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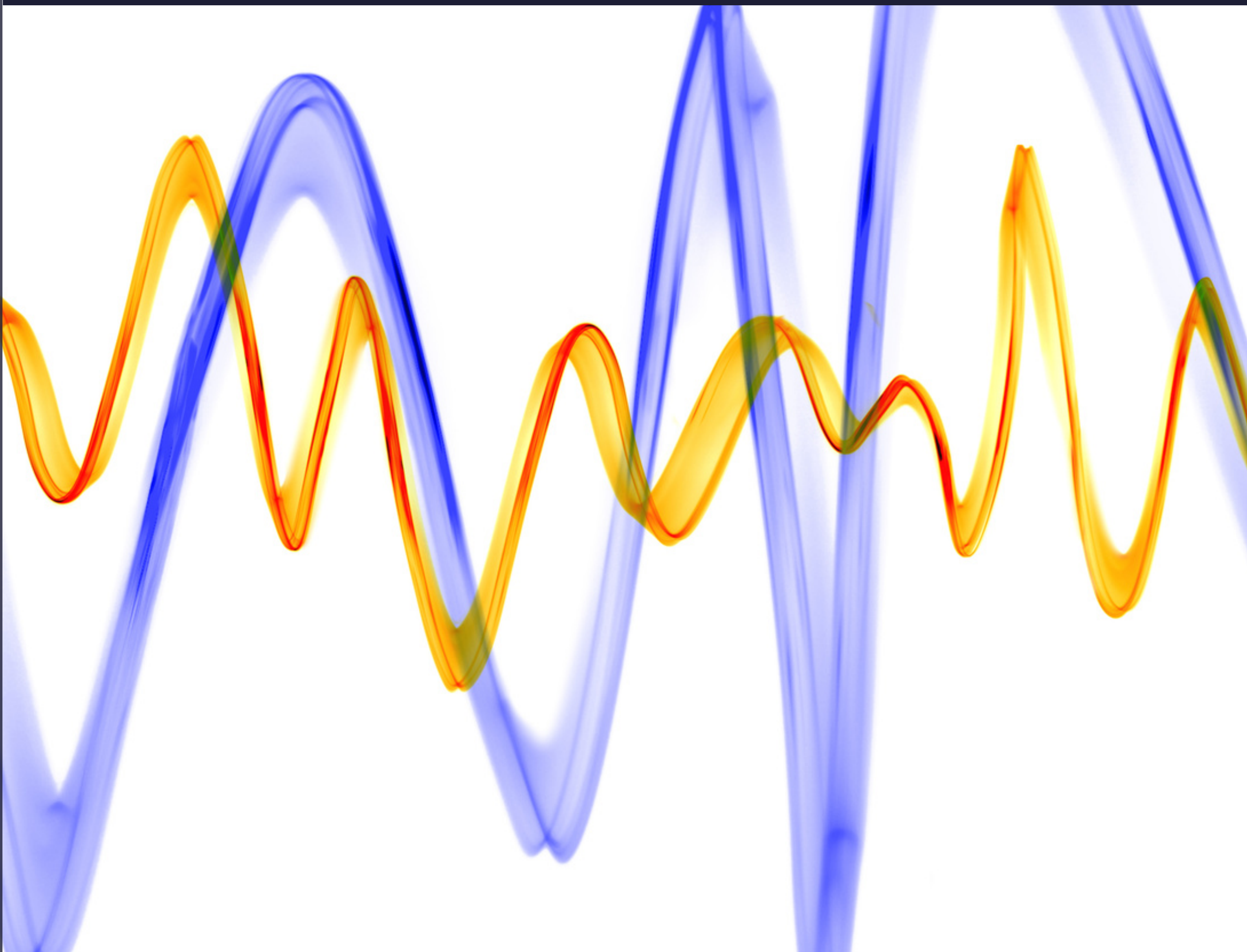
PAPR REDUCTION OF OFDM SYSTEM BY CONVERT IN-BAND SIGNALS DISTORTION TO IMPULSE NOISE (ERROR)



Khalid Awaad Humood 1
Qahtan Khalaf Omran 2
Hayder Khaleel AL-Qaysi 3
Tahreer Mahmood 4

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Authored By

Khalid Awaad Humood[1]

Qahtan Khalaf Omran[2]

Hayder Khaleel AL-Qaysi[3]

Tahreer Mahmood[4]

[1,2,3,4] Department of Electronic Engineering

College of Engineering

University of Diyala

Iraq

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Abstract

OFDM systems suffer from the problem of high peak to average power ratio (PAPR). Amplitude clipping is one of the most techniques used to reduce PAPR, this technique does not demand side information; therefore, no reduction in the system's data throughput. However, it leads to additional distortion (in-band signal distortion and out-of-band radiation). To overcome this problem, low complexity non-distortion clipping technique is proposed for OFDM system, the main concepts of this proposed method is how to convert the generated distortion (in-band and out-of-band signals distortion) when using clipping at transmitter to impulse noise (error) which is possible easily be covered by using a simple coding technique to cover error at the receiver. the proposed method does not clipping the signal in time domain, the clipping use for discrete samples directly after IFFT. Simulation outcome detect that the proposed non-distortion clipping technique provides an efficient reduction in PAPR, best performance compared with conventional clipping technique, and less cost and complexity.

List of Abbreviations

Abbreviation	Definition
AWGN	Additive White Gaussian Noise
BER	Bit Error Rate
BPSK	Binary Phase Shift Keying
CCLs	Cross Correlation levels
CDMA	Code Division Multiple Access
CP	Cyclic Prefix
DFE	Decision Feedback Equalizer
DFT	Discrete Fourier Transform
D-BPSK	Differential Binary Phase Shift Keying
DS	Direct Sequence
EGC	Equal Gain Combining
FDMA	Frequency Division Multiple Access
FFT	Fast Fourier Transform
FH	Frequency Hoping
F-FH	Fast Frequency Hoping
FIR	Finite Impulse Response
FM	Frequency Modulation
3G	Third Generation
GMC	General Multi-Carrier
GO	Group Orthogonal
GSM	Global System of Mobile
IBI	Inter-Block Interference
ICI	Inter-Carrier Interference
IFFT	Inverse Fast Fourier Transform
ISI	Inter-Symbol Interference
LOS	Line Of Sight
LPF	Low Pass Filter

LSI	Large Scale Integration
MAI	Multiple Access Interferance
MC	Multi-Carrier
MCM	Multi-Carrier Modulation
MF	Match Filter
MMSE	Minimum Mean Square Error
M-OFDM	Multiuser Orthogonal Frequency Division Multiplexing
MPIC	Multistage Parallel Interference Cancellation
MRC	Maximum Ratio Combining
MSIC	Multistage Successive Interference Cancellation
MSK	Minimum Shift Keying
M-Sequence	Maximum length Sequence
MT	Multi-Tone
MUD	Multi-User Detection
MUI	Multi-User Interference
NBI	Narrow Band Interference
OFDM	Orthogonal Frequency Division Multiplexing
PAP	Peak Average Power
PBI	Partial Band Interference
PIC	Parallel Interference Cancellation
PN	Pseudo-random Noise
P/S	Parallel to Serial converter
PSK	Phase Shift Keying
QoS	Quality of Service
RF	Radio Frequency
RMS	Root Mean Square
SD	Selective Diversity
S-FH	Slow Frequency Hoping
SIC	Successive Interference Cancellation
SNR	Signal to Noise Ratio
S/P	Serial to Parallel converter
SS	Spread Spectrum
SUD	Single User Detection
TDMA	Time Division Multiple Access
TH	Time Hoping
VLSI	Very Large-Scale Integration

List of Symbols

Symbol	Definition
A	Amplitude of fade signal
B_D	Doppler spread
BW	Band width
C	Speed of light
C_n	Spreading code matrix
D	Diagonal matrix
E	Expected value
$F^{(s)}$	Partial cancellation factor
f_c	Carrier frequency
f_d	Doppler frequency
f_i	FFT matrix column
F_{Nc}	FFT matrix
g(x)	Generator polynomial
G_p	Process gain
G_{MC}	Process gain for MC-CDMA
H_k	Hadamard matrix
h(t)	Fading channel impulse response
$I_0(.)$	Bessel function of zero order kind one
K	Spreading code length
K	Rice factor
L	Number of baths for fading channel
m	Number of users
Na	Number of active users
Nc	Number of subcarriers
Ng	Number of groups
n(t)	Noise vector
Q	Number of supcarriers per groups
R	Cross correlation matrix
r(t)	Base band received signal
s(t)	Transmitted signal
T	Transformation matrix
T_c	Chip time
T_m	Multipath delay spread
T_h	Hoping duration

T_s	Symbol duration
v	Speed difference between source and transmitter
W	Transmitted signal bandwidth
x	Input signal
y	Output signal
*	Complex conjugate
^{T}	Matrix transpose
τ	Delay time
σ	Noise power
θ	Phase of fade signal
$(\Delta f)_c$	Coherence bandwidth
H	Conjugate transpose

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Chapter One

Introduction

1.1 Introduction

Nowadays, third generation (3G) mobile communication systems have become popular all around the world. However, its services cannot provide a very hig dynamic range of datu rates. nor can it meet the requirements of a variety of business types. Besides, voice transportation in 3G still relies on circuit switching technology, which is the same method as used in second-generation (2G) communication systems, rather than pure Internet Protocol (IP) approach. Thus, based on consideration fisted above, y countries have already carried out research on the next completely evolutionary fourth generation (4G) communication systems which provide a comprehensive and secure IP solution where voice, data, and multimedia can be offered to users at "anytime, anywhere" with higher data rates than previous generations [1). Since bandwidth resource in 4G mobile communications is still scarce, in order to improve spectrum efficiency and achieve as high as 10NOMbps wireless transmission rate, it requires more advanced techniques to be employed. The limitation of modulation schemes in existing communication systems has become an obstruction in further increasing the data rate. Hence, next generation mobile communication systems need more sophisticated modulation scheme and information transmission structure.

