymbol{Lp(f)} \times (\boldsymbol{f_p}) = \sum_{\boldsymbol{p=0}}^{\boldsymbol{P-1}} \boldsymbol{Lp(f)}\ \boldsymbol{\widehat{H}}\ (\boldsymbol{f_p}) \quad \text{(I.44)}$$

Where $LP \times f_p = 1$ and $\chi(f_p) = \widehat{H}\left(f_p\right)$. this phenomenon, called the Runge effect, makes interpolation by Lagrange polynomials hardly applicable in practice. One solution to limit the Runge effect is to cut all the control points into packets of four consecutive points {f_p, ..., f_p+ 3δp} and apply an interpolation by a polynomial of degree three on each of the intervals. This method called cubic piece interpolation is commonly used. However, this technique makes the interpolating function discontinuous on each node between the different pieces considered. To obtain a continuous function over the entire study interval (corresponding to band B).[14]

IV.3. Spline Cubic interpolation:

The cubic spline interpolation uses Hermite polynomials as a basis, which ensures continuity at each control point by adding a condition on the first derivative of the polynomial at each control point. An additional difference with Lagrange's cubic interpolation by piece is that a degree three polynomial is used between each node.

Spline interpolations works better than linear interpolation for comb pilot arrangement [19]

IV.4. Low pass interpolation :

Low pass interpolation consists of inserting zeros in the original sequence then we apply a low pass RIF filter to allow the original data to pass through unchanged and interpolates between such that the mean-square error between the interpolated points and the ideal values is minimized.[4]

V. Equalization:

Although the guard time which has longer duration than the delay spread of a multipath channel can eliminate ISI because of the previous symbol, but it is still have some ISI because of the frequency selectivity of the channel. In order to compensate this distortion, a one-tap channel equalizer is needed. At the output of FFT on the receiver side, the sample at each subcarrier is multiplied by the coefficient of the corresponding channel equalizer. The coefficient of an equalizer can be calculated based on the zero-forcing (ZF) criterion or the minimum mean square error (MMSE) criterion [20]. The ZF criterion forces ISI to be zero at the sampling instant of each subcarrier. The coefficient of a one-tap ZF equalizer is calculated as follows:

$$C_n = \frac{1}{H_n} \qquad \text{(I.45)}$$

where Hn is the channel frequency response within the bandwidth of the n-th subcarrier. The disadvantage of ZF criterion is that it enhances noise at the n-th subcarrier if Hn is small, which corresponds to spectral nulls.[21]

In order to achieve a good performance the receiver has to know the impact of the channel. The problem is how to extract this information in an efficient way. Conventionally, known bits are multiplexed into the data sequence in order to estimate the channel. In OFDM systems, there are some specific opportunities for channel estimation since several adjacent carriers are used for the transmission as opposed to signal-carrier systems. [14]

The task of the equalizer is to compensate for the influence of the channel. This compensation requires, however, that an estimate of the channel response of available. There are many alternatives identifying the channel response and depending on the channel statistics and the modulation scheme different solutions are preferable. Often the channel impulse response or the frequency derived from training sequences or pilot symbols, but it's also possible to use non-pilot-aided approaches, so called blind algorithms. The use of training symbols means that the complexity of the receiver can be kept rather low and, in case that the training symbols are received without influence from each other, the processes of channel estimation and detection can be separated. Also, training symbols are suitable for packet based communication since this result in robust behavior with no significant transient effects. [14]

In the general case the optimal receiver, in the sense of minimum sequence error probability when no prior channel estimates are available is the maximum-likelihood (ML) receiver making joint detection and channel estimation. The task is to find a channel estimate and a data sequence that make the actual received sequence that make the actual received sequence as probable as possible

$$(\hat{\boldsymbol{c}}, \hat{\boldsymbol{h}})\boldsymbol{ML} = \mathbf{arg\,max}\,\boldsymbol{f}\ (\boldsymbol{r}|\boldsymbol{c}, \boldsymbol{h}) \qquad \text{(I.46)}$$

Where **r** is the received sequence **c** is the data sequence, **h** is the sequence of the channel responses, **f (.)** denotes the probability density function and arg max denotes the arguments maximizing the function. This approach leads, however to complex receiver structures and therefore

other less complex alternatives are of interest. In case channel estimation and data detection can be performed separately, the structure can be simplified by deriving the channel estimates before detection takes place. Given a certain channel estimate **h** the optimal detector, this equalizer gives the maximum-likelihood sequence, which means that it makes a decision in favor of the sequence that maximizes the probability of receiving the actual received sequence.[4]

$$\hat{\boldsymbol{C}} = \mathbf{arg\,max}\,\boldsymbol{f}(\boldsymbol{r}|\boldsymbol{c}, \hat{\boldsymbol{h}}). \qquad \text{(I.47)}$$

VI. Conclusion:

This chapter allowed us to describe the radio link environment in a

Telecommunication system.

We first explained the principle of a wave radio propagation and its characteristics and properties, and then we did a little description of some multipath propagation and fading phenomenon as well as the modeling of the channel in the case of multipath channel.

Also a variety of channel estimation techniques were investigated in details.

CHAPTER II

Multi-carriers solution

I. Introduction:

In this chapter, we will define a new transmission technology, which is FBMC (Filter Bank Multi-Carrier); It is also considered as a continuation or rather an alternative to the famous orthogonal frequency division multiplexing (OFDM).

In order to guarantee interference-free communication, OFDM uses a cyclic prefix (CP) and a very high level of the side lobes causing a power leak between the different subcarriers.

The OFDM uses a rectangular pulse-shaped filter; however, the FBMC uses a different modulation scheme and a prototype filter. Indeed, the FBMC technique retains the advantages of OFDM and improves its weak points.

II. Single carrier transmission

➢ System model

Figure 1 shows a typical scheme configuration for a single-carrier communication system.

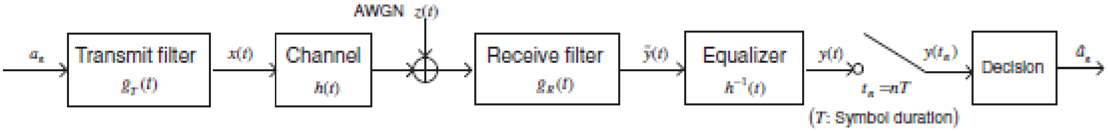


Figure 1 single-carrier baseband communication system model [14]

Consider that our channel is a band limited channel h(t) and its bandwidth is W and we consider also The transmit symbols {an}, each symbol has symbol period of T seconds, that is, a data rate of R=1/T, are pulse-shaped by a transmit filter $g_t\ (T)$in the transmitter.

When we receive them through the channel, they are processed with the receive filter, equalizer, and detector in the receiver.

Consider $g_t\ (T)$ is the impulse response of the transmit filter, and $g_r\ (R)$ is the receive filter, and $h^{-1}\ (t)$ is the equalizer.

The output of the equalizer can be expressed as:

$$\boldsymbol{y(t) = \sum_{m=-\infty}^{\infty} a_m g(t - mT) + z(t)} \qquad \text{(II.1)}$$

Where z (t) is an additive noise and g (t) is the impulse response of overall end-to-end system, Given as:

$$\boldsymbol{g(t) = g_T(t) \times h(t) \times g_R(t) \times h^{-1}(t)} \qquad \text{(II.2)}$$

We use the equalizer here to correct the effect of channel.

In this part, we consider that the equalizer, as given in Equation (2) corrects the effect of the channel perfectly.

Consequently, the overall impulse response is submissive to only the transmit and the receive filters.

If we ignored the noise term, we can express the sampled output signal of the equalizer as:

$$\boldsymbol{y(t_n) = \sum_{m=-\infty}^{\infty} a_m\, g\big((n-m)T\big)\ with\ t_n = nT} \qquad \text{(II.3)}$$

Isolating the nth sample to detect an, Equation (3) can be written as:

$$\boldsymbol{y(t_n) = a_n g(0) + \sum_{m=-\infty, m\neq n}^{\infty} a_m g((n-m)T)} \qquad \text{(II.4)}$$

Due to the finite channel bandwidth the impulse response of transmit filter cannot be time-limited.

In case that:

$$\boldsymbol{g\big((n-m)T\big) \neq 0\ for\ \forall_m \neq n} \qquad \text{(II.5)}$$

The second term in equation (4) rest as an inter-symbol interference (ISI) to a_n.

Actually, the chain of the overall impulse response is the cause of inter-symbol interference, that what degrade the performance of a digital communication system.

Wherefore, the transmit filter and receive filter must be produced by design in a way to minimize maximum or eliminate the ISI in a practical system.

In the figure 2 we illustrate how the ISI goes through the chain of the overall impulse response in the receiver.

As illustrated her, the extent of ISI depends on the duration of a symbol period T more the symbol period is short more the ISI is important. This implies that unless

$$\boldsymbol{g\big((n-m)T\big) \neq 0\ for\ \forall_m \neq n,} \qquad \text{(II.6)}$$

The ISI becomes Significant as the data rate increases (it means, decreasing T in Figure 2) in a single-carrier system.[14]

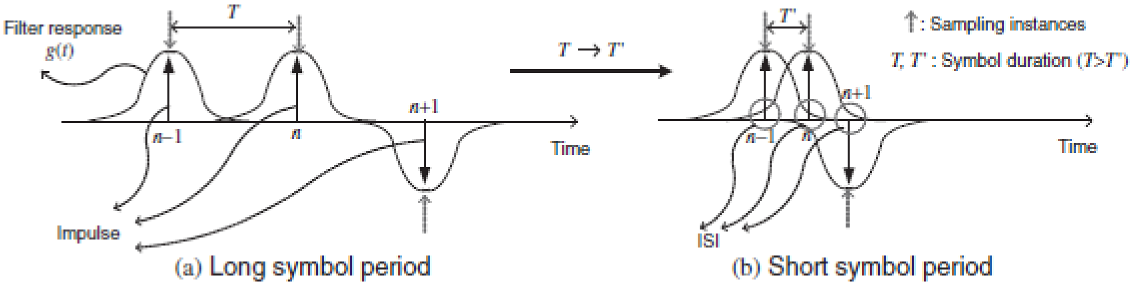


Figure 2 illustration inter-symbol interference (ISI) and symbol period [14]

➢ ISI and Nyquist criterion

In Equation (II.6), by fulfilling the following time-domain condition on the overall impulse response ISI can be totally eliminated.

$$\boldsymbol{g(nT) = \delta[n] = \begin{cases} 1, & n = 0 \\ 0, & n \neq 0 \end{cases}} \qquad \text{(II.7)}$$

Note that the condition in equation (II.7) is equivalent to the following frequency domain condition:

$$\boldsymbol{\sum_{i=-\infty}^{\infty} G\left(f - \frac{i}{T}\right) = T} \qquad \text{(II.8)}$$

Consider the Fourier Transform of g(t) is G(f) which the equation (II.7) and equation (II.8) are known as the Nyquist criterion, that what guaranty an ISI free communication.

Also when we have a short symbol period T for high data rate transmission in a single carrier transmission system.

The filters that satisfied the Nyquist criterion are called: Nyquist filters.
A Nyquist filter is an ideal low pass filter; it has impulse response type sinc function, or rectangular pulse (or brick-wall) type of frequency response as described by:

$$\boldsymbol{G_I = \frac{1}{2W} rect\left(\frac{f}{2W}\right) = \begin{cases} T, & f \leq \frac{1}{2T} \\ 0, & f > \frac{1}{2T} \end{cases}} \qquad \text{(II.9)}$$

In equation (II.9) W= R/2 =1/2T denote that R is the Nyquist rate and W is Nequist bandwidth. Moreover, the Nyquist bandwidth W is the minimum possible bandwidth that is required to realize the date rate R without ISI.

However, the ideal filter in Equation (II.9) is not physically realizable because its impulse response is not causal.

$$\boldsymbol{G_{RC}}\,(\boldsymbol{f}) = \begin{cases} \boldsymbol{T}, & \boldsymbol{f} \le \frac{\mathbf{1}-\boldsymbol{r}}{\mathbf{2}\boldsymbol{T}} \\ \frac{\boldsymbol{T}}{\mathbf{2}}\left\{\mathbf{1} + \mathbf{cos}\,\frac{\boldsymbol{\pi T}}{\boldsymbol{r}}\left(\boldsymbol{f} - \mathbf{1} - \frac{\boldsymbol{r}}{\mathbf{2}\boldsymbol{T}}\right)\right\}, & \frac{\mathbf{1}-\boldsymbol{r}}{\mathbf{2}\boldsymbol{T}} < \boldsymbol{f} \le \frac{\mathbf{1}+\boldsymbol{r}}{\mathbf{2}\boldsymbol{T}} \\ \mathbf{0}, & \boldsymbol{f} > \frac{\mathbf{1}+\boldsymbol{r}}{\mathbf{2}\boldsymbol{T}} \end{cases} \qquad \text{(II.10)}$$

Consider r is the roll-off factor that tailors the total bandwidth and$0 \le r \le 1$.
Here we have the equation (II.10) which satisfied the ISI-free condition Equation (II.8), but is not as sharp as the frequency response of an ideal LPF.

Note that the raised cosine frequency response in Equation (II.10) occupies a frequency range wider than the Nyquist bandwidth.
The actual bandwidth is governed by the roll-off factor r.

Figures 3(a) and (b) show the impulse and frequency responses of raised cosine filters with the roll-off factors of r = 0, 0.5, and 1, respectively.

Note that the raised cosine filter with r = 0 happens to be identical to the ideal LPF, and the raised cosine filter with r = 1 occupies twice the Nyquist bandwidth.

In the special case where the channel is ideal, we require:

$$\boldsymbol{G_R^*}(\boldsymbol{f}) = \boldsymbol{G_R^*}\,(\boldsymbol{f})\ \ \boldsymbol{where}\quad \boldsymbol{G_T}(\boldsymbol{f}) = \ \boldsymbol{G_R}\,(\mathrm{f}) \qquad \text{(II.11)}$$

Where are the frequency response of the transmit filter $g_T(t)$ and receiver filter $g_R(t)$ respectively. Since,

$$(\boldsymbol{f}) = \boldsymbol{G_T^*}(\boldsymbol{f}), \boldsymbol{G_{RC}}(\boldsymbol{f}) = (\boldsymbol{G_T}(\boldsymbol{f}))^{\mathbf{2}}\ \ \boldsymbol{or}\ \boldsymbol{G_T}(\boldsymbol{f}) = \sqrt{\boldsymbol{G_{RC}}(\boldsymbol{f})} \qquad \text{(II.12)}$$

And thus, the transmit filter must have the following frequency response, which is known as the square-root raised cosine filters: [14]

$$\boldsymbol{G_{SRRC}}(\boldsymbol{f}) = \begin{cases} \sqrt{\boldsymbol{T}}, & \boldsymbol{f} \le \frac{\mathbf{1}-\boldsymbol{r}}{\mathbf{2}\boldsymbol{T}} \\ \sqrt{\frac{\boldsymbol{T}}{\mathbf{2}}\left\{\mathbf{1} + \mathbf{cos}\,\frac{\boldsymbol{\pi T}}{\boldsymbol{r}}\left(\boldsymbol{f} - \frac{\mathbf{1}-\boldsymbol{r}}{\mathbf{2}\boldsymbol{T}}\right)\right\}}, & \frac{\mathbf{1}-\boldsymbol{r}}{\mathbf{2}\boldsymbol{T}} < \boldsymbol{f} \le \frac{\mathbf{1}+\boldsymbol{r}}{\mathbf{2}\boldsymbol{T}} \\ \mathbf{0}, & \boldsymbol{f} > \frac{\mathbf{1}+\boldsymbol{r}}{\mathbf{2}\boldsymbol{T}} \end{cases} \qquad \text{(II.13)}$$

➢ Limitation of single-carrier Transmission for high Data Rate:

In the interest of supporting the symbol rate of Rs, symbols per second, the minimum required bandwidth is the Nyquist bandwidth, which is given by Rs/2 [Hz].