During the last few decades, growth rate of wireless technology has been accelerated to such a level that it has become ubiquitous. Progress in fiber-optics with assurance of almost limitless bandwidth and predictions of universal high-speed wireless internet access in the not-too-distant future thrive in both the popular press and technical journals

.[2] Wireless communication is having the fastest growth phase in history because of unprecedented evolution in the field. The kid of wireless communication is experiencing golden days due to various wireless standards such as Wi-Fi, GSM, Wimax and LTE. These standards operate within lower microwave range (2-4GHz). Due to intrinsic propagation losses at these frequencies and problem of

multipath fading, it was necessary to provide a solution which can offer robustness in multipath environments and against narrowband interference and is efficient. OFDM, in all these aspects, proves to be an apt candidate by not only providing high-capacity, high-speed wireless broadband multimedia networks but also coexists with current and future systems.

Orthogonal frequency-division multiplexing (OFDM) is a method of digital modulation in which a signal is split into several narrowband channels at different frequencies. OFDM has been adopted by several technologies such as Asymmetric Digital Subscriber Line (ADSL) services, IEEE 802.11a/g, IEEE 802.16a, Digital Audio Broadcast (DAB), and digital terrestrial television broadcast: DVD in Europe, ISDB in Japan 4G, IEEE 802.

. Orthogonal frequency division multiplexing (OFDM) is a multicarrier modulation scheme for wireless communications [3]. The OFDM technique is widely used to obtain the high-speed transmission over frequency-selective fading channels [4]. Due to its multicarrier nature, the OFDM signals have usually amplitude variation in the time domain and have a relatively large dynamic range which is referred to as the peak-to-average power ratio (PAPR) [5]. In the case of high PAPR, the OFDM signal will be clipped when it passes through a non-linear high-power amplifier (HPA) and consequently, the performance will be degraded and in-band distortion and out-of-band radiation will occur. Thus, the OFDM transmitters require expensive linear HPA with a wide dynamic range [6]. There are many PAPR reduction approaches for OFDM systems, as clipping, coding, non-linear compounding, ton reservation and ton injection, selective mapping (SLM) and partial transmit sequence (PTS) [7]. Among these approaches, the PTS technique is the most efficient and distortion-less scheme for PAPR reduction in OFDM systems. In PTS technique, the input data block is split into several independent sub-blocks, the inverse FFT (IFFT) procedure is applied to each independent sub-block and each corresponding time-domain signal is multiplied by a phase rotation factor. The objective of the PTS scheme is the selection of phase factors in order to minimize the PAPR of the combined signal of all sub-blocks [7]. In the PTS, the exhaustive search complexity of optimal phase factors exponentially increases with the number of sub-blocks and phase rotation factors. To mitigate the search complexity, several different suboptimal PTS methods have been investigated in the previous studies [[8], [9], [10]].

This paper presents a novel approach based on particle swarm optimization (PSO) to overcome the computational complexity of the PAPR reduction problem in the PTS technique.

1.2 Problem Statement.

The peak-to-average power ratio (PAPR) is proportional to the number of sub-carriers used for OFDM systems. Thus, an OFDM system with a large number of sub-carriers will have a very large percentage of PAPR when adding sub-carriers coherently. The large system PAPR makes it extremely difficult to implement digital to analog converter (DAC) and analog to digital converter (ADC). The design of the radio frequency amplifier has also become increasingly difficult with the increase in PAPR.

The cutting and windowing technology reduces PAPR through nonlinear distortion of the OFDM signal. Hence it presents self-interference as the maximum level of amplitude is limited to a constant plane. It also increases out-of-range radiation, but this is the simplest way to reduce PAPR. To reduce the error rate, additional forward debugging codes can be used in conjunction with the cut-and-window method. Another technique called linear peak cancellation may also be used to reduce PAPR. In this method, the time-displacement reference function and the temporal scale are subtracted from the signal, such that the subtracted reference function temporarily reduces the peak power of at least one signal sample. By choosing a suitable reference function with roughly the same bandwidth as the transmitting function, it can be ensured that reducing the peak power does not cause out-of-band interference. One example of a suitable reference function is the raised cosine window.

In OFDM system The OFDM technique divides the total bandwidth into many narrow sub - channels and sends data in parallel. It has various advantages, such as high spectral efficiency, immunity to impulse interference and, frequency selective fading without having powerful channel equalizer, but one of the major drawbacks of the OFDM system is high PAPR. OFDM signal consists of lot of independent modulated subcarriers, which are created the problem of PAPR. It is impossible to send this high peak amplitude signals to the transmitter without reducing peaks. So, we have to reduce high peak amplitude of the signals before transmitting.

Major disadvantage of OFDM is that it has a high peak-to-average power ratio (PAPR) which means that a very linear output amplifier with large dynamic range is needed. Such amplifiers are inefficient and expensive to operate. Any amplifier non-linearity causes unwanted out-of- band power. However, the high magnitude peaks in the OFDM signals occur rarely. One of the simplest ways to reduce the PAPR is to clip the high amplitude peaks. Several clipping techniques have been described in the literature. Some clip the signal at the output of the inverse discrete Fourier transform (IDFT). But subsequent interpolation causes regrowth of the signal peaks Other clipping techniques clip the signal after interpolation and use a filter to reduce the resulting out-of-band power. However, the filters, which have been proposed, are complicated and computationally expensive. In addition, they cause peak regrowth and result in significant distortion of the wanted signal. The technique presented in clips the signal after interpolation and uses a new form of frequency domain digital filtering to filter the out-of-band power. This form of filtering results in less peak regrowth and causes no distortion to the wanted signal components, while reducing the out-of-band power to the same level as Orthogonal Frequency Division Multiplexing (OFDM) is widely used in many digital communication systems due to its advantages such us high bit rate, strong immunity to multipath and high spectral efficiency but it suffers a high Peak-to-Average Power Ratio (PAPR) at the transmitted signal. It is very important to deal with PAPR reduction in OFDM systems to avoid signal degradation. Currently, the PAPR problem is an active area of research and in this paper, we present several techniques and that mathematically analyzed. Moreover, their advantages and disadvantages have been enumerated in order to provide because they can be used in a multitude of applications, ranging from signal amplification to digital logic and memory. readers the actual situation of the PAPR problem.

1.3 OFDM Background

OFDM systems have been widely recognized as an efficient transmission technique for wireless communications and are extensively used in the standards for digital audio/video broadcasting. OFDM is a frequency-domain approach to communications, and has important advantages when dealing with the frequency-selective nature of high data rate communication channels. As the demand for operating with higher data rates has increased, OFDM systems have emerged proble an effective physical-layer solution in their environment [11].

The book is organized as follows. section 2 describes the classification of powerline noise and presents the mathematical algorithms and characteristics of the noises used in this book, section 3 describes the simulation response and noise modelling; Finally, Section 4 contains the concluding remarks.

1.4 Book Structure

This book is organized as follows:

1. **Chapter 1:** starts out with the introduction of OFDM technology respectively. That includes their basic structure and technical essential. Then, the PAPR problem and its definition are given. In the end, three distinctive types of PAPR reduction techniques are introduced as a lead-in to Chapter 3 in terms of basic principle and algorithm overhead.
2. **Chapter 2:** Advantages and Disadvantages of OFDM System.
3. **Chapter 3:** Orthogonality of OFDM System.
4. **Chapter 4:** Non-Distortion Clipping Technique PAPR Reduction of OFDM System Two sub-type algorithms, selected mapping (SLM) and partial transmit sequence (PTS) are investigated. A comprehensive analysis and comparison are conducted in terms of all possible influencing factors and PAPR reduction performance, respectively. Some research findings are obtained based on the simulation results. At last, we also compare SLM and PTS algorithm with respect to auxiliary information and PAPR reduction performances.

5. **In Chapter 5:** PAPR Reduction Methods and Noise Qualification Performances

6. **Finally, Chapter 6:** a conclusion of book drawn and some suggestions are provided for the future work.

Chapter Two

Orthogonal Frequency Division Multiplexing (OFDM)

2.1 Introduction

Orthogonal Frequency Division Multiplexing (OFDM) is a multicarrier modulation technique that divides the available spectrum into subcarriers, with each subcarrier containing low-rate data stream. The subcarriers have proper spacing and pass-hand filter shape to satisfy orthogonality as shown in Figure 2.1. OFDM will play an important role in realizing Cognitive Radio (CR) concept by providing a proven, scalable, adaptive technology for wireless communications [16). Inter-symbol interference (1SI) is reduced completely by using a guard band in every OFDM symbol.

In OFDM, using guard band is cyclically extended in order by avoid inter-curricr interference (ICI). The advantage of OFDM system is robustness channel fading in wireless communication environment. Frequency selective fading is reduced by increasing the number of subcarriers. By choosing the coherence bandwidth is greater than the subcarrier spacing of the channel, each subcarrier is going to be affected by a flat channel and thus no or simple channel equalizer is needed. OFDM is used in many wineless applications today, already it is used in different WLAN tandands (e.g. HIPERLAN 2, IEEE S02.1 la), Wireless Metropolitan Area Networks (WMAN), Digital Video Broakasting (DVB), 3GPP-LTE, Asymmetric Digital Subscriber Line (ADSL) and power line communications. Despite of OFDM advantages, it has a major potential drawback in the form of high Peak-to- Average Power Ratio (PAPR). The high PAPR has nonlinear manure in the transmitter and it degrades the power efficiency of the system

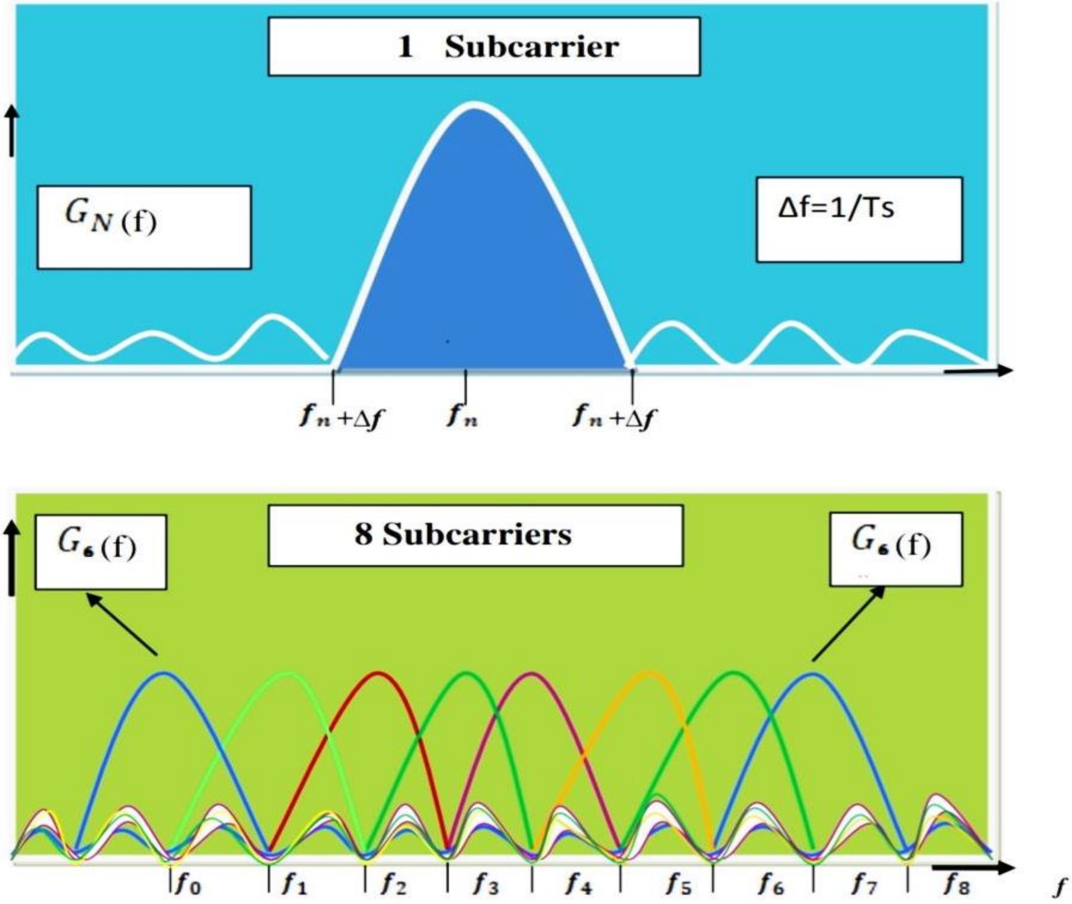

Figure 2. 1-OFDM subcarriers in frequency domain

2.2 Advantages and Disadvantages of OFDM System.

some advantages and disadvantages of OFDM are summarized below:

2.2.1 Advantages of OFDM.

Some of the advantages of an OFDM system are as follows:

• OFDM is computationally efficient to employ the modulation and demodulation techniques by using FFT.

• The OFDM signal is robustness in multipath propagation environment and more tolerant of delay spread.

• OFDM is more resistant to frequency selective fading than single carrier transmission systems. OFDM system gives good protection against co-channel interference and impulsive parasitic noise. Pilot subcarriers are used in OFDM system to prevent frequency and phase shift errors.

• It is possible to use maximum likelihood detection with reasonable complexity.

• OFDM is a good candidate for CR because of its flexibility and adaptability [16]. The orthogonality preservation procedures in OFDM are much simpler compared to CDMA/TDMA technique in multipath conditions [17].

2.2.2 Disadvantages of OFDM

Some of the disadvantages of an OFDM system are as follows:

• The OFDM signal suffers high peak to average power ratios (PAPR) of transmitted signal.

• OFDM is very sensitive to carrier frequency offset.

• It is difficult to synchronize when subcarriers are shared among different transmitters.

2.3 OFDM System Model

A Basic OFDM system is described in Figure 2.2. Here an input data symbols are supplied into a channel encoder that data are mapped on BPSK/QPSK/QAM constellation. The data symbols are converted from serial to parallel and using Inverse, Fast Fourier Transform (IFFT) to

$$x_n = IFFT\,\{X_k\}$$

$$= \frac{1}{N}\sum_{k=0}^{N-1} X_k e^{\frac{j2\pi kn}{N}} \qquad 0 \le n \le N-1$$

achieve the time domain OFDM symbols. Time domain symbols can be represented as where,X, is the transmitted symbol on the kh subcarriers N is the number of subcarriers Time domain signal is cyclically extended to prevent Inter Symbol Interference (ISI) from the former OFDM symbol using cyclie prefix (CP).

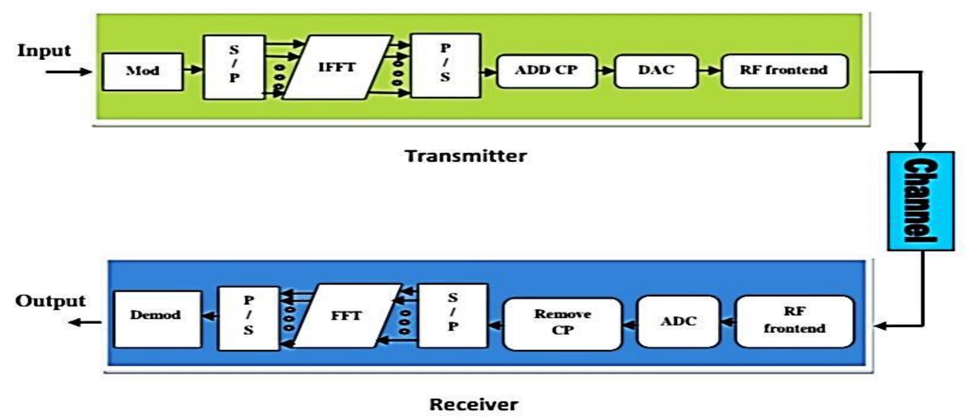

Figure 2. 2-A basic diagram of OFDM Transceiver

baseband digital signal into analog signal. This operation is executed in DAC block of diagram. Then, the analog signal is proceeded to the Radio Frequency (RF) frontend. The RF frontend performs operations after receiving the analog signal. The signal is up converted to RF frequencies using mixer and amplified by using Power Amplifier (PAs) and then transmitted through antennas. At the receiver side, the received signal is down converted to base band signal by RF frontend. The analog signal is digitized and re-sampled by the Analog to Digital Converter (ADC). The ADC is used to digitize the analog signal and re-samples it. In the figure, frequency and time synchronization block are not shown because of simplicity. Cyclic prefix is removed from the signal in frequency domain. This step is done by the Fast Fourier Transform (FFT) block. The received symbols in the frequency domain can be represented as:

$$Y (k) = H (k) Xu(k)+W (k) \qquad (2.2)$$

where, Y (k) is the received symbol on the kh subcarrier, H (k) is the frequency response of the channel on the same subcarrier and W (k) is the additive noise added to kth subcarrier which is generally assumed to be Gaussian random variable with zero mean and variance of o2. Thus, simple one tap frequency domain equalizers can be employed to get the transmitted symbois. After FFT signals are de-interleaved and decoded to recover the original signal.