It means that it is necessary to have a wider bandwidth in order to support a higher data rate in a single-carrier transmission.

Until now, it has considered that the channel is perfectly corrected by the equalizer.

However, the signal bandwidth becomes larger when the symbol rate increase. The link udergo from multipath fading incurring the inter-symbol interference (ISI) when the signal bandwidth becomes larger than the coherence bandwidth in wireless channel.

Generally, we use adaptive equalizers to deal with the ISI incurred by the time-varying multi-path fading channel. Moreover, the complexity of an equalizer increases with the data rate.

More precisely, adaptive equalizers are implemented by finite impulse response (FIR) filters, with the adaptive tap coefficients that are adjusted so as to minimize the effect of ISI.

Actually, more equalizer taps are required as the ISI becomes significant, for example, when the data rate increases. The optimum detector for the multi-path fading channel is a maximum-likelihood sequence detector (MLSD), which bases its decisions on the observation of a sequence of received symbols over successive symbol intervals, in favor of maximizing the posteriori probability.

Note that its complexity depends on the modulation order and the number of multi-paths.
Let M and L denote the number of possible signal points for each modulation symbol and the span of ISI incurred over the multi-path fading channel, respectively.
Due to a memory of length L for the span of ISI, ML corresponding Euclidean distance path metrics must be evaluated to select the best sequence in the MLSD.

When a more efficient transmission is sought by increasing M and a high data rate is implemented, the complexity of the optimum equalizer becomes prohibitive, for example:

$$\boldsymbol{M^L = 64^{16} \; for \; L \approx 16}$$

with 64-QAMat the data rate of10Mbps over the multi-path fading channel with a delay spread of 10 μsWhen Man d L are too large, other more practical yet suboptimum equalizers, such as MMSE or LS equalizer, can be used.

However, the complexity of these suboptimum equalizers is still too enormous to be implemented as the ISI increases with the data rate.
This particular situation can be explained by the fact that the inverse function (a frequency-domain response of equalizer) becomes sharper as the frequency-selectivity of the channel increases.[14]

In conclusion, a high data rate single-carrier transmission may not be feasible due to too much complexity of the equalizer in the receiver.

I. Multi-carrier transmission

➢ Basic structure of a multi-carrier transmission scheme

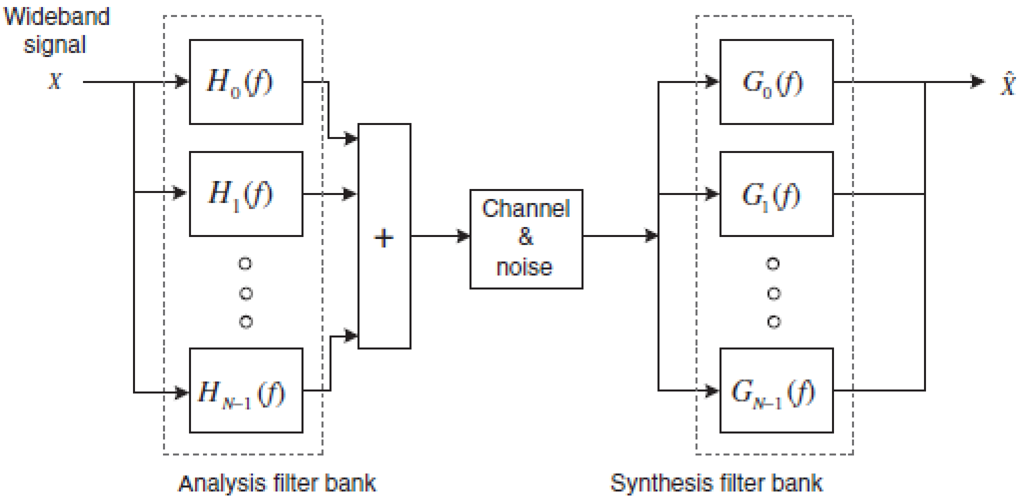
To get over the frequency selectivity of the wideband channel undergo by single-carrier transmission, multiple carriers can be used for high rate data transmission.

Figure 3(a) shows the basic structure and concept of a multi-carrier transmission system [23].

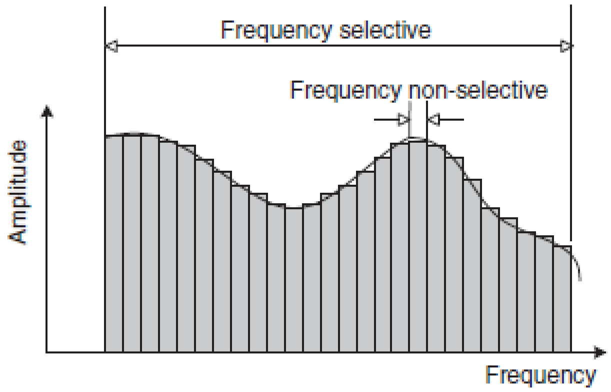
Here, a wideband signal is analyzed (through multiple narrowband filter $H_k(f)$ into several narrowband signals at the transmitter and is synthesized (through multiple narrowband filter $G_k\ (f)$each being matched to $H_k(f)$ at the receiver so, that the frequency-selective wideband channel can be approximated by multiple frequency-flat narrowband channels as depicted in Figure 3(b).

Note that the frequency-non selectivity of narrowband channels reduces the complexity of the equalizer for each sub channel.

As long as the orthogonality among the sub channels is maintained the ICI (inter-carrier interference) can be suppressed leading to distortion less transmission [23].



(a) The structure of multichannel transmission system



(b) The frequency response of multichannel transmission system

Figure 3 Structure and frequency characteristic of multichannel transmission system.[23]

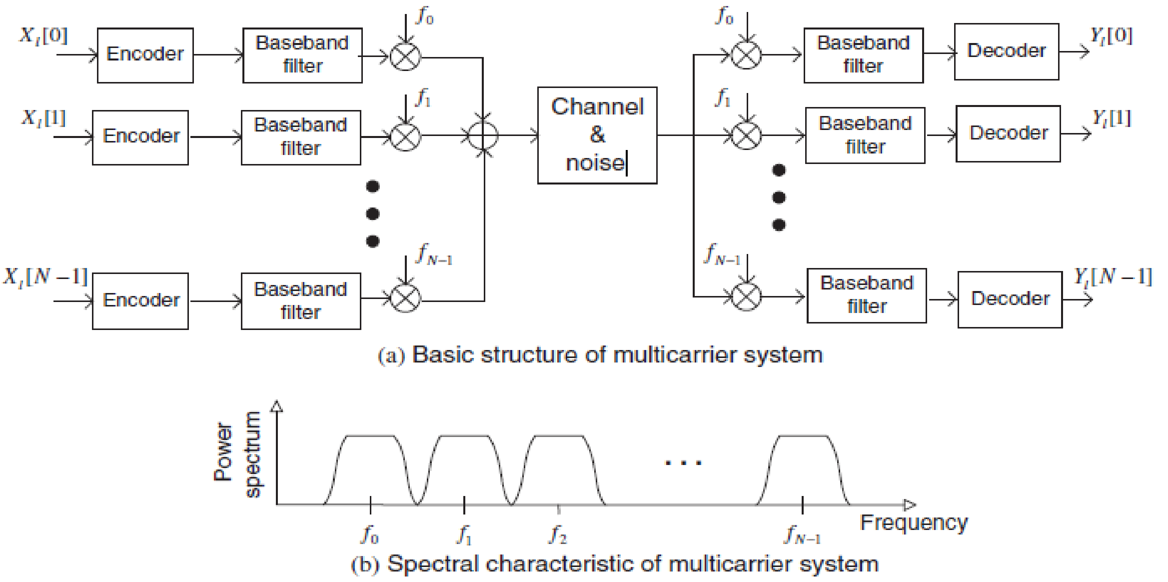
In the multi-channel system, let the wideband be divided into N narrow band sub channels, which have the subcarrier frequency of $f_k, k = 0,1,2 \ldots.., N-1$.

Figure 4(a) shows the basic structure of a multi-carrier communication scheme, which is one specific form of the multichannel system, where the different symbols are transmitted with orthogonal sub channels in parallel form.

Let $\boldsymbol{X_l}\,[\boldsymbol{k}]\ and\ \boldsymbol{Y_l}\,[\boldsymbol{k}]$ denote the transmitted and received signals carried at the carrier frequency $\boldsymbol{f_k}$ in the $\boldsymbol{l_{th}}$ symbol interval, respectively.
It implies that multi-carrier transmission can be regarded as a kind of FDMA (frequency division multiple access) method.

Figure 4(b) illustrates a transmitted signal spectrum in the multi-carrier transmission system, which occupies multiple sub bands of equal bandwidth, each centered at the different carrier frequency.

If each sub channel is band limited as depicted in Figure (b), it becomes an FMT (Filtered Multi-Tone) transmission, While a FMT type of multicarrier transmission system can cope with the frequency selectivity of a wideband channel, its implementation becomes complex since it involves more encoders/decoders and oscillators, and higher quality filters as the number of subcarriers increases.[14]



(a) Basic structure of multicarrier system

(b) Spectral characteristic of multicarrier system

Figure 4 structure and spectral characteristic of multicarrier transmission.[14]

➢ Single carrier VS Multi carrier :

In the previous sub sections, we have discussed the single-carrier and multi-carrier transmission schemes.

It is clear that each of these schemes has its own advantages and disadvantages. The single-carrier scheme may not be useful for a high rate wireless transmission, simply because it requires a high-complexity equalizer to deal with the inter-symbol interference problem in the multi-path fading channel or equivalently, frequency-selective fading channel.

Meanwhile, the multi-carrier scheme is useful for a high rate wireless transmission, which does not involve the complexity of channel equalization.

II. OFDM:

➢ Definition

OFDM Modulation (Orthogonal Frequency Division Multiplexing) is one of the techniques where multi-carrier transmission schemes are implemented.

This technique divides the frequency band into a number of sub channels equal to N (or subcarriers) orthogonal and uniform, [23] it means the symbols will be distributed over a large

Number of carriers at low speed, unlike conventional systems which transmit the symbols in series, each symbol then occupies all the bandwidth available.

➢ Basic principle of OFDM

1. Orthogonality

The OFDM modulation technique allows a high spectral overlap between the sub carriers. However, the carriers of OFDM must be orthogonal to each other in the time domain and frequency domain in the same domain for a proper functioning figure:

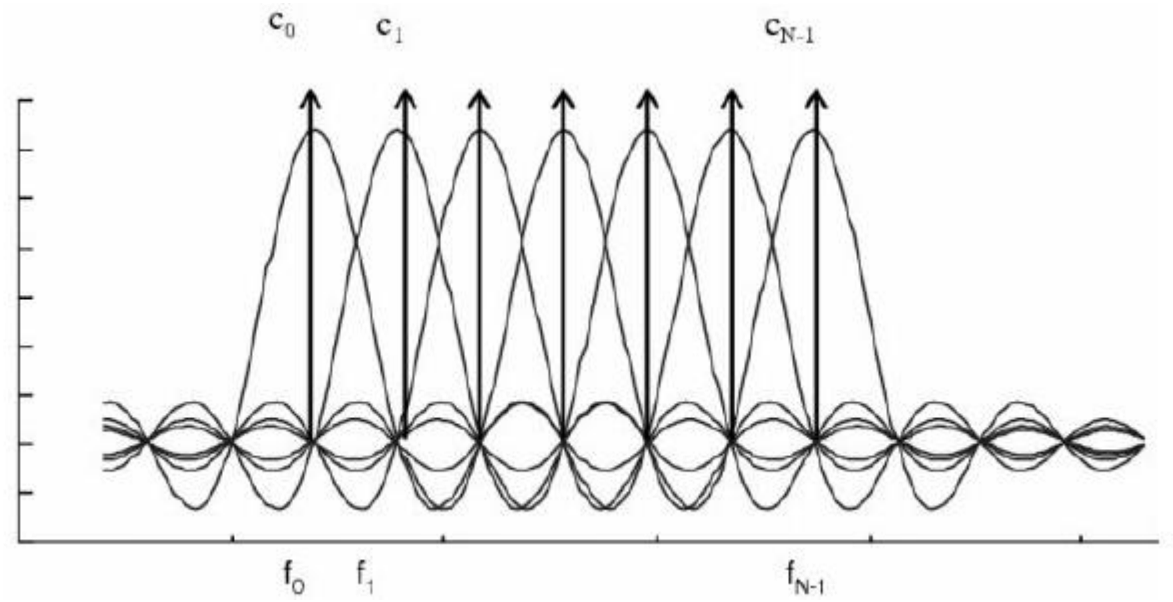


Figure 5 spectral of the different carriers

These signals are defined to be orthogonal if:

- The frequency $f_n$of two adjacent frequencies is: $\boldsymbol{f_n = \frac{1}{T_n}}$. (II.14)
- It can be seen that the spectral value of all other sub channels is exactly zero at the maximum of each subcarrier frequency. In other word the integral of the products for their periods is zero, that is,

$$\frac{1}{T_{sym}}\int_0^{T_{sym}} e^{-j2\pi f_k t} e^{-j2\pi f_i t}\, dt = \frac{1}{T_{sym}}\int_0^{T_{sym}} e^{\frac{j2\pi k}{T_{sym}}t} e^{-\frac{j2\pi i}{T_{sym}}t}\, dt \qquad \text{(II.15)}$$

$$= \frac{1}{T_{sym}}\int_0^{T_{sym}} e^{j2\pi\frac{k-i}{T_{sym}}t}\, dt \qquad \text{(II.16)}$$

$$= \begin{cases} 1, & \forall\ ineger\ k = i \\ 0, & otherwise \end{cases} \qquad \text{(II.17)}$$

$\{e^{j2\pi f_k t}\}_{k\,=\,0}^{N\,-\,1}$ Is the time limited, which represent different subcarriers at $f_k = \frac{k}{T_{sym}}$

In the OFDM signal where:

$$0 \le t \le Tsym$$

This orthogonality between OFDM subcarrier signals avoids the interference inter symbol.[8]

Figure 6 shows that then the frequency band is optimally occupied, since the spectrum is almost flat in this band.