2.4 Why PAPR reduction in OFDM system

The OFDM technique divides the total bandwidth into many narrow sub-channels and sends data in parallel. It has various advantages, such as high spectral efficiency, immunity to impulse interference and, frequency selective fading without having powerful channel equalizer. But one of the major drawbacks of the OFDM system is high PAPR. OFDM signal consists of lot of independent modulated subcarriers, which are created the problem of PAPR. It is impossible to send this high peak amplitude signals to the transmitter without reducing peaks. So we have to reduce high peak amplitude of the signals before transmitting. to the transmitter without reducing peaks. So, we have to reduce high peak amplitude of the signals before transmitting.

2.5 OFDM is used:

in many communication techniques as a method for mixing a group of small sub-channels to operate in a single band or band. It is distinguished from others by operating the band in an effective way, due to the (Orthogonality) property, which makes the sub-channels very close to each other but without interference between them. The speeds it offers are very fast when compared to other types such as FDM.

2.6 PAPR Techniques

There have been many new approaches developed during the last few years. Several PAPR reduction techniques have been proposed in the literature. These techniques are divided into two groups. These are signal scrambling techniques and signal distortion techniques. The signal scrambling techniques are:

- Block coding
- Selective Level Mapping (SLM)
- Partial Transmit Sequences (PTS)

Signal scrambling techniques work with side information which minimized the effective throughput since they commence redundancy. Signal distortion techniques introduce band interference and system complexity also. Signal distortion techniques minimize high peak dramatically by distorting signal before amplification.

The signal distortion techniques are:

- Clipping
- Peak windowing
- Peak cancellation
- Peak power suppression
- Weighted multicarrier transmission.

2.7 Clipping and Filtering:

High PAPR is one of the most common problems in OFDM. A high PAPR brings disadvantages like increased complexity of the ADC and DAC and also reduced efficiency of radio frequency (RF) power amplifier.

One of the simple and effective PAPR reduction techniques is clipping, which cancels the signal components that exceed some unchanging amplitude called clip level. However, clipping yields distortion power, which called clipping noise, and expands the transmitted signal spectrum, which causes interfering [18]. Clipping is nonlinear process and causes in- band noise distortion, which causes degradation in the performance of bit error rate (BER) and out-of-band noise, which decreases the spectral efficiency [19].

Clipping and filtering technique is effective in removing components of the expanded spectrum. Although filtering can decrease the spectrum growth, filtering after clipping can reduce the out-of-band radiation, but may also cause some peak re-growth, which the peak signal exceeds in the clip level [20]. The technique of iterative clipping and filtering reduces the PAPR without spectrum expansion. However, the iterative signal takes long time and it will increase the computational complexity of an OFDM transmitter [18].

But without performing interpolation before clipping causes, it out-of-band. To avoid out-of- band, signal should be clipped after interpolation. However, this causes significant peak re- growth. So, it can use iterative clipping and frequency domain filtering to avoid peak re- growth. In the system used, serial to parallel converter converts serial input data having different frequency component which are base band modulated symbols and apply interpolation to these symbols by zero padding in the middle of input data. Then clipping operation is performed to cut high peak amplitudes and frequency domain filtering is used to reduce the out of band signal, but caused peak re-growth [20]. This consists of two FFT operations. Forward FFT transforms the clipped signal back to discrete frequency domain. The in-band discrete components are passed unchanged to inputs of second IFFT while out of band components are null. The clipping and filtering process is performed iteratively until the amplitude is set to the threshold value level to avoid the peak out of band and peak re- growth.

Chapter Three

Orthogonality of OFDM System

3.1 Introduction

A number of non-zero subcarriers are included in an OFDM symbol period which last T seconds. Therefore, the frequency spectrum of the OFDM symbol can be seen to be a result of convolution between the spectrum of rectangular pulse and a group of sub-carriers at different frequencies. The duration of rectangular pulse is T. The spectrum of rectangular pulse is $\text{sinc}(f \cdot T)$.The zero points of this function only take place at integer multiples of *1/T*. For an assigned sub-carrier frequency point, only the corresponding sub-carrier can have a maximum value with all the other sub-carriers taking the value of zero at this point.

Therefore, based on this special property, symbols of each sub-carrier can be extracted from a number of overlapped sub-carriers during the modulation process and without causing any interference effects. Eq. (3.1) shows the mathematical expression for this phenomenon.

$$\frac{1}{T}\int_0^T e^{-j2\pi f_m t} \cdot e^{-j2\pi f_n t} dt = \begin{cases} 1 & m = n \\ 0 & m \neq n \end{cases} \qquad (3.1)$$

The expression for demodulating the k *-th* sub-carrier. The estimated transmitted discrete complex-valued signal $\hat{d}_k$ is as a result of integration over time T:

$$\begin{aligned} \hat{d}_k &= \frac{1}{T}\int_0^T e^{-j2\pi\frac{kt}{T}} \sum_{i=0}^{N-1} d_i \, e^{j2\pi\frac{it}{T}} dt \\ &= \frac{1}{T}\sum_{i=0}^{N-1} d_i \int_0^T e^{j2\pi\frac{i-k}{T}} dt = d_k \end{aligned} \qquad (3.2)$$

Observing the mathematical derivation above, we learned that for correct demodulation of the *k-th* sub-carrier, the value of integration must equal to 1 with the condition of $i = k$ being satisfied. If $i \neq k$, the power factor of complex integral variable $\frac{i-k}{T}$ is integer multiple of 1 / T, integration result is equals to 0.

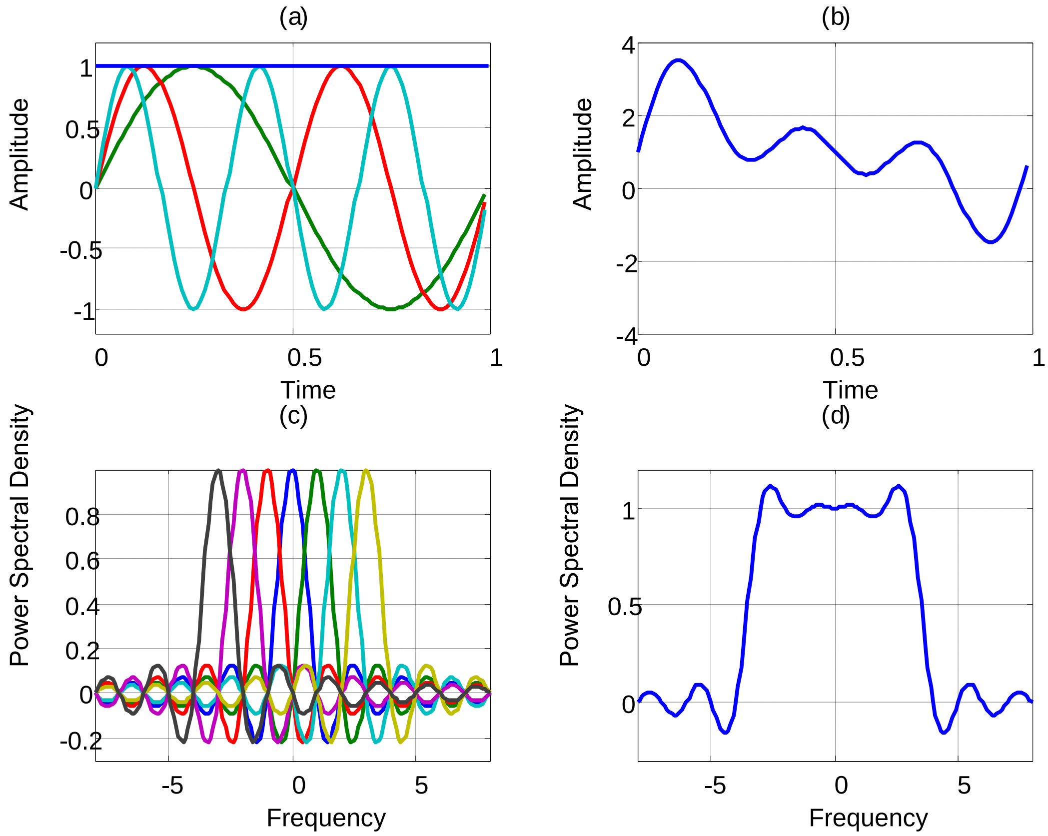

Figure 3.1 Four-carrier OFDM signal waveform.

Fig. 3.1 is an example of a four-carrier OFDM signal. In this example, it is assumed that all the waveforms have the same amplitude and phase. Fig. 3.1 (a) illustrates four independent sub-carrier waveforms in time-domain. Fig. 3.1 (b) shows the waveform of a composite OFDM signal in time domain. Fig. 3.1 (c) illustrates four independent sub-carrier waveforms in frequency domain. Fig. 3.1 (d) shows the waveform of a composite OFDM signal in frequency domain.

As can be seen from the Fig. 3.1, the adjacent sub–carrier has an integer number of cycles over one OFDM symbol period, and the discrepancy among all adjacent subcarriers is one carrier period, which ensures that subcarriers are orthogonal. In the frequency domain, all the overlapped subcarriers undergo a rectangular waveform shaping to generate the frequency spectrum in a form of sinc function.

3.2 Cyclic Prefix of OFDM System

In OFDM system, the use of Cyclic Prefix (CP) can guarantee orthogonality of signals even when they travel through multi-path channels [8]. To avoid ISI, the condition; $T_G > T_{max}$ should be satisfied, where T_G is the length of CP and T_{max} is the maximum delay spread [9].

As shown in Fig. 3.2, a CP is a copy of the last part of a OFDM symbol moved to the front of symbol. Assuming that the number of the extended OFDM symbol is N_G, then the period of a practical OFDM symbol is $T+T_G$, where T is cycle for the FFT transform, T_G is the length of guard interval, which is inserted to suppress ISI caused by multipath distortion. An OFDM symbol including CP can be expresses as follows:

$$s'_n = s'(t)|_{t=nt_x} = \sum_{i=0}^{N-1} d_i \cdot e^{j2\pi\frac{in}{N}}, n = N_G, \cdots, -1, 0, \cdots, N-1 \qquad (3.3)$$

Operation between the signal and channel changes from linear convolution to cyclic convolution when CP is used with OFDM. In the frequency domain, linear weighing will be used. These changes avoid inter-symbol interference, while ensuring orthogonality among the sub-carriers all the time.

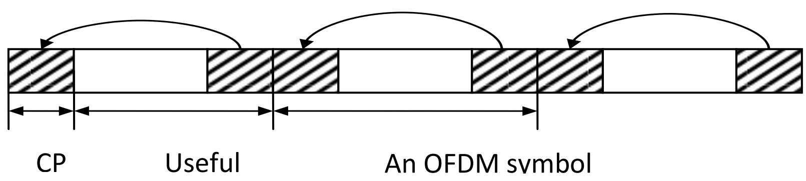

Figure 3.2 OFDM symbols with added cyclic prefix.

3.3 Peak-to-Average Power Ratio in OFDM System

The instantaneous output of an OFDM system often has large fluctuations compared to traditional single-carrier systems. This requires that system devices, such as power amplifiers, A/D converters and D/A converters, must have large linear dynamic ranges. If this is not satisfied, a series of undesirable interference is encountered when the peak signal goes into the non-linear region of devices at the transmitter, such as high out of band radiation and inter-modulation distortion. PAPR reduction techniques are therefore of great importance for OFDM systems [12].

3.3.1 PAPR Definition

Theoretically, large peaks in OFDM system can be expressed as Peak-to-Average Power Ratio, or referred to as PAPR, in some literatures, also written as PAR. It is usually defined as [13]:

$$PAPR = \frac{P_{peak}}{P_{average}} = 10\log_{10}\frac{\max\ [|x_n|^2]}{E[|x_n|^2]} \tag{3.4}$$

Where P_{peak} represents peak output power, $P_{average}$ means average output power. $E[\cdot]$ denotes the expected value, x_n represents the transmitted OFDM signals which are obtained by taking IFFT operation on modulated input symbols X_k. Mathematical, x_n is expressed as:

$$x_n = \frac{1}{\sqrt{N}}\sum_{k=0}^{N-1} X_k\, W_N^{nk} \tag{3.5}$$

For an OFDM system with N sub-carriers, the peak power of received signals is N times the average power when phase values are the same. The PAPR of baseband signal will reach its theoretical maximum at $PAPR(dB) = 10\log N$. For example, for a 16 sub-carriers system, the maximum PAPR is 12 dB. Nevertheless, this is only a theoretical hypothesis. In reality the probability of reaching this maximum is very low.

Fig. 3.3 shows the amplitude characteristic of an OFDM system with 16 sub-carriers. According to the graph, it can be seen that the maximum magnitude of the OFDM signals is less than the upper limit value 16 and corresponding PAPR is also lower than the theoretical maximum 12dB.

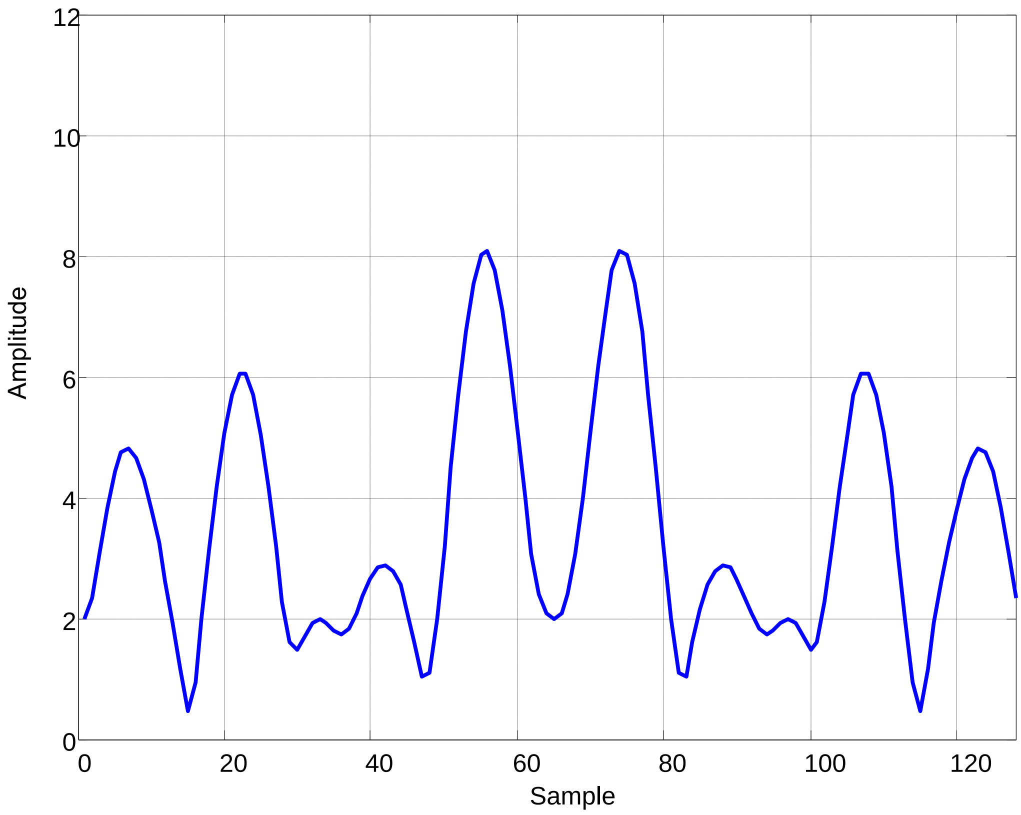

Figure 3.3 An OFDM signal waveform in time domain.

The special case happens when signal sub-carriers are modulated by symbols which have the same initial phase. Assuming that input binary sequence contains „1" for the whole sequence. After PSK constellation mapping and IFFT operation, instant power reaches its theoretical maximum. Fig. 3.4 shows the result when input binary sequence contains 16 „1", denoted by [1111111111111111]. In this scenario, the maximum amplitude reaches the value of 16. The PAPR can be calculated from $PAPR(dB)$ = 10log N and in this case it is 12dB.