The band occupied is roughly B = N / TS (excluding the side lobes on either side of the band), each subcarrier occupying approximately 1 / TS.

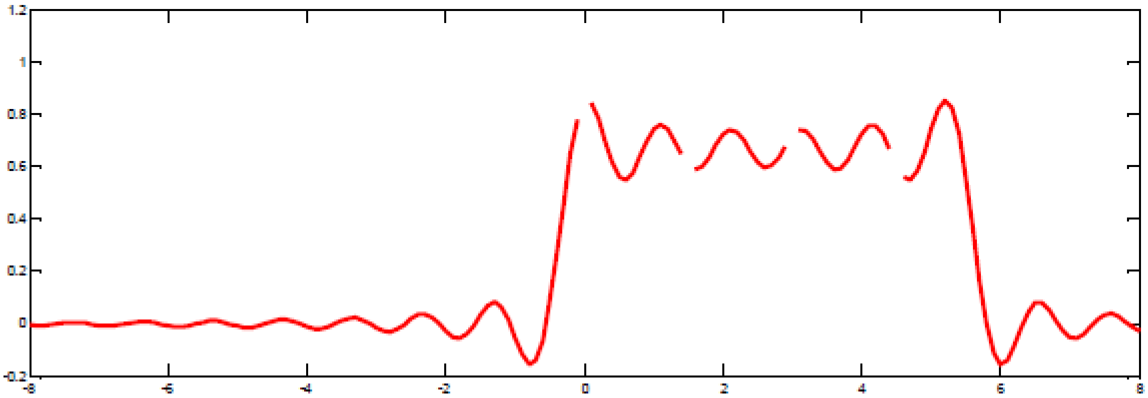


Figure 6 spectral of OFDM signal for 8 carriers [8]

2. OFDM scheme:

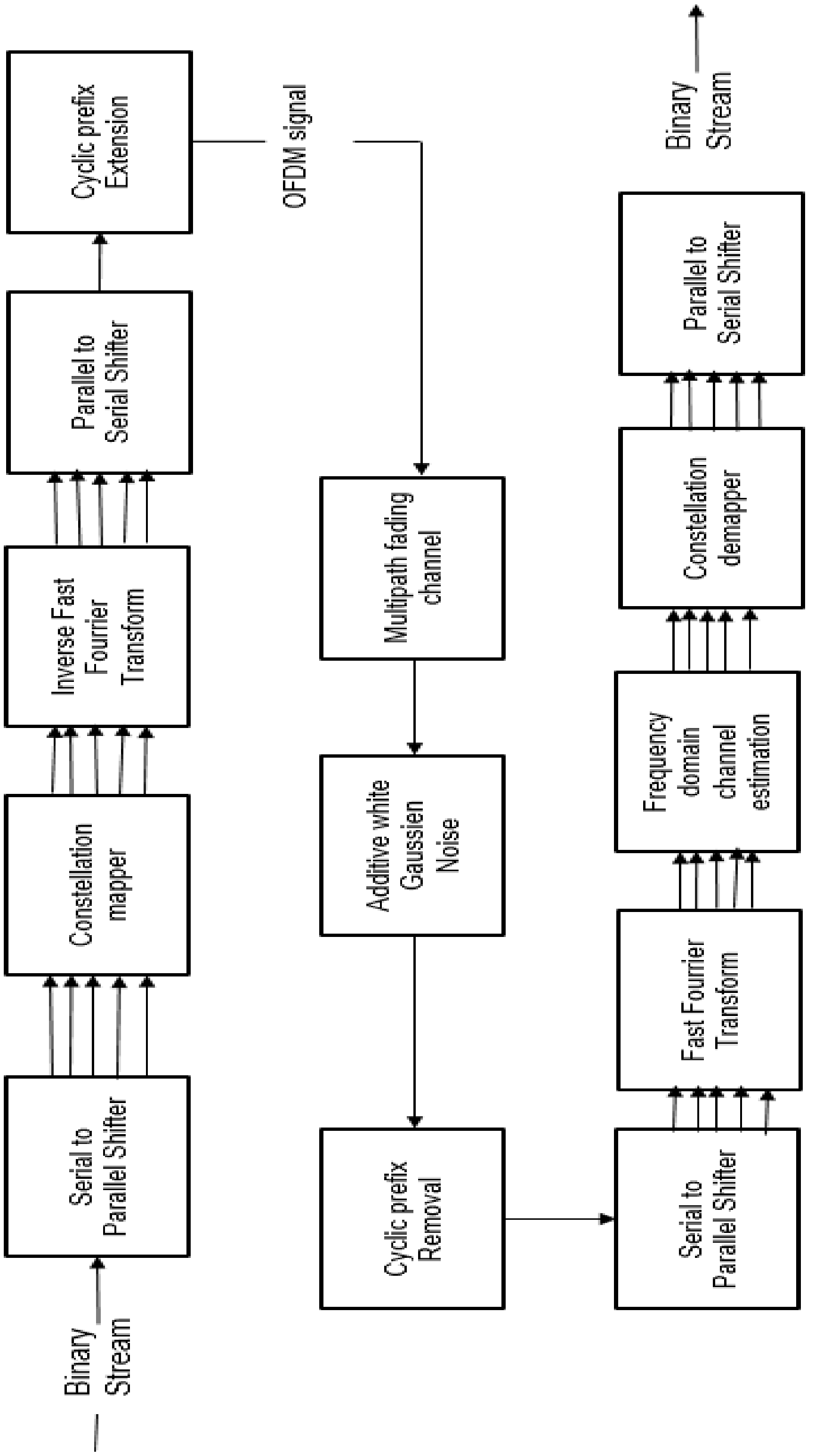


figure 7 OFDM scheme

➢ OFDM transmitter maps the message into modulation mapping module, there are many different kinds of modulation schemes like PSK and QAM.

- After modulation the symbols will be subsequently converted into N parallel signal (streams).

- We insert pilot here to ensure at receiver the frequency offset corruption and phase noise.

- Then we have IFFT, it’s used to transform signal from discreet frequency domain to discreet time domain.

- We insert cyclic prefix.

- The parallel signal is transmitted into serial signal.

- We convert our digital signal to analog signal.

- there are many different kinds of noises in channel. But additive white Gaussian noise is the typical noise in channel.

- At the receiver suite, inverse transforms are used corresponding to the transmitter side separately.

3. Principle of Modulation:

For the repair of the data to be transmitted on the N carriers, The symbols must be grouped by packets of N.

Symbols are complex numbers offer the constellations often of QAM or PSK type he signal is modulated by the K_{th} train of symbols among the N trains.

And C_k the complex form of modulated carrier signal of train K. gathering of OFDM symbols gives us the total signal s (t) (figure 8).[24]

$$\boldsymbol{s(t) = \sum_{K=0}^{N-1} C_k e^{j2\pi f_k t}} \qquad \text{(II.18)}$$

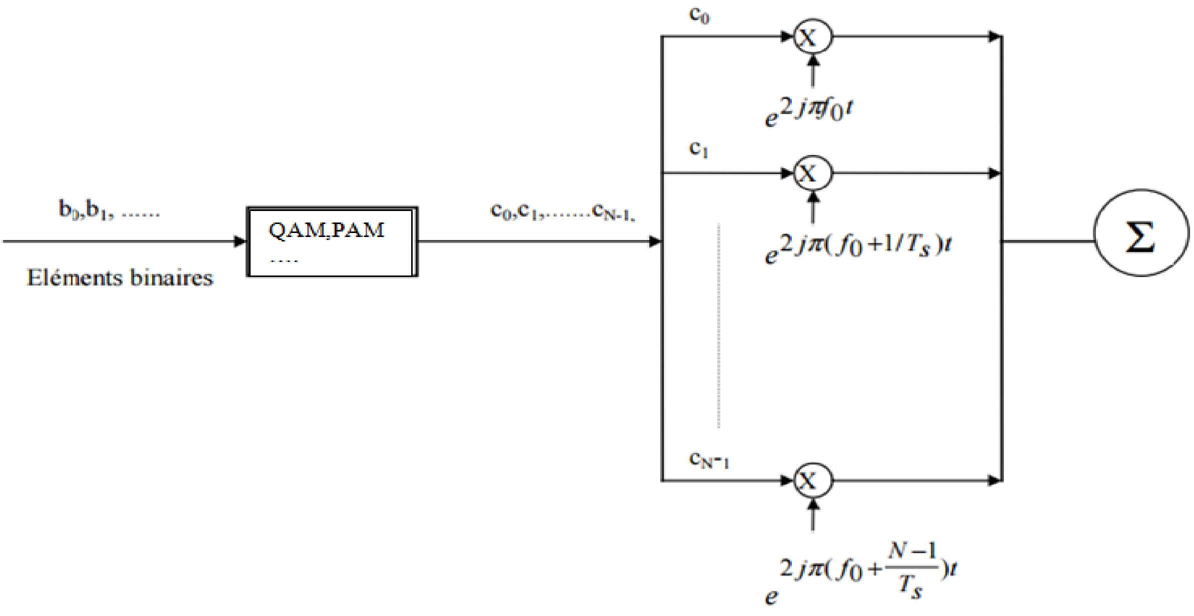


Figure 8 OFDM modulation scheme.[24]

4. Principle of demodulation:

The signal received by the receiver is written over a symbol duration Ts:

$$\boldsymbol{y(t) = \sum_{K=0}^{N-1} C_K H_K\ (t) e^{-j2\pi f_k t}} \qquad \text{(II.19)}$$

$H_k(t)$ is the transfer function of the channel which varies slowly.
The demodulation would consist in demodulating the signal according to the N subcarriers according to figure9.[24]

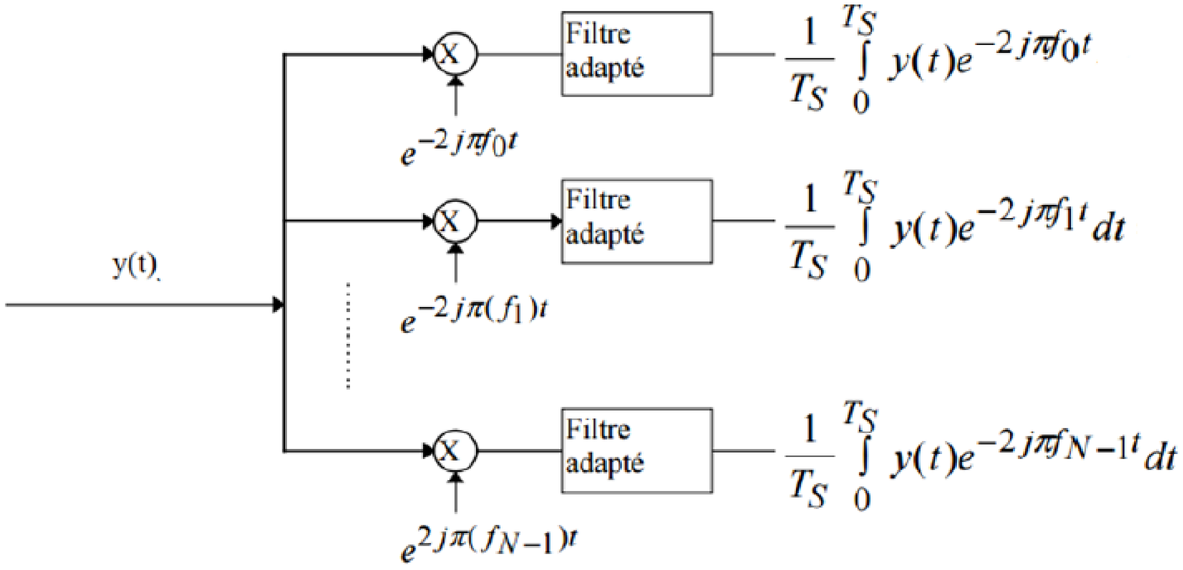


Figure 9 OFDM demodulation scheme [24]

5. The guard interval:

The perturbation of the channel of propagation induces loss of orthogonality between the subcarriers and appearance of interference inter symbol due to the multipath.

In order to eliminate these interferences, there is a simple solution for this, by increasing the number N of subcarriers to increase the symbol duration Ts.

However, this technique faces several constraints, consistency time of channel, the Doppler Effect and technological constraints like phase noise of oscillators, all these constraints limit the use of this technique.

Another technique permits to eliminate these ISI which is adding interval of guard of period Tg higher or equals the sprawl T of channel impulse response Previously the symbol OFDM transmit.[8]

The guard interval of OFDM can be inserted in two different ways:

1. **Cyclic Prefix (CP):** is a technique, which extends the OFDM symbol by inserting a copy of the last samples of the OFDM symbol into its front.

Consider Tg the length of CP then the extended OFDM symbols have a duration of:

$$\mathbf{T_{sym} = T_{sub} + T_g}$$

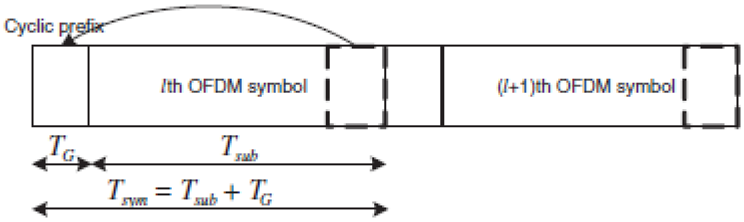


Figure 10 OFDM symbols with CP.[14]

2. **Cyclic suffix (CS):** is also a technique, which extends the OFDM symbol by inserting the difference between CP and CS is only that CS is the copy of the head part of an effective OFDM symbol, and it is inserted at the end of the symbol.

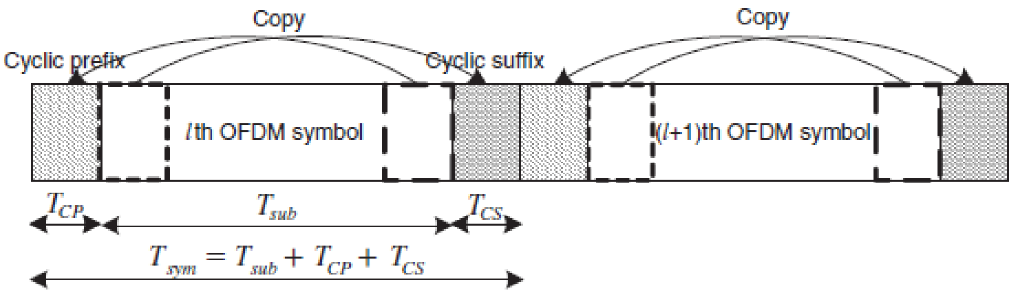


Figure 11 OFDM symbols with both CP and CS.[14]

3. **The Zero padding (ZP):** Insert zeros instead of guard interval and no signal is transmitted during the guard interval.