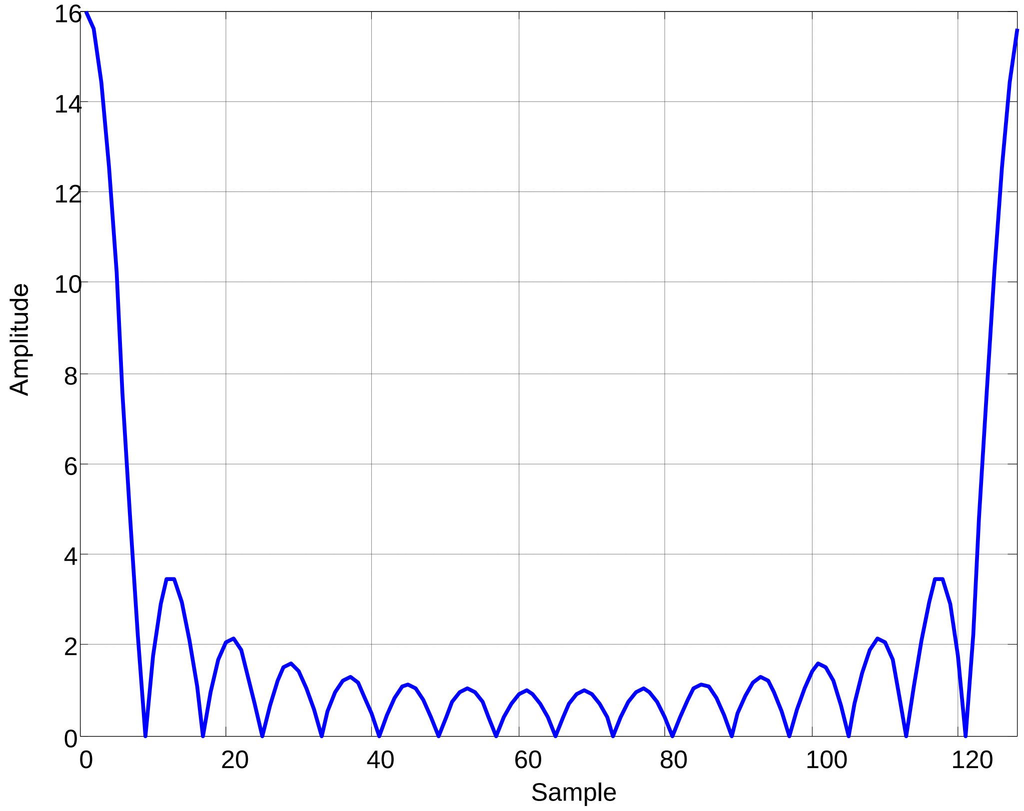

Figure 3.4 High PAPR when sub-carriers are modulated by same symbols.

By observing the simulation result in Fig. 3.4, we can make a conclusion that the amplitude of OFDM signal reaches its peak value when the input data sequence has a larger consistency. At the same time, the maximum PAPR value will be reached as well.Another commonly used parameter is the Crest Factor (CF), which is defined as the ratio between maximum amplitude of OFDM signal $s(t)$and root-mean-square (RMS) of the waveform. The CF is defined as [14]:

$$CF\left(s(t)\right) = \frac{\max\ [|s(t)|]}{E[||s(t)|^2|]} = \sqrt{PAPR} \qquad (3.6)$$

In most cases, the peak value of signal $x(t)$is equals to maximum value of its envelope

$|x(t)|$.However, it can be seen from Fig. 3.3 that the appearance of peak amplitude is very rare, thus it does not make sense to use $\max\left(|x(t)|\right)$to represent peak value in real application. Therefore, PAPR performance of

OFDM signals is commonly measured by certain characterization constants which are related to probability.

3.3.2 Probability Distribution Function of PAPR

According to central limit theorem, for a large number of sub-carriers in multi-carrier signal, the real and imaginary part of sample values in time-domain will obey Gaussian distribution with mean value of 0 and variance of 0.5. Therefore, the amplitude of multi-carrier signals follows Rayleigh distribution with zero mean and a variance of N times the variance of one complex sinusoid [15]. Its power value obeys a χ^2 distribution with zero mean and 2 degrees of freedom. Cumulative Distribution Function (CDF) is expressed as follows

$$F(z) = 1 - \exp(-z) \qquad (3.7)$$

Assuming that the sampling values of different sub-channels are mutually independent, and free of oversampling operation, the probability distribution function for PAPR less than a certain threshold value,is therefore expressed as

$$P(PAPR < z) = F(z)^N = (1 - \exp(-z))^{\mathrm{N}} \qquad (3.8)$$

In practice, it is preferred to take the probability of PAPR exceeding a threshold as measurement index to represent the distribution of PAPR. This can be described as "Complementary Cumulative Distribution Function" (CCDF), and its mathematical expression as

$$P(PAPR > z) = 1 - P(PAPR \leq z) = 1 - F(z)^N = 1 - (1 - \exp(-z))^{\mathrm{N}} \quad (3.9)$$

Fig. 3.5 shows the theoretical PAPR"s CCDF distribution with different number of subcarriers (i.e. N = 32, N = 128, N = 1024). The x-axis represents the PAPR thresholds while the y-axis represents the probability of CCDF. As can be seen from the graph, for a given PAPR threshold, the appearance probability of OFDM symbols which above this threshold PAPR0 will decrease with the increase of sub-carriers number *N*. In this thesis, we will use CCDF to evaluate the performance of various PAPR reduction techniques.

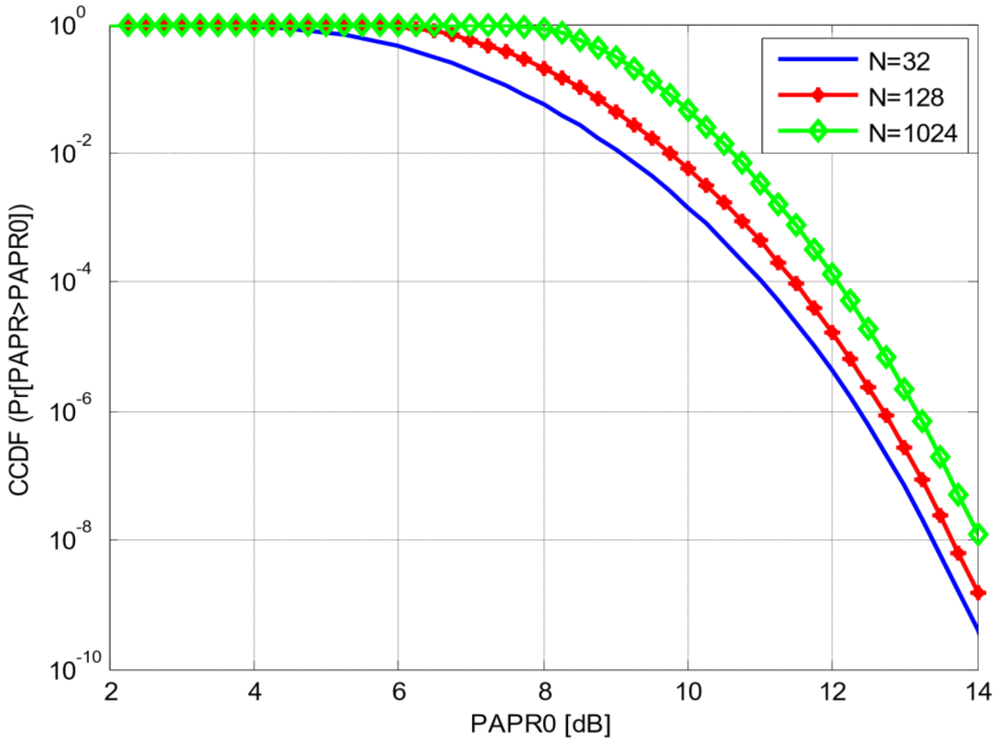

Figure 3.5 Theoretical PAPR"s CCDF curve of OFDM signal.

3.4 Partial Transmit Sequence

3.4.1 Principle of PTS (Partial Transmit Sequence)

Partial Transmit Sequence (PTS) algorithm was first proposed by Müller S H, Huber J B [21][22], which is a technique for improving the statistics of a multi-carrier signal. The basic idea of partial transmit sequences algorithm is to divide the original OFDM sequence into several sub-sequences, and for each sub-sequence, multiplied by different weights until an optimum value is chosen.

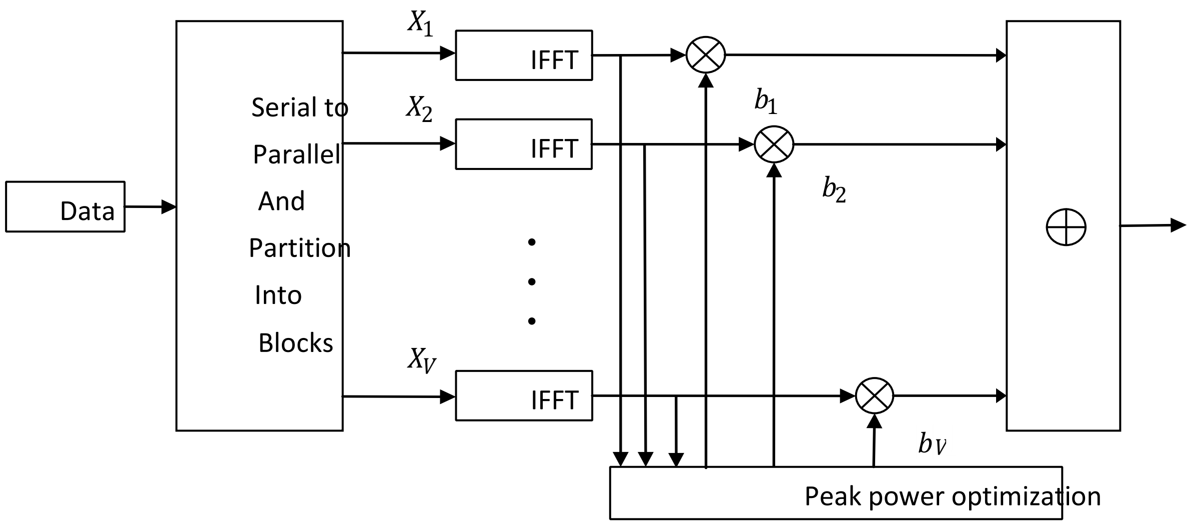

Figure 3.6 Block diagram of PTS algorithm

Fig. 3.6 is the block diagram of PTS algorithm. From the left side of diagram, we see that the data information in frequency domain $\mathbf{X}$ is separated into V non-overlapping sub-blocks and each sub-block vectors has the same size N. Hence, we know that for every sub-block, it contains N/V nonzero elements and set the rest part to zero. Assume that these sub-blocks have the same size and no gap between each other, the sub-block vector is given by

$$\hat{\mathbf{X}} = \sum_{v=1}^{V} b_v \mathbf{X}_v \qquad (3.10)$$

where $b_v = e^{j\varphi_v}(\varphi_v \in [0,2\pi])\ \{v = 1,2,\dots,V\}$ is a weighting factor been used for phase rotation. The signal in time domain is obtained by applying IFFT operation on X_v, that is

$$\hat{\mathbf{x}} = IFFT(\hat{\mathbf{X}}) = \sum_{v=1}^{V} b_v IFFT(\mathbf{X}_v) = \sum_{v=1}^{V} b_v \cdot \mathbf{x}_v \qquad (3.11)$$

Select one suitable factor combination $\mathbf{b} = [b_1, b_2, \dots, b_v]$ which makes the result achieve optimum. The combination can be given by

$$\mathbf{b} = [b_1, b_2, \dots, b_v] = \arg\min_{(b_1,b_2,\dots,b_v)}(\max_{1\le n\le N} \ |\sum_{v=1}^{V} b_v\ x_v|^2) \qquad (3.12)$$

where $\arg\min(.)$ is the judgment condition that output the minimum value of function. In this way we can find the best $\mathbf{b}$ so as to optimize the PAPR performance. The additional cost we have to pay is the extra V-1 times IFFTs operation.

In conventional PTS approach, it requires the PAPR value to be calculated at each step of the optimization algorithm, which will introduce tremendous trials to achieve the optimum value [21]. Furthermore, in order to enable the receiver to identify different phases, phase factor **b**is required to send to the receiver as sideband information (usually the first sub-blockb_1 ,is set to 1). So the redundancy bits account for$(V-1)\log_2 W$, in which V represents the number of sub-block, W indicates possible variations of the phase. This causes a huge burden for OFDM system, so studying on how to reduce the computational complexity of PTS has drawn more attentions, nowadays.

The optimization is achieved by searching thoroughly for the best phase factor. Theoretically, $\mathbf{b} = [b_1, b_2, \dots, b_v]$ is a set of discrete values, and numerous computations will be required for the system when this phase collection is very large. For example, if φ_v contains W possible values, theoretically, **b**will haveW^V different combinations, therefore, a total of$V \cdot W^V$ IFFTs will be introduced.

By increasing the V, W, the computational cost of PTS algorithm will increase exponentially.

For instance, define phase factor$b_v$contains only four possible values, that meansb_v $\in$

$[\pm 1, \pm j]$, then for each OFDM symbol, $2 \cdot (V-1)$bits are transmitted as side information. Therefore, in practical applications, computation burden can be reduced by limiting the value range of phase factor $\mathbf{b} = [b_1, b_2, \dots, b_v]$ to a proper level. At the same time, it can also be changed by different sub-block partition schemes.

$[\pm 1, \pm j]$

3.5 Mathematical Definition of OFDM Signal

OFDM consists of multiple carriers. Each carrier can be presented as a complex waveform like:

$$s_c(t) = A_c(t)e^{j[\omega_c t+\phi_c t]}, \quad (3.13)$$

where,

$A_c(t)$ is the amplitude of the signal $s_c(t)$

ϕ_c(t) is the phase of the signal $s_c(t)$

The complex signal can be described by

$$s_s(t) = \frac{1}{N}\sum_{n=0}^{N-1} A_n(t)e^{j[\omega_n t+\Phi_n t]}, \quad (3.14)$$

This is a continuous signal. Each component of the signal over one symbol period can take fixed values of the variables like:

$\phi_n(t) \Rightarrow \phi_n$,

$A_n(t) \Rightarrow A_n$,

where, n is the number of OFDM block.

T is a time interval and the signal is sampled by 1/T then it can be represented by:

$$s_s(kT) = \frac{1}{N}\sum_{n=0}^{N-1} A_n e^{j[(\omega_0+\omega\Delta n)kT+\phi_n]}, \quad (3.15)$$

Let ω_0=0 then the signal becomes:

$$s_s(kT) = \frac{1}{N}\sum_{n=0}^{N-1} A_n e^{j[(\omega\Delta n)kT+\phi_n]},$$

(3.16)

The signal is compared with general Inverse Fourier Transform (IFT):

$$g(kT) = \frac{1}{N}\sum_{n=0}^{N-1} G(\frac{n}{NT})e^{j[2\pi nk/N]} \tag{3.17}$$

Here, $s(kT)$ is time frequency domain. Both are equivalent if

$$\Delta f = \frac{\Delta\omega}{2\pi} = \frac{1}{NT} = \frac{1}{\tau} \tag{3.18}$$

where,

τ is symbol duration period

The OFDM signal can be defined by Fourier Transform. The Fast Fourier Transform (FFT) can obtained frequency domain OFDM symbols and Inverse Fast Fourier Transform (IFFT) can obtain time domain symbols. They can be written as:

Fast Fourier Transform

$$X(k) = \sum_{n=0}^{N-1} x(n)e^{-j(2\pi/N)kn} \tag{3.19}$$

Inverse Fast Fourier Transform

$$x(n) = \sum_{n=0}^{N-1} X(n)e^{j(2\pi/N)kn} \tag{3.20}$$

where,

$0 \leq n \leq N-1$

3.6 Mathematical Definition of PAPR

The PAPR of the OFDM signal can be written as:

$$\text{PAPR}\{s(t), \tau\} = \frac{max_{t\in\tau}[s(t)]^2}{E\{[s(t)]^2\}} \quad (3.21)$$

where, s(t) is the original signal

τ is the time interval

$max_{t\in\tau}[s(t)]^2$ is the peak signal power $E\{[s(t)]^2\}$ is the average signal power

3.7 Additive White Gaussian Noise (AWGN) Channel

The simplest channel model in wireless commutations is the well known Additive White Gaussian Noise (AWGN) model. The mathematical expression of the AWGN channel as follows:

$$Y(t) = X(t) + N(t), \quad (3.22)$$

where, X(t) is the transmitted signal and plus N(t) is the white Gaussian Noise.