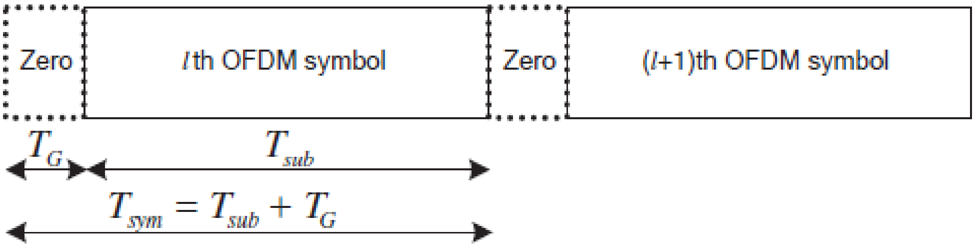


Figure 12 OFDM symbols with ZP.[14]

6. Merit:

- It can sand a lot of data in a narrowband.
- OFDM technology can continuously monitor the sudden changes in communication characteristics on the transmission medium.

- This technique automatically detects the specific carriers in the transmission medium, which has high signal attenuation or interference pulses.

- OFDM has a strong anti-fading ability through the sub-carrier joint coding.

- OFDM technology is very resistant to narrowband interference because, these disturbances only affect a small part of the sub-channels.

- The OFDM implementation method based on IFFT/FFT can be used.

- Channel utilization is very high.
- OFDM greatly improves the capacity of the system to transmit information.

- OFDM system can be used to achieve an attractive single frequency network.

7. Demerits:

- OFDM is more sensitive to carrier frequency offset and phase noise than single carrier systems.

- Signal OFDM system has a high peak-to-average ratio (PAR) which results in a decrease in the power efficiency of the RF amplifier.

III. Filter Bank Multi Carrier:

➢ Definition :

FBMC systems have the same principle of OFDM systems but it uses prototype filter.

In the largest sense, a filter bank is a network of M filters which processes M input signals to produce M output signals.

There are two different types of filter banks: synthesis filters and analysis filters which are placed respectively on the transmitter / receiver side.

These two cases are shown in Figure The system of figure (a) is called synthesis filter bank (SFB), the system of figure (b) is called analysis filter bank (AFB).

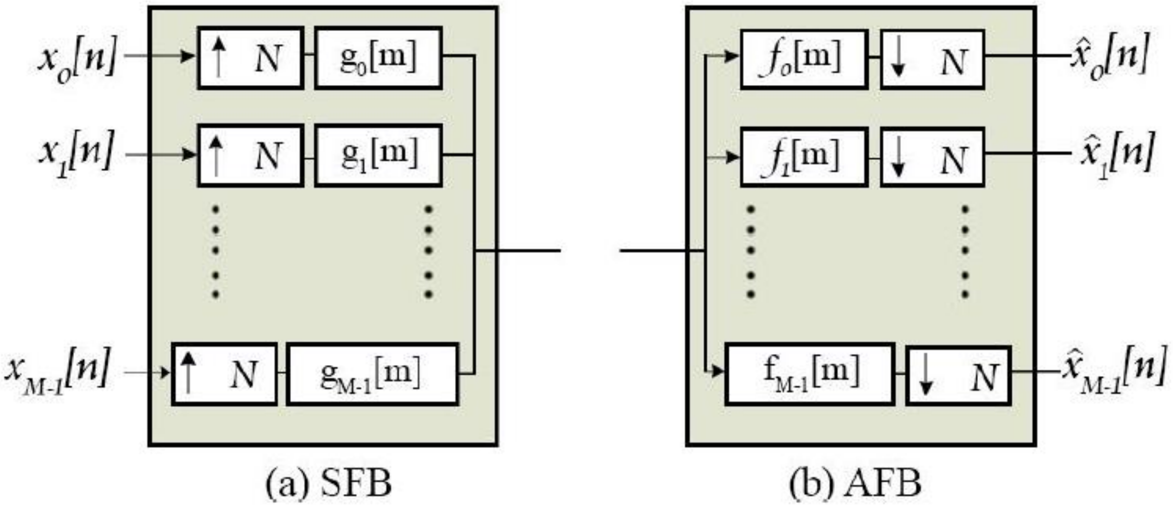


Figure 13 Filter Bank SFB and AFB.[24]

At the transmitter, M oversamplers and M synthesis filters will form a bank of synthesis filters.

The data X k (z) or K = 0.1 ... M-1 are then oversampled by M / 2 and filtered with a synthesis filter GK (z), finally all the sub-channels are added to form RK (z).

At the receiver level, M subsamplers and M analysis filters will form the analysis filter bank so that the signal YK (z) is filtered with the analysis filter FK(z), then subsampled by M /2 to for Xk(z).

The analysis and synthesis filter banks are generally associated: the first decomposes a signal to apply processing to each sub-band signal, and the second recombines the processed sub-band signals to build the modified signal.

If the signals X (n) and X (n) are equal (possibly to digital errors and to a near delay) we say that the system is perfectly reconstructed.

This property can be verified by a couple of analysis / synthesis filter banks.[24]

1. Basic Multirate opertations:

two basic operations in digital multi-rate signal processing are decimation and interpolation. These operations can be carried out by building blocks called decimators and interpolators (detail in the following sections). [24]

Decimation:

Decimation is done in AFB, it consists of the filtering of the input signlas and the subsequent downsampling as we can see in the branche of analysis filter banks of figure.

Downsimpling reduces the sampling rate by selecting every Nth sample of the filtered signal.

A divice of decimation M which take input X(n) and generate the sequence of output:

$$\boldsymbol{y\ (n) = x\ (Mn)} \qquad \text{(II.20)}$$

Where M is integer.

In general we haven't a possibility of recuperation of X(n) of Yd(n) cause of loss of information.[24]

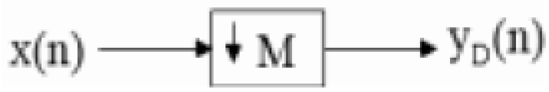


Figure 14 decimation device [24]

The operation of decimation in frequency domain and tomporel domain :

- **Decimation in tomprel domain** is characterized by the delation and elimination of some samples.

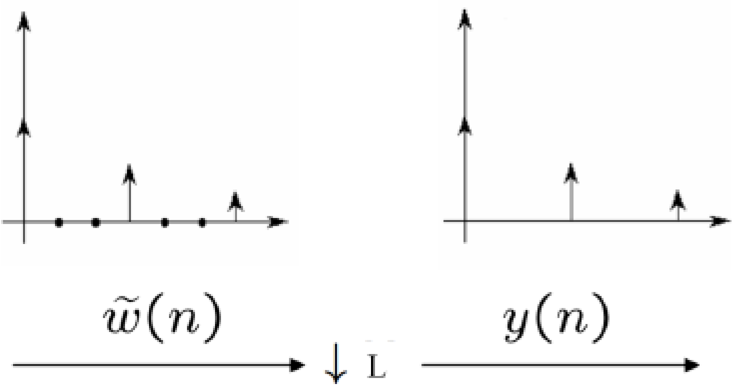


Figure 15 Downsampling by N=2.[25]

- **Decimation in frequency domain** presentation showing the aliasing effect produced because the highest frequency of the input signal is larger than pi/2.

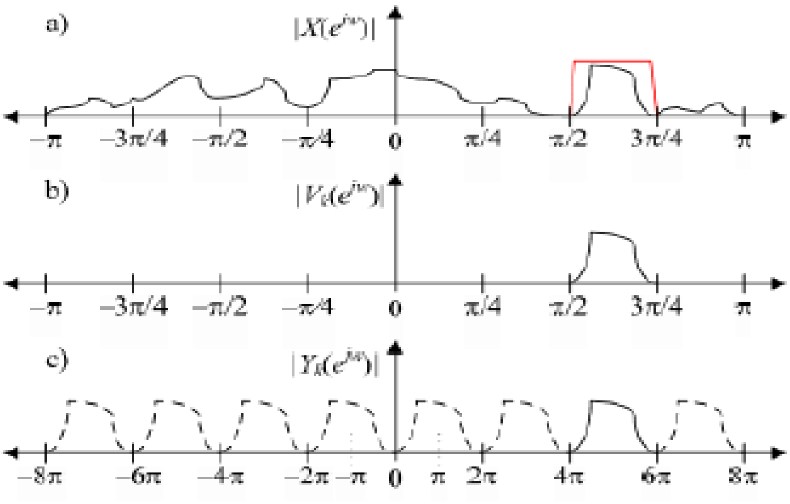


Figure 16 Decimation of a bandwidth signal with complex filters. a) spectrum of the input signal. b)Spectrum after anti-alias filtering bandwidth. c) resulting from the spectrum after subsampling by N = 8. [25]

Interpolation:

The oversampling increase the frequency of sampling by L and introduced L null samples between symbols with low sampling rate, the sequence in the SFB It consists of a sampler and an interpolation filter gk [m].

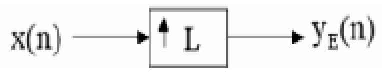


Figure 17 interpolation device.[25]

The function of interpolation in tomporel domain and frequency domain:

- **In tomporel domain** it adds zeros between samples:

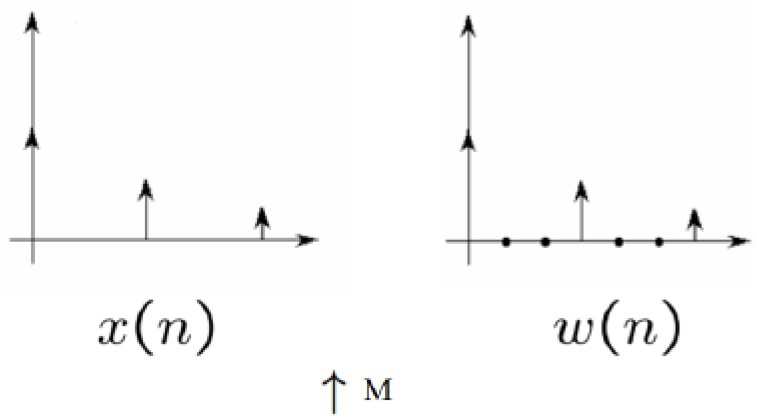


Figure 18 oversampling by N=2.[25]

- **In frequency domain**, the spectrum of the input signal (a) goes through oversampling it highlights the band of interest (b) and after interpolation by a bandpass filter which decides the choice of signal spectrum.

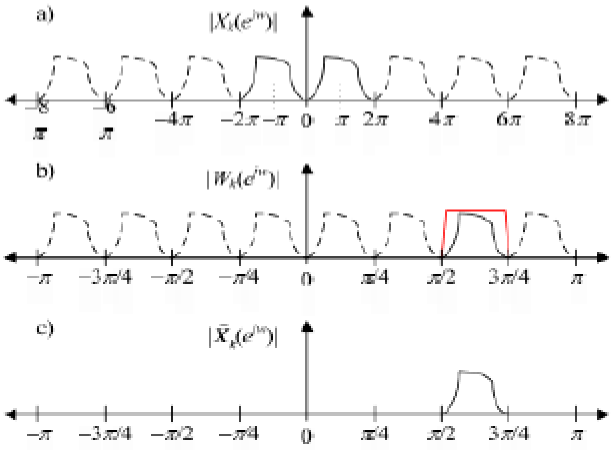


Figure 19 representation of interpolation steps in frequency domain.[25]

2. FBMC scheme:

The main idea of FBMC/OQAM is to replaced QAM modulation by OQAM modulation, for this an offset of half a T / 2 symbol period is introduced between the real part and the imaginary part of a given QAM symbol.

If the imaginary part is delayed by T / 2 on a subcarrier, it is the real part which will be delayed on the next subcarrier.[25]

The FBMC system has the same principle of OFDM system, but it uses prototypes filters.

In general way: the signal is transmitted into parallel signals, these signals are input into modulation mapping module (QAM or OQAM).

After modulation the signals goes through the IFFT block to convert the signal from the frequency domain to the time domain.

Than the signal goes through the polypahse filter block.

Finally, the remaining signals will be grouped parallel to the serial and sent on our channel.

At the receiver sit inverse transforms are used corresponding to the transmitter side separately.

Figure 20 presents the general structure of the FBMC / OQAM system.

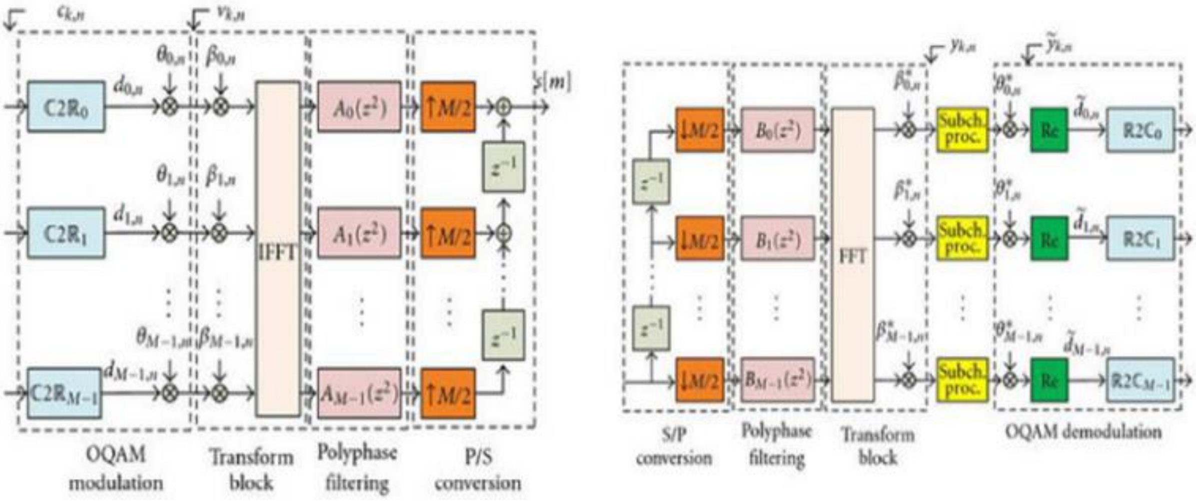


Figure 20: General structure of the FBMC / OQAM technique (transmission part / reception part).[25]

The OQAM post pre-processing block:

In this block, the OQAM symbols are transmitted rather than the QAM symbols.

To perform this modulation, post preprocessing blocks are placed respectively on the transmitter / receiver side.

The OQAM post pre-processing consists of two operations :

- **The first operation** is a complex / real conversion where the real and imaginary parts of a complex symbol Ck [l], transmitted at a rate 1 / T, are separated to form two new symbols.

➢ **The second operation** is a multiplexing is the multiplication by ϴk [n] in order to maintain the orthogonal symbols.

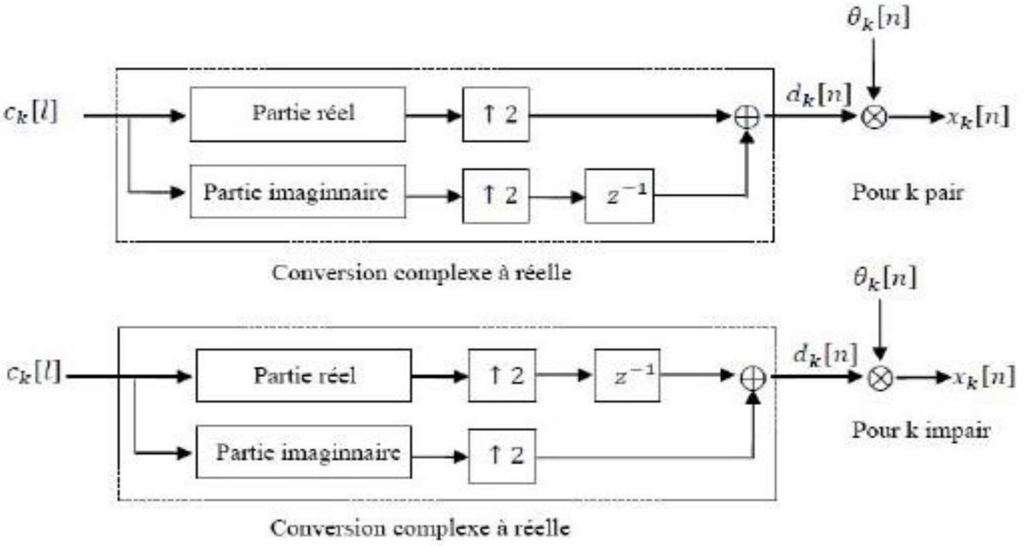


Figure 21 Post pre-processing OQAM.[25]

Post processing OQAM:

The post-processing OQAM is located in the reception section and consists of two main operations.

➢ **The first operation**: is a multiplication by the conjugate complex of ϴk [n] denoted **ϴk*[n]** followed by the operation which consists only of the real part.

➢ **The second operation**: is the real / complex conversion in which two successive real symbols form a symbol of complex value (one of the symbols is multiplied by j).

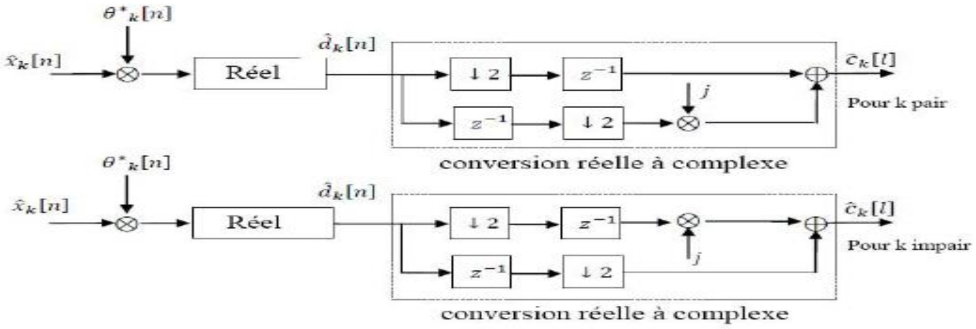


Figure 22 post processing OQAM.[25]

3. Merits of FBMC:

- One of merts of FBMC is tge impulse in time domain and frequency domain is well located.

- No cyclic prefix needed.

- Provide efficient spectrum and a more selective system.

- The separation of signals is done by filtering so, we don’t need of perfect synchronization between users.

- FBMC technique is less sensitive to frequency shift and Doppler Effect.

4. **Demerits:**

- allows the disadvantages that a problem poses is the great complexity of implementation from PPN.

- Provide robust narrowband jammers.

Conclusion:

In this chapter, the filter bank and its operation were explained, the FBMC modulation and after an example of a FBMC / OQAM modulation chain, we will simulate in the next chapter the two techniques of the theoretical part: FBMC modulation and OFDM modulation and comparison between them.

CHAPTER III

Simulation

Introduction:

This chapter is devoted to the results obtained during our simulations, we propose within the framework of this simulation the use of the MATLAB language.

In this work, we compare Spectral Density, Spectral Efficiency, BER of FBMC and OFDM and also Response Magnitude of filter of the FBMC and OFDM modulation techniques to analyze the merits of them.

Then we will establish the simulation of the CP-OFDM technique in a noisy channel with performance calculation in terms of BER = f (SNR). In presence of channel estimation and without it in order to show the effect of channel estimation.

Finally, we will compare between filters used in FBMC

MATLAB:

MATLAB, Matrix Laboratory is a technical computer language. It can be used for algorithm development, data analysis, visualization and calculation digital.

It is a tool for the manipulation of mathematical calculation.

An interactive system, its advantage is that it allows you to plot and therefore visualize the data in many ways.

Work with matrix algebra, polynomials and functions integration is very easy with MATLAB.
Programming calculation software facilitates problem solving and improves the learning process.

The resolution of complex numerical problems are solved easily and in a fraction of the time required with a programming language such as FORTRAN or what makes it therefore desirable.

MATLAB is discrete in nature, and therefore all inputs and outputs of MATLAB codes are discrete.

Parameters used for the evaluation:

a. Bit Error Rate (BER):

The bit error rate is the primary parameter describing the quality of digital transmission. It is defined as the ratio between the errored bits and the total number of bit received. This rate determines the number of errors that appeared before the modulation and just after the demodulation. It increases due to disturbances: faulty equipment or network, incorrect pointing of antenna, channel length, etc.

b. Signal to Noise Ratio (SNR):

This term refers to the demodulated signal perceived after the demodulation process. The total noise is that extracted from the transmission network plus the noise integrated into the modulation signal in the form of amplitude noise, phase noise and inter-symbol interference as well as other degradations of the modulation.

It is calculated using one or the other of the formulas shown below:

$$S/N\ (dB) = 10\log PS/PN$$
$$S/N = PS/PN$$

PS: Puissance of the Signal in (W)
PN: Puissance of Noise in (W)

c. Power spectral density:

A Power Spectral Density (PSD) is the measure of signal's power content versus frequency. A PSD is typically used to characterize broadband random signals. The amplitude of the PSD is normalized by the spectral resolution.

I. Comparative study between OFDM & FBMC :

➢ objective :

The objective of this simulation is to compare BER/SNR, Spectral Density, Spectral Efficiency of FBMC and OFDM and also Response Magnitude of filter of the FBMC and OFDM modulation techniques to analyze the merits of them.

a. Comparison of BER vs SNR :

✚ Parameters :

Proprieties	Values
FFT Length	512
Bits per sub carrier	4
OFDM	
Cyclic prefix length	43
FBMC	
Spreading factor	4

Table -1- parameters

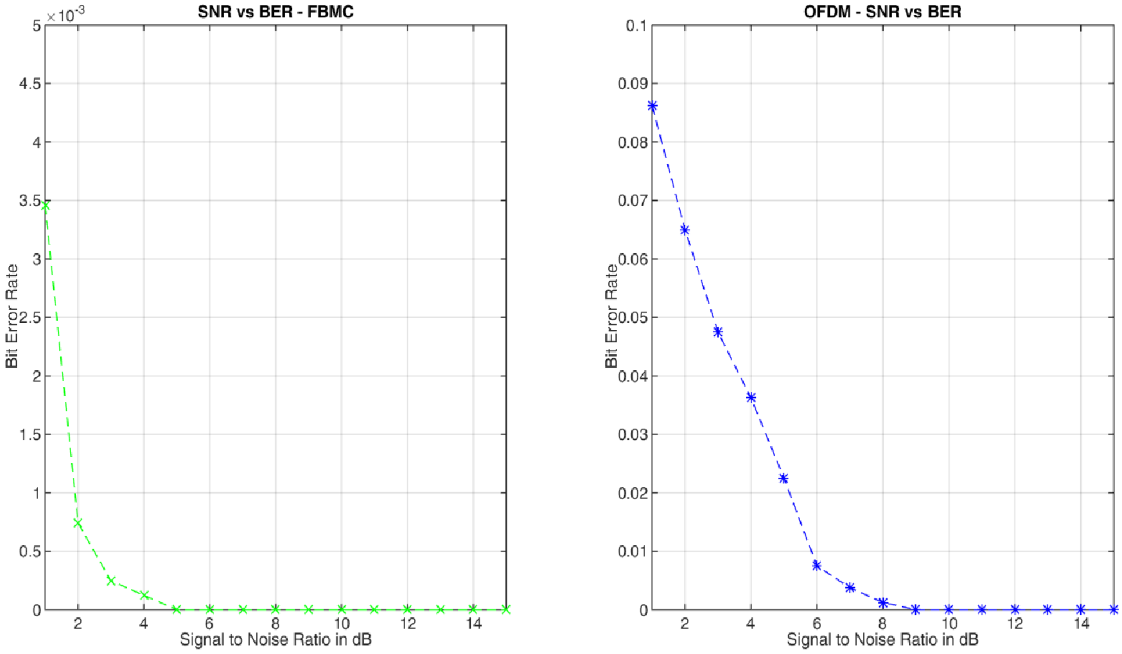


Figure-1- SNR vs BER of FBMC and OFDM

Discussion:

Large value BER indicates low quality of communication, while large value SNR indicates Better communication as the signal becomes stronger compared to noise.

In figure 1, the green curve represent SNR/BER of FBMC while blue that of OFDM.

Here we have the SNR, which varies from zero to 15db, and we calculate the BER. So in FBMC the highest value recorded BER= 3, 5*10.. Which is a very lower value compared to the highest value recorded OFDM which is BER ≈ 0.9.

Variation in the SNR affects the quality of the constellation. The simulation of BER vs SNR of OFDM was generated for SNR from zero to 8 dB we have BER= 0.

FBMC has the best performance compared to the other technique. It is closer to zero from 4 dB we have BER=0.

b. Comparision of spectral density:

The spectral density of FBMC over OFDM is compared. The simulation plots a graph with reference to the same.

The spectral density represents the strength of the signal over a time period it means, the possible bandwidth over which the bits can be sent successfully. A modulation's spectral density is efficient if the strength is closer to the normalized frequency.

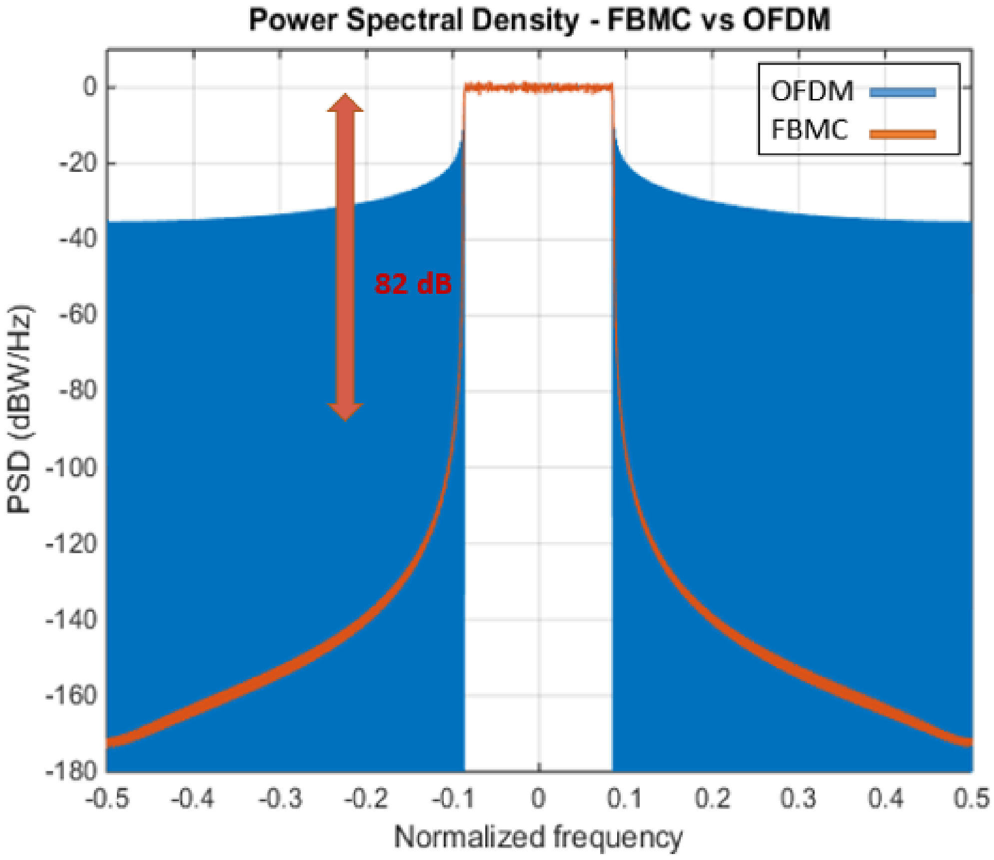


Figure-2- Power spectral Density FBMC vs OFDM

Disscusion :

In figure 2 , The red shaded region represent the spectral density of FBMC while blue that of the OFDM.

The power spectral density of the FBMC signal is plotted to highlight the low out of band leakage.

Comparing the plots of the spectral densities for OFDM and FBMC is grater than that of the OFDM. When the normalized frequency is 0.2 Hz , The PSD of FBMC is 82dB lower than that PSD of OFDM.

The FBMC has the spectral density closest to the normalized frequency when compared to OFDM modulation, This allows a higher utilisation of allocated spectrum, leading to increased spectral efficiency.

c. Comparison between spectral efficiency:

The spectral efficiency / spectrum efficiency or bandwidth efficiency refers to the information rate that can be transmitted over a given bandwidth in a specific communication system.

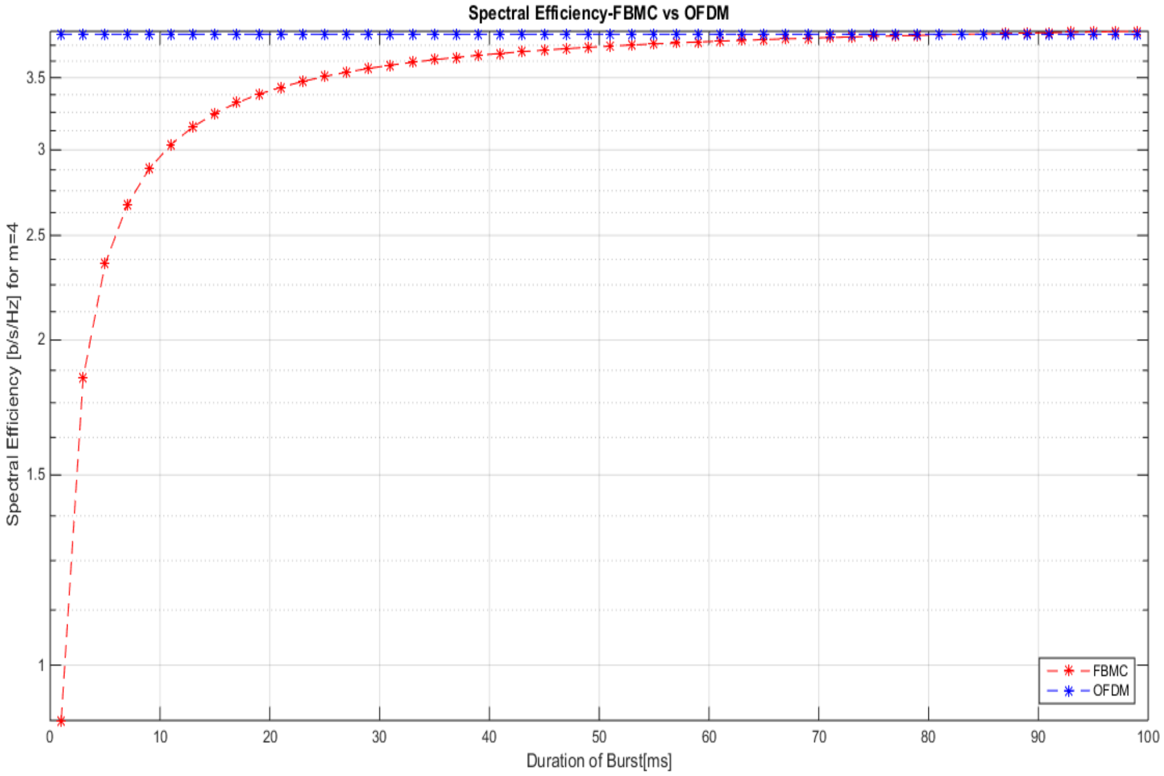


Figure-3.a- Spectral efficiency FBMC vs OFDM

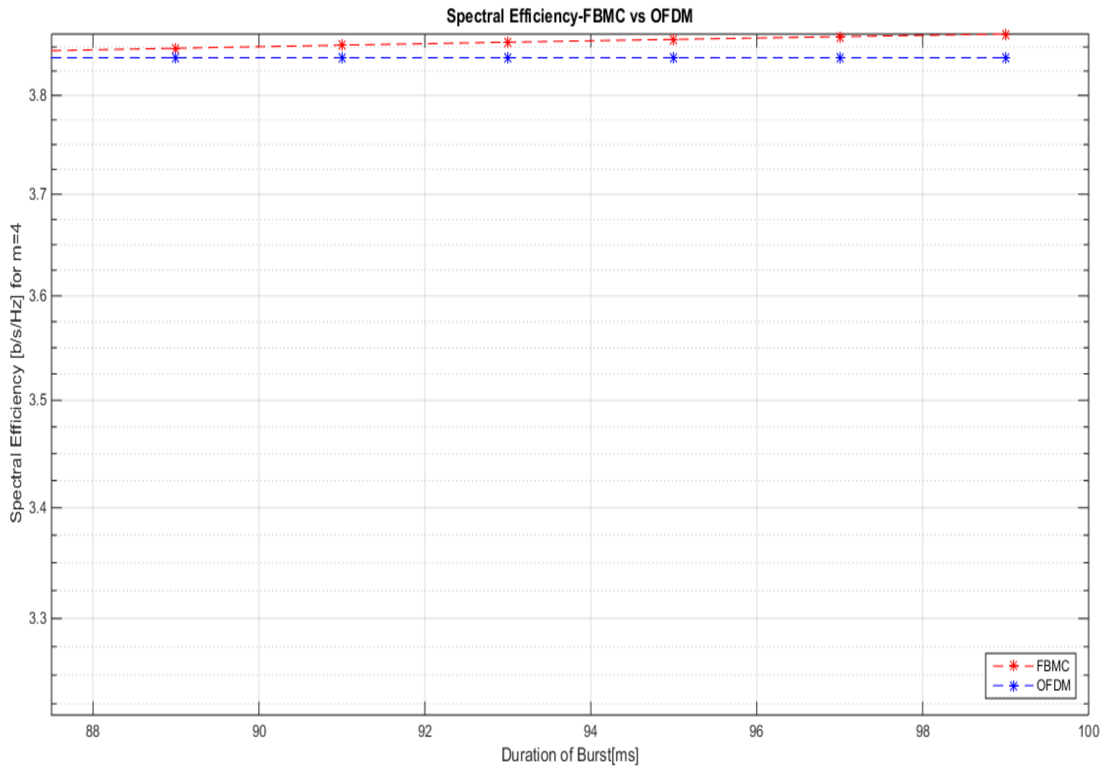


Figure-3.b- Spectral efficiency FBMC vs OFDM

Discussion:

The graph denotes the spectral efficiency of the two, OFDM and FBMC. The graph is generated by varying the duration of burst from zero to 100ms. Since the no. of cyclic prefix and the filter length are equal.

In figure 3, the red curve represent spectral efficiency of FBMC while blue that of OFDM.

From the figure -3.a- It’s observed that the FBMC‘s spectral efficiency increases with the increase in duration of bursts. It is greater than OFDM if duration of bursts is larger.

From figure -3.b-When the duration of burst is longer than 90ms, the spectral efficiency of FBMC exceeds that of OFDM. So, this allows a higher information rate than can be transmitted over the bandwidth.

d. Comparison between the FBMC and OFDM filters:

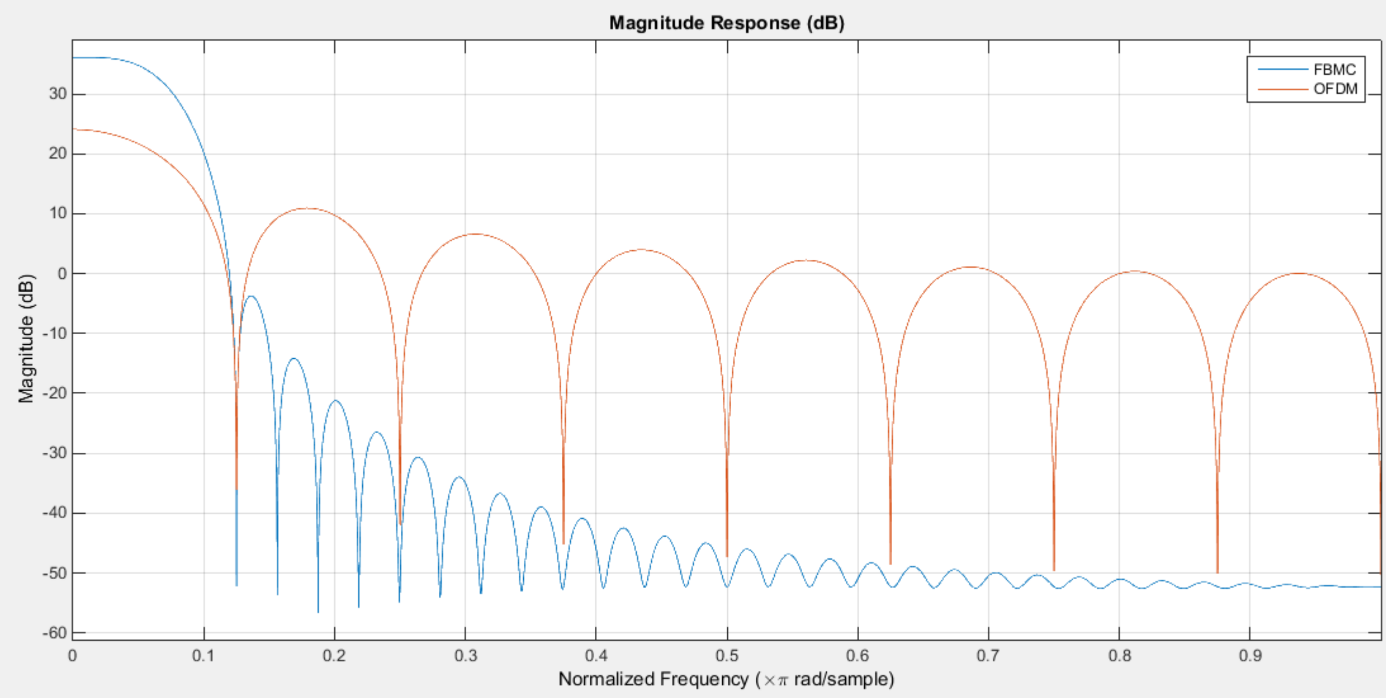


Figure -4- Response Magnitude of filter of the FBMC and OFDM

Discussion:

One of the differences between OFDM and FBMC modulations lies in the property spectral leakage, as shown in Figure of the OFDM represented by filter response in blue and The FBMC represented by the filter response in red.

We can observed that the OFDM modulation has large lateral lobes, which impose strict orthogonality constraints for all subcarriers. On the contrary, the modulation FBMC has negligible side lobes in the frequency domain.

With a spectral leak very limited, high resolution spectral analysis and low interference on the adjacent frequency bands can be achieved.

II. Simulation of technique of modulation OFDM and calculate of BER=f(SNR):

- **Objective** :

The objective of this study is to demonstrate the effectiveness of channel estimation with the effect of multipath fading.

Study of a set of cases by changing parameters from SNR to BER for the cases of BPSK and 16QAM modulations and in two cases without channel estimation and with channel estimation.

- **Simulation environnement :**

Propreties	**Valus**
FFT length	64
Size of cyclic prefix	16
Number of channel taps	8
Modulation methods	BPSK – 16QAM
Channel estimation method	LS
Type of image	PNG
Size of image	144 ko
Resolution of image	512x512

Table-2- simulation environnement

- **Study of different cases for OFDM modulation :**

BPSK :

- 1 bit ⟶ 1 symbol
- No channel estimation
- 10dB SNR
- 0.5 BER

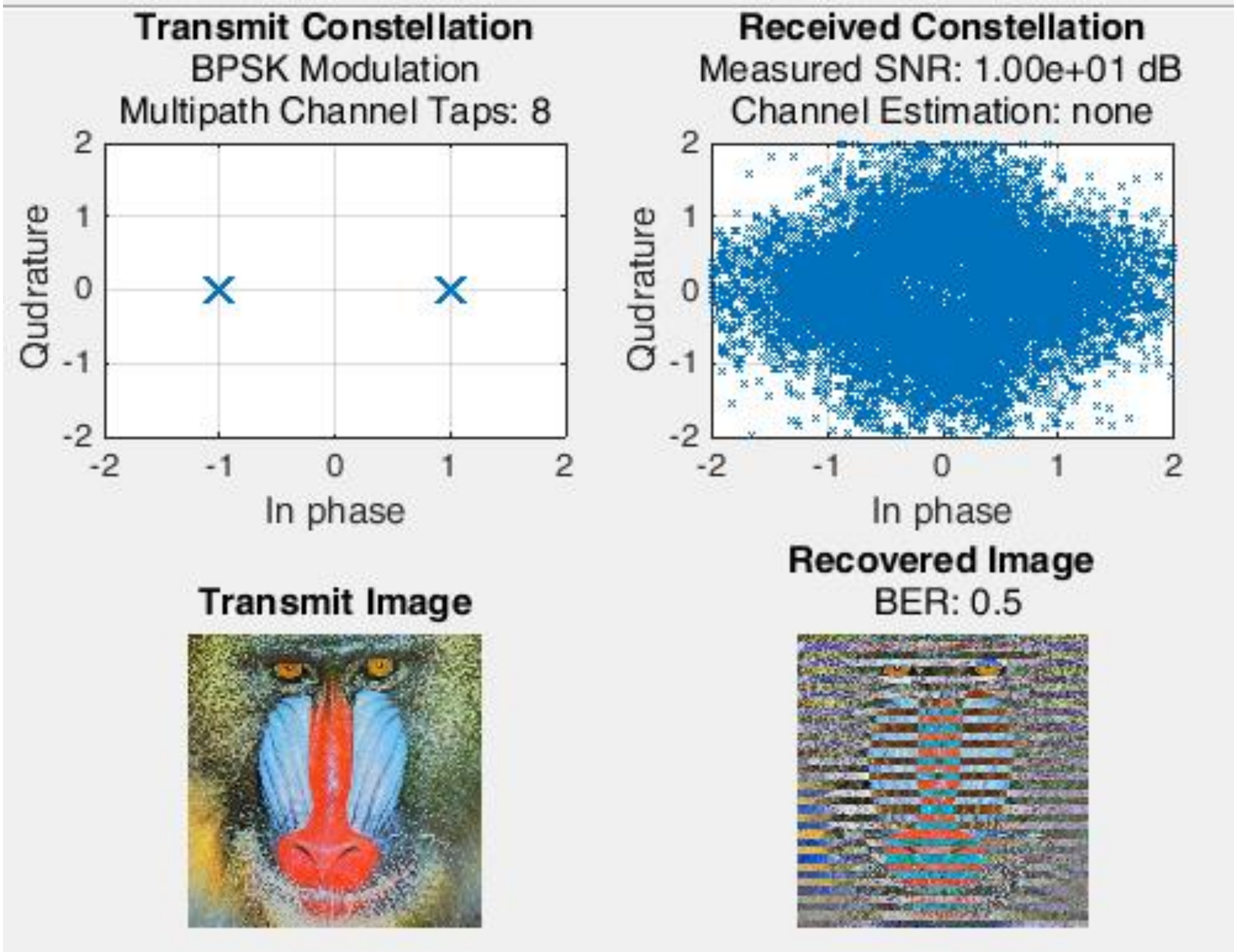


Figure -5- Results of simulation of modulation BPSK with SNR=10 Without using channel estimation

- ✓ Shown here is the received constellation and image for BPSK in a fading channel without channel estimation.

We look at BPSK consists of two points on the phase axis opposite the quadrature axis, and maps one bit to one symbol. And for 10dB SNR we achieve a bit error rate.

BPSK :

- 1 bit ⟶ 1 symbol
- Channel Estimation: LS
- 10dB SNR
- 0.0069 BER

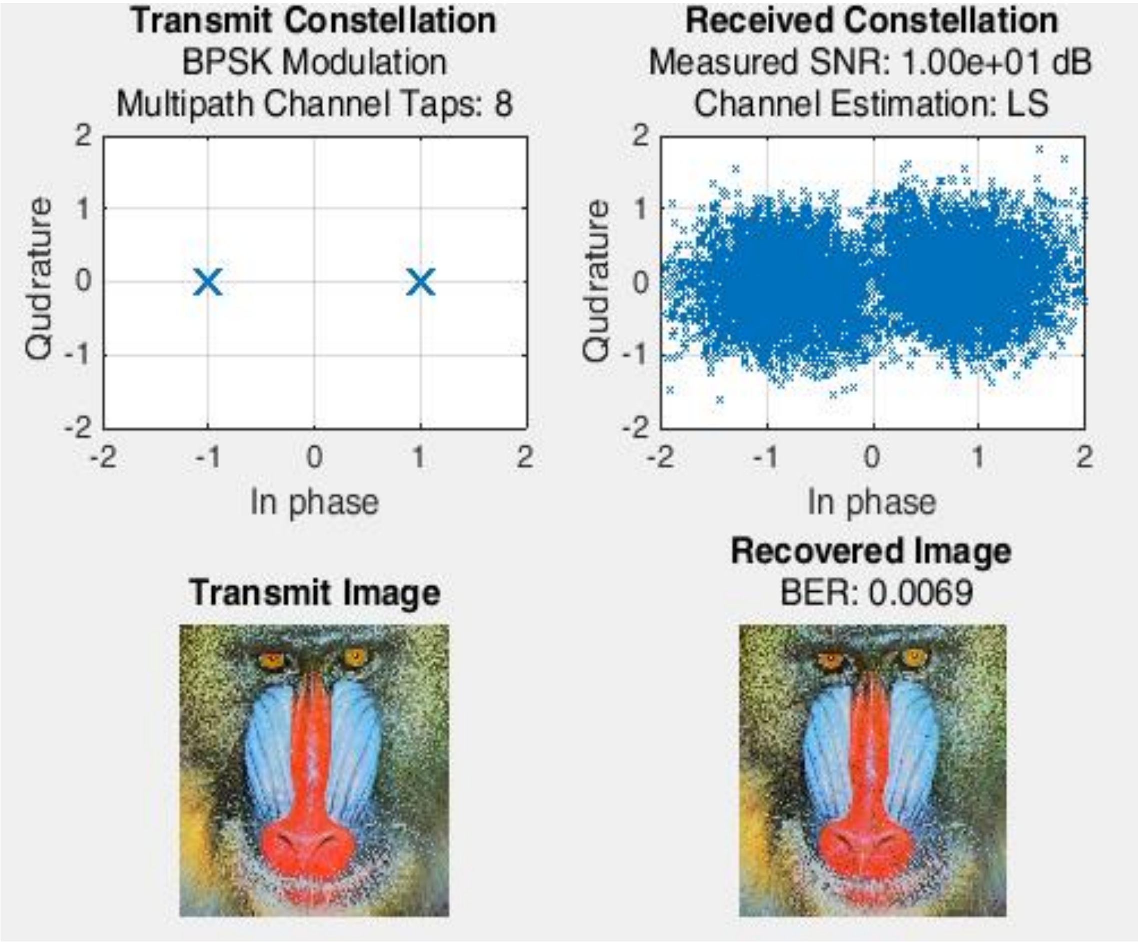


Figure -6- Result of modulation BPSK with SNR=10 using channel estimation LS

- ✓ By applying estimation, we can now recognize the image

BPSK :

- 1 bit ⟶ 1 symbol
- No channel estimation
- 20dB SNR
- 0.5 BER

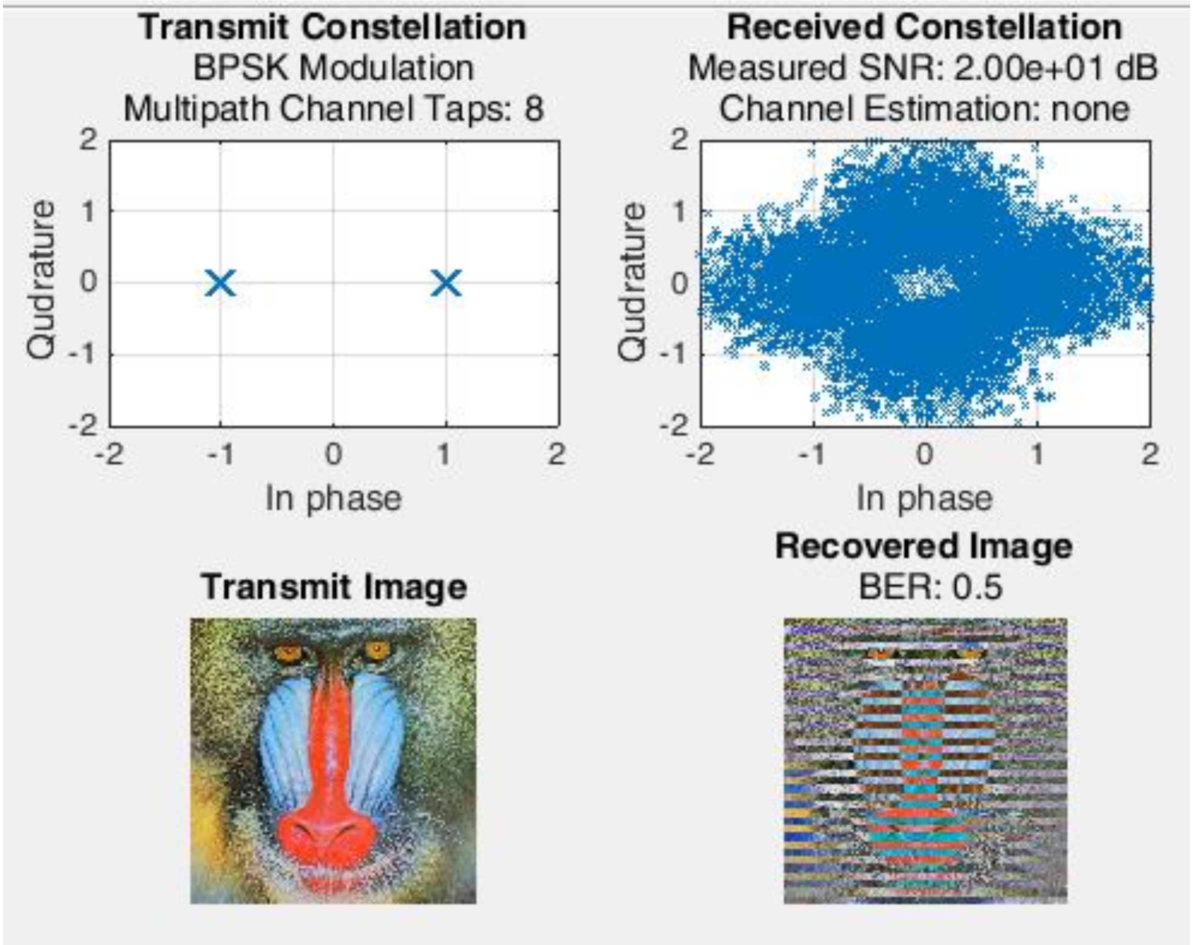


Figure-7- Results of simulation of modulation BPSK with SNR=20 Without using channel estimation

BPSK :

- 1 bit → 1 symbol
- Channel Estimation: LS
- 20dB SNR
- 0.00051 BER

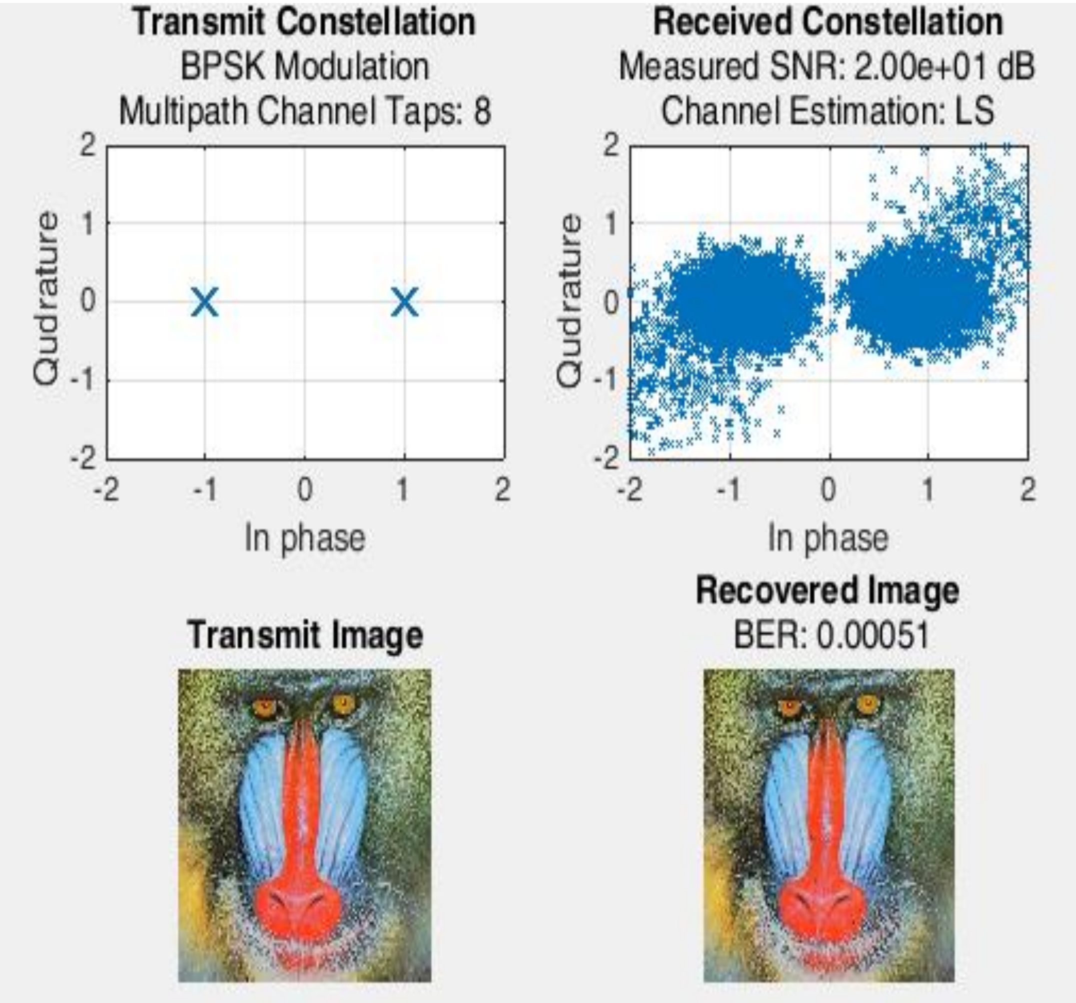


Figure -8- Results de la simulation de modulation BPSK avec un SNR=20 Using channel estimation

We now examine 16 QAM in the fading channel with no estimation

16QAM :

- 1 bit ⟶ 1 symbol
- No channel estimation
- 10dB SNR
- 0.5 BER

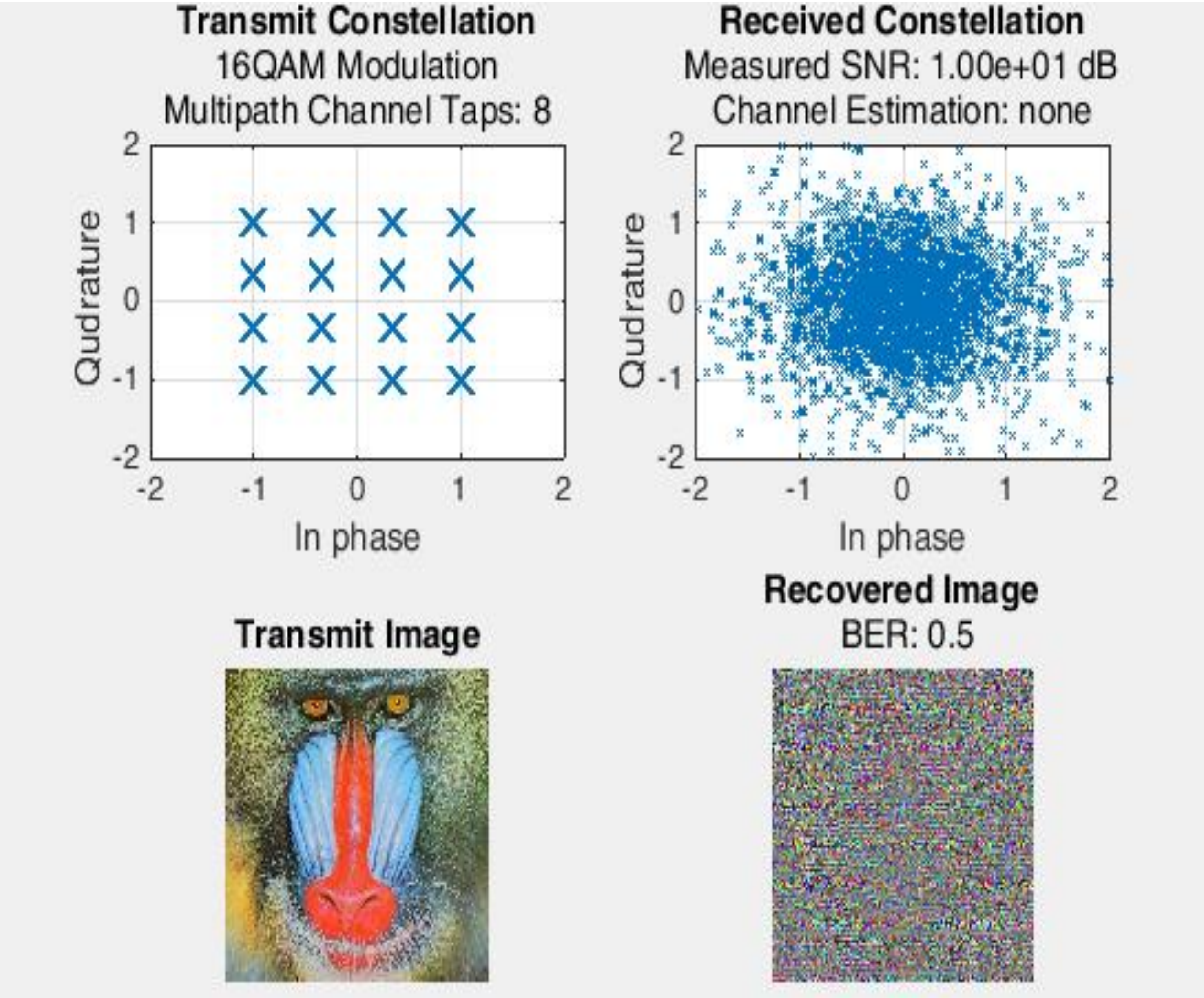


Figure -9- Results de la simulation de modulation 16QAM avec un SNR=10 Without using channel estimation.

- ✓ The recovered image is unrecognizable although BER is still relatively high.

.

16 QAM :

- 1 bit → 1 symbol
- Channel estimation: LS
- 10dB SNR
  0.31 BER

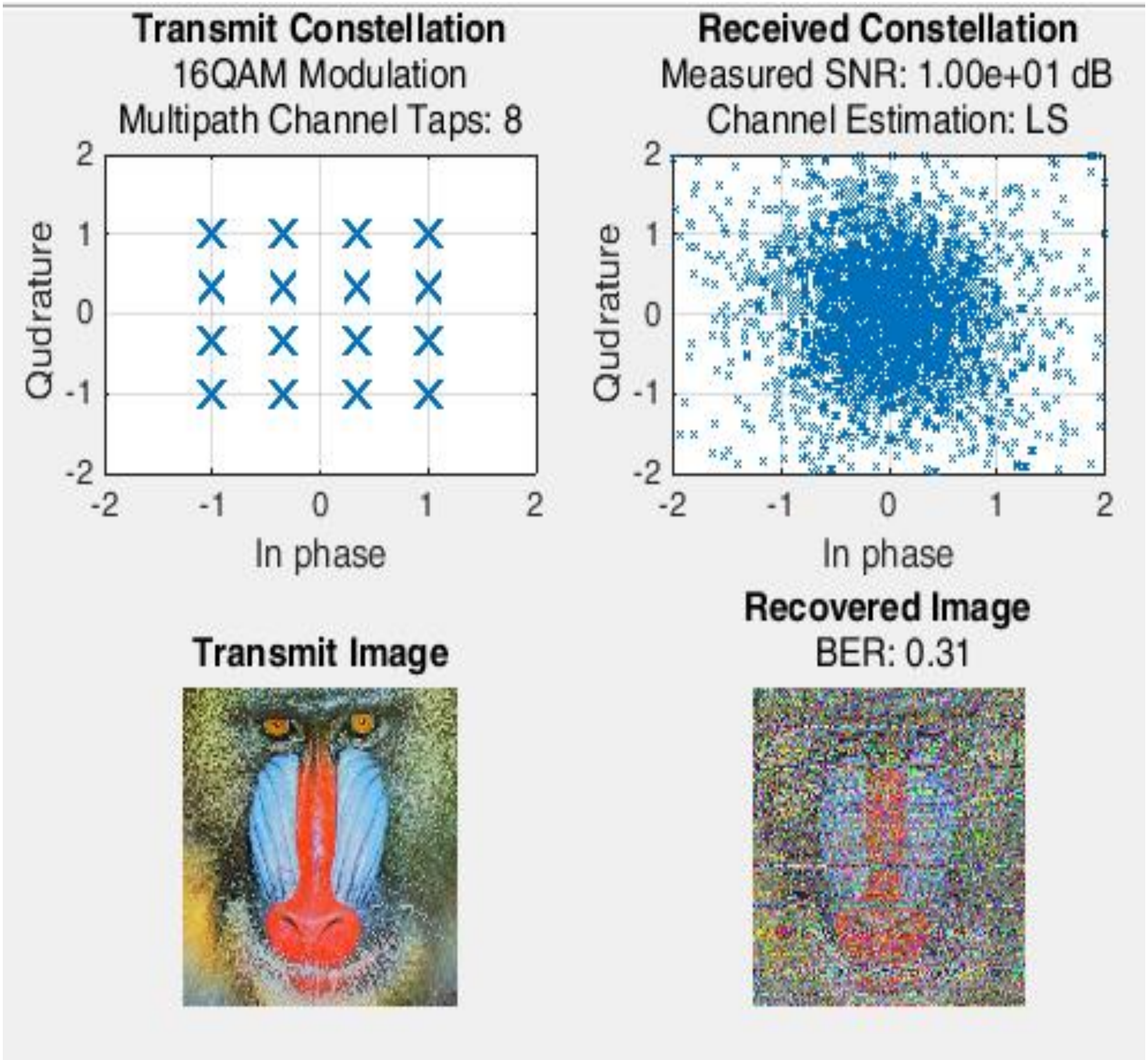


Figure -10- Results of simulation of modulation 16QAM with SNR=10 Using channel estimation

Discussion :

From the simulation results, the figures represent a constellation for BPSK and 16QAM modulations in transmission and receiving in the presence of a multi-path channel and AWGN noise. Since the noise generated acts on the signal, it has a direct effect on the constellation as shown in the figures.

Note that the points in the constellation are no longer fixed in specific coordinates but in an interval. This happens because the added noise distorts the signal.

We also notice a direct effect on the image in the form of spots and starting from the variations of the SNR values in little noticing the change of our image.

Through these results, we also observe that various digital modulation schemes are effective when paired with OFDM we specifically looked at BPSK and 16 QAM all showing robust performance to noise and inter symbol interference.

When simulated with multipath fading channel estimation was necessary to produce acceptable results.

In general we see that channel estimation methods trade-off accuracy with computational complexity.

III. Comparison between filters used in FBMC:

- **Magnitude response:**

The frequency response of the filter is a complex function whose magnitude gives as a function whose magnitude gives the gain of the system.

So the magnitude as a function of frequency shows which frequencies are attenuated and which are not, which is one of the most important proprieties of the filter.

The following figure presents the results of the simulation showing the performance of the FBMC technique by varying the order of the prototype filter. We express these performances in terms of response magnitude.

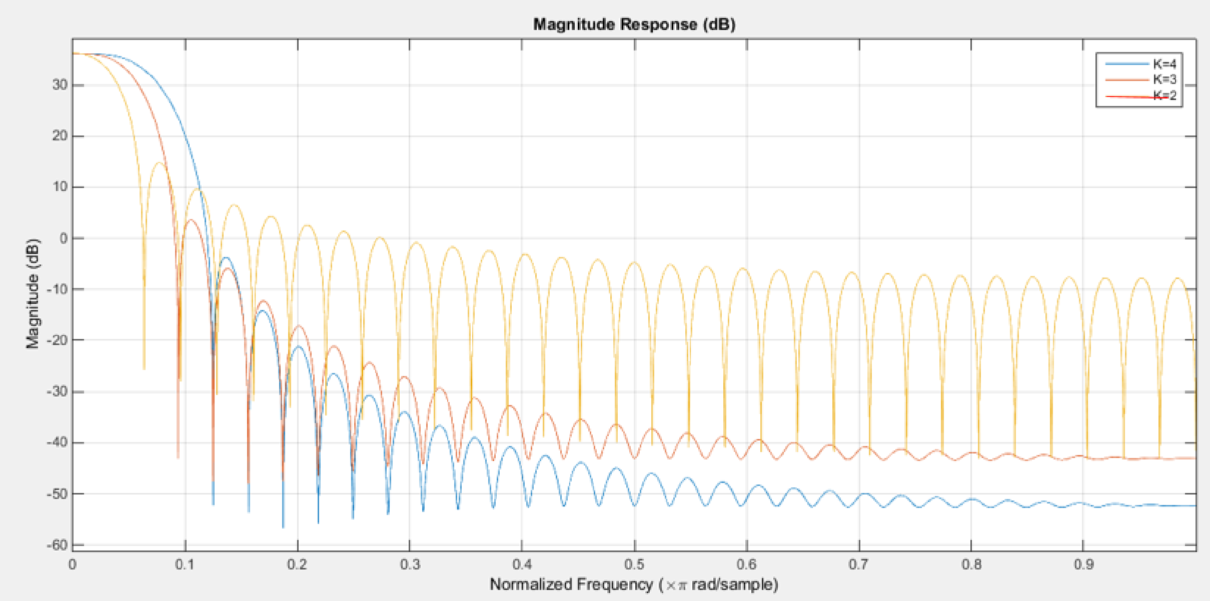


Figure -11- magnitude response of prototype filter for, k=2, k=3, k=4

Discussion:

The Filter in blue represent the prototype filter of factor of overlap K=2, the filter in red represent the prototype filter of factor of overlap K=3, and the filter in yellow represent the prototype filter of factor of overlap k=4.

We notice that the most efficient filter which is attenuated for k=4 in the temporal domain and similarly in the frequency domain because the appearance of the filter window decreases almost vertically, the ripples at the edge diminishes in a finite manner.

Conclusion:

The first simulation results show that FBMC has the lower BER, greater Spectral Efficiency (if the burst is larger) and better performance of Spectral Density compared to OFDM modulation.

From the second simulation that we carried out made it possible to highlight a CP-OFDM modulation and demodulation from a BER point of view, as well as the transmission chain parameters (the choice of constellation and SNR). So we can conclude that the OFDM link, which uses QPSK modulation, gives us better link performance in BER function compared to 16 QAM modulation.
This simulation also allowed us to see the effect of channel estimation in the presence of multipath fading.

And from the last simulation results, allow us that the FBMC system is more efficient when the order of the prototype filter is increased (for K = 4).

From the third simulation results, allow filters and us that FBMC has best performance compared to OFDM technique from a point of view BER.

General conclusion

The demand for new wireless services and applications as well as the number of users is increasing rapidly, which imposes flow constraints.

Among the solutions studied to meet these flow constraints, there are multi modulations carriers: OFDM and FBMC modulation.

The main objective of this work is to develop Comparative Simulation of FBMC and OFDM techniques for 5G networks in order to increase the data rate and quality in 5G communication.

The MATLAB program simulation made it possible to highlight the interests of an OFDM modulation from a point of view of the parameters of the primordial transmission chain (number of carriers, choice of constellation, and types different modulation) to be adjusted according to the channel and the data to be transmitted.

The BER of a digital communication system is an important figure of merit used to quantify the integrity of the data transmitted by the system. The criterion is to compare the variation of BER for the OFDM link using different modulation using channel estimation and without it in presence of multipath fading in order to show the necessity of channel estimation.

By comparing, using simulations, the performance in terms of bit error rate of the system described using the different types of subcarrier modulation, of the different channels, we found that with QAM modulation, the degradations system performance becomes important compared to the OFDM-QPSK structure. Our study highlights the importance of the OFDM technique in wireless networks.

In this work at the end of the study, the performance comparison between the most used multi-carrier modulation technique (OFDM) and a less known technique (FBMC) is carried out in terms of their operating principle.

Simulation results show that FBMC provides an overall performance improvement over conventional OFDM for all parameters considered, proving FBMC as an ideal candidate for the future development of wireless communications

Channel estimation can be performed by many ways: either inserting pilot tones into all of the subcarriers of OFDM symbols with a specific period or inserting pilot tones into each OFDM symbol. In this memory, we explored these two pilot arrangements in detail for different Doppler frequencies. Channel estimation based on block type pilot arrangement is presented, and it is shown that this type of arrangement performs better, when the channel is changing slowly. Channel estimation based on comb type pilot arrangement was presented by giving channel estimation method at pilot frequencies and interpolation of channel at data frequencies.

Multicarrier modulations and in particular OFDM have been used during this last decade for a wide range of applications. We can notably cite ADSL for a wired context, Wi-Fi, LTE (4G), 5G NR for wireless systems and for FBMC.

We can foresee that the development of OFDM which is FBMC prelude to the opening of new markets for different types of applications, in different environments, these are the new technologies of the 5th generation of wireless telecommunications.

we would like in future work to broaden our study on the FBMC technique to highlight another important factor in terms of the study of the spectral mask and not taken into account here in order to carry out a study based on this new technique because, This technique creates a new window on development in the field of wireless communication.

Referances

Referances:

[1]: S. Kazimierz & B.Yasman, (2007).Radio wave propagation and antennas for personal: third edition

[2]: Design of wireless communication systems –issues in synchronization, channel estimation and multi-carrier systems

[3]: M. Beach, "Propagation and System Aspects," *University of Bristol*, April

1994.

[4]: A.Kamran, Channel estimation in OFDM systems, Thesis, KING FAHD UNIVERSITY OF PETROLEUM AND MINERALS, 2002

[5] : Alaa CHOUMANE, Synthèse d'un canal de propagation par système multi-antennes pour la caractérisation de terminaux mobiles à diversité, Thesis, UNIVERSITE DE LIMOGES, 2011

[6] : HAMOU CHEHRI, ÉTUDE ET CARACTÉRISATION D'UN CANAL DE PROPAGATION POUR LES RÉSEAUX V ANET, MÉMOIRE, UNIVERSITÉ DU QUÉBEC EN ABITIBI-TÉMISCAMINGUE.

I. **[7]:** Karasawa, Y. Statistical multipath propagation modeling for broadband wireless systems. // IEICE Trans. Commun. E-90- B(2007), pp. 468-484.

[8]: MERAH Hocine, Conception d'un MODEM de la quatrième génération (4G) des réseaux de mobiles à base de la technologie MC-CDMA, MEMOIRE de magistère, UNIVERSITE FERHAT ABBAS – SETIF, 2012

[9] : J. Proakis, *Digital Communications*. Prentice-Hall, 3rd ed., 1995

[10] : Arnaud Massiani, "Prototypage de Systèmes Haut Débit combinant étalement de Spectre, Multi-porteuses et Multi- antennes", Thèse de Doctorat, Institut National des Sciences Appliquées, Rennes, Novembre 2005.

[11] : R. broadcasting systems, "Digital Audio Broadcasting (DAB) to mobile, portable and fixed receivers," European Telecommunications Standards Institute, February 1995.

[12] : J. Proakis, Digital Communications. Prentice-Hall, 3rd ed., 1995.

[13] : BV. Engles and H. Rohling, "Multilevel differential modulation techniques for multicarrier transmission systems," Eur. Trans. Telecommun. Reh. Technol., vol. 6(6), pp. 633–640, November 1995.

[14] : Yong Soo Cho, Jaekwon Kim, Won Young Yang, Chung G. Kang, 2010, MIMO-OFDM WIRELESS COMMUNICATIONS WITH MATLAB, IEEE PRESS

[15]: Lunsheng Xue, Shangfei Qiu,Peng Wuand Dejiang Chen, "An Improved Interference CancellationChannel Estimation Method forOQAM/OFDM System", IOP Conf. Series: journal of physics: Conf. Series 1169 (2019) 012053

[16]: Chunling Hao, Shaochuan Wu, Xiaoqing Liu, Kangjian Ma, "A New Type of Comb Pilot in FBMC/OQAM", 2018 IEEE Wireless Communications and Networking Conference (WCNC), 17841871, 2018**.**

[17]: J. Rinne and M. Renfors, "Pilot Spacing in OFDM systems in practical channels," IEEE Trans. on Consumer Electronics, vol. 12, pp. 959–962, November 1996.

[18]: C. W. Farrow, "A contiously variable digital delay element," IEICE Int. Symp Circuits and Systems, pp. 2641–2645, June 1988

[19]: Sinem Coleri, Mustafa Ergen, Anuj Puri and Ahmad Bahai, "Channel Estimation Techniques based on Pilot Arrangement in OFDM systems," IEEE Transactions on Broadcasting, September 2002.

[20]: H. Sari, G. Karam, and I. Jeanclaude, "Transmission Techniques for Digital Terrestrial TV Broadcasting," IEEE Communications Magazine,, pp. 100–109, February 1995.

[21]: Bing-Leung Patrick Cheung, Simulation of Adaptive Array Algorithms for OFDM and Adaptive Vector OFDM Systems. PhD thesis, Blacksburg, Virginia, September 2002.

[22]: Bingham, J.A.C. (1990) Multi-carrier modulation for data transmission: an idea whose time has come. IEEE Commun. Mag., 28(5), 17–25.

[23] : G.A. Franco and G. Lachs, « An orthogonal coding technique for Communications», IRE Intern. Conv. Rec., Vol. 9, pp. 126-133, 1961.

[24]: Tobias Hidalgo Stitz Filter Bank Techniques for the Physical Layer in Wireless, on the 29 of October 2010, at 12 noon. Tampere enteknillinenyliopisto - Tampere University of Technology Tampere 2010

[25]: DAOUD Khedidja, SIMULATION COMPARATIVE DES TECHNIQUES FBMC ET OFDM POUR LES RESEAUX 5G, Mémoire, université Abou Baker Belkaid