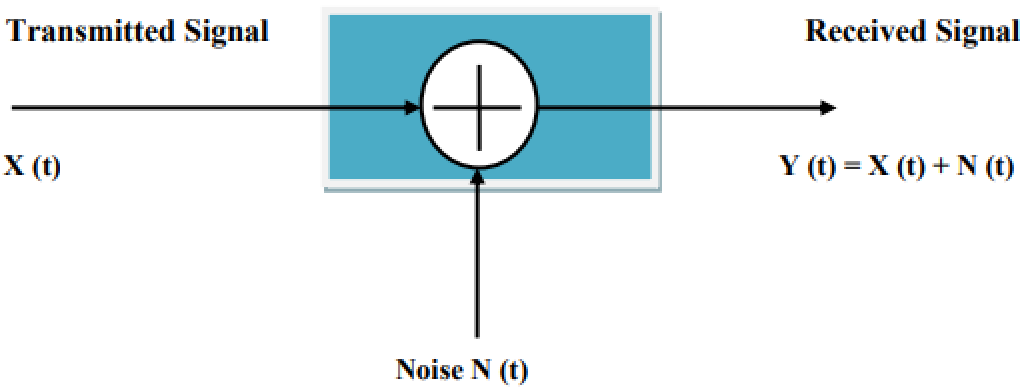

Figure 3.7- AWGN channel

AWGN channel is considered as an important reference or benchmark model for comparing the performance evaluation of communication systems and modulation formats. However, when the signal travels from transmitter to receive via multiple propagation paths then a practical fading channel model must be used to model the propagation environment. There are some factors affecting fading including multipath propagation, speed of surrounding objects, speed of the mobile, the transmission symbol duration and the transmission bandwidth of the signal.

3.8 Mathematical Expression

Clark model is applicable to mobile reception in general scattering environments. R. k Clark modelled the mobile channel as a Rayleigh fading channel in. R. K Clark considered non-LOS between the transmitter and the receiver. The radio signal is reflected and scattered due to obstacles such as buildings, mountains and trees. Clark has also considered the Doppler effect because of motion of mobile unit.

The phase and angle of arrival of each element wave will be statistically independent. Clark model expresses the carrier signals received at the mobile whose phases are supposed to be Gaussian random variables and the phase angle is uniformly distributed on the interval 0 to 2π. Doppler shift is given by:

$$f = f_m \, Cos \, \varphi \tag{3.23}$$

where, $f_m = \frac{v}{\lambda}$ f_m is the carrier frequency v is the mobile velocity φ is the angle λ is the carrier wave length .According to the Clark assumption, the E–filed can be represented as an in phase and quadrature.

$$E_z = T_c(t) \, Cos(2\pi f_c t) - T_s(t) \, Sin \, (2\pi f_s t) \tag{3.24}$$

where,

$$T_c \; (t) = E_0 \sum_{n=1}^{N} \; C_n \mathrm{Cos} \left(2\pi f_n \mathrm{t} + \phi_n \right)$$

$$\mathrm{T_s(t)} = \; \mathrm{E_0} \sum_{n=1}^{N} \; C_n \, Sin \left(2\pi f_n \; t + \Phi_n \right)$$

Here, T_c and T_s both are Gaussian random processes. Those are uncorrelated zero-mean Gaussian random variables and variance is equal which given by:

$$\sigma^2 = \frac{{E_0}^2}{2}$$

Then, the magnitude of E-field is specified by:

$$r(t) = |E_z(t)| = \sqrt{{T_c}^2(t) + {T_s}^2(t)} \tag{3.24}$$

Rayleigh distribution is given by in [52]:

$$P(t) = \begin{cases} \frac{r}{\sigma^2} exp\left[\frac{-r^2}{2\sigma^2}\right], & 0 \le r \le \infty \\ 0, & r < 0 \end{cases} \tag{3.25}$$

where, σ^2 is the variance of the Rayleigh distributed variables

3.9 OFDM SIGNAL MODEL AND PEAK TO AVERAGE POWER RATIO (PAPR)

An OFDM systems use orthogonal and independent subcarriers, where the modulation process can be easily implemented using simple N-point inverse fast Fourier transform (IFFT) operation, the output signal is a summation of modulated complex harmonics [12] that can be expressed as [13, 18]:

$$x(t) = \frac{1}{N}\sum_{k=0}^{N-1} s_k e^{j2\pi f_k t} \qquad 0 \le t \le T_s \tag{3.26}$$

where s_k is the data symbol transmitted on the kth sub carrier fx, the subcarrier index is denoted by k and T, is the symbol duration OFDM systems introduce high PAPR due to the use of a large number of subcarriers, for a continuous time baseband OFDM signal, the definition of PAPR of any signal is the proportion of the maximum instantaneous power of the signal to its mean power. If x(t) represents a baseband transmitted OFDM signal, then

$$PAPR[x(t)] = \frac{\underset{0 \le t \le T_s}{MAX} |x(t)|^2}{P_{av}} \quad (3.27)$$

where, Pav is the average power of x(t) which can be computed in frequency domain because IFFT is a unitary transformation. To represents the useful duration of an OFDM symbol .

The complementary cumulative distribution function (CCDF) has used to describe a distribution of a random variable [11]. The distribution of PAPR may be expressed in terms of (CCDF), which is represents the probability that the PAPR of an OFDM symbol exceeds a given threshold PAPRO, which is denoted as [1,20).

CCDF(PAPRO) = P(PAPR > PAPRO) = 1 - P(PAPRO) (3.28)

The CCDF has also used to evaluate the performance of PAPR reduction in OFDM systems (1, 20). Tone reservation method is one of the many techniques, which have introduced in the literatures to reduce the PAPR of an OFDM signal.

can be presented as

[b1..., bv]

$$= \arg\ \min_{[b_1,\ldots,b_V]} \left(\max_{n=0,1,\ldots,N-1} \left| \sum_{v=1}^{V} b_v \cdot x_v[n] \right| \right) \quad (3.29)$$

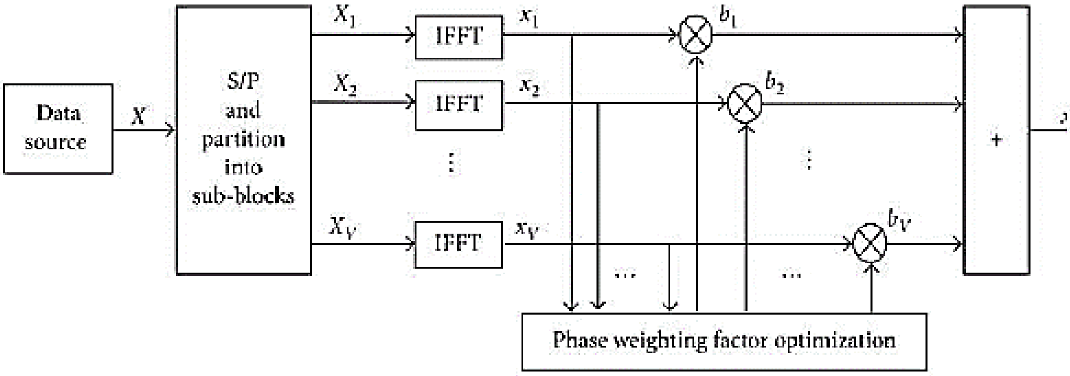

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Figure 3.8 Block diagram of PTS technique.

In the practical application of wireless communication systems using the PTS approach, the PAPR performance will be improved as the number of subblocks V is increased and to match the optimal phase weighting sequence for each input data sequence, W' possible combinations should be checked (W number of phase factors), which needs a huge quantity of computations to analyze all possible applicant rotation phase vectors. Therefore, the computational complexity increases.

Chapter Four

Non-Distortion Clipping Technique PAPR Reduction of OFDM System

4-1 Introduction

The output of an OFDM system frequently has large variation or fluctuations compared to conventional single-carrier systems. This imposes new requirements on system devices like power amplifiers, D/A converters and A/D converters, which are forced to have huge linear dynamic ranges and hence increased cost. If the requirements are not met, a series of unwanted interferences such as inter-modulation distortion and large out-of-band radiation take place when the peak signal swings into the non-linear range of the devices at the transmitter. Therefore, PAPR reduction techniques have become very important in OFDM systems [36].

In theory, PAPR, sometimes referred to as PAR, in OFDM is defined as:

$$PAPR_{SISO-OFDM} = \frac{P_{peak}}{P_{average}} = \frac{\max\,[|x_n|^2]}{E[|x_n|^2]} \qquad (4.1)$$

4-2 PAPR Reduction Technique

Several techniques have been suggested in the literature to minimize the PAPR. These techniques can fundamentally be categorized into signal scrambling techniques and signal distortion techniques. The signal scrambling techniques are all diversity on how to scramble the codes to reduce the PAPR. Coding techniques can be used for signal scrambling. More practical solutions of the signal scrambling techniques are Partial Transmit Sequences (PTS), block coding, and Selective Level Mapping (SLM). Signal scrambling techniques with side information reduce the efficacious throughput. The signal distortion techniques introduce both in-band and out-of-band interference and complexity to the system. The signal distortion techniques reduce high peaks directly by distorting the signal prior to amplification. Clipping the signal is a simple method to limit PAPR. More actual solutions are peak cancellation, peak windowing, weighted multicarrier transmission, companding, and peak power suppression. The distribution of PAPR reduction techniques is shown in Fig.4.1 [37].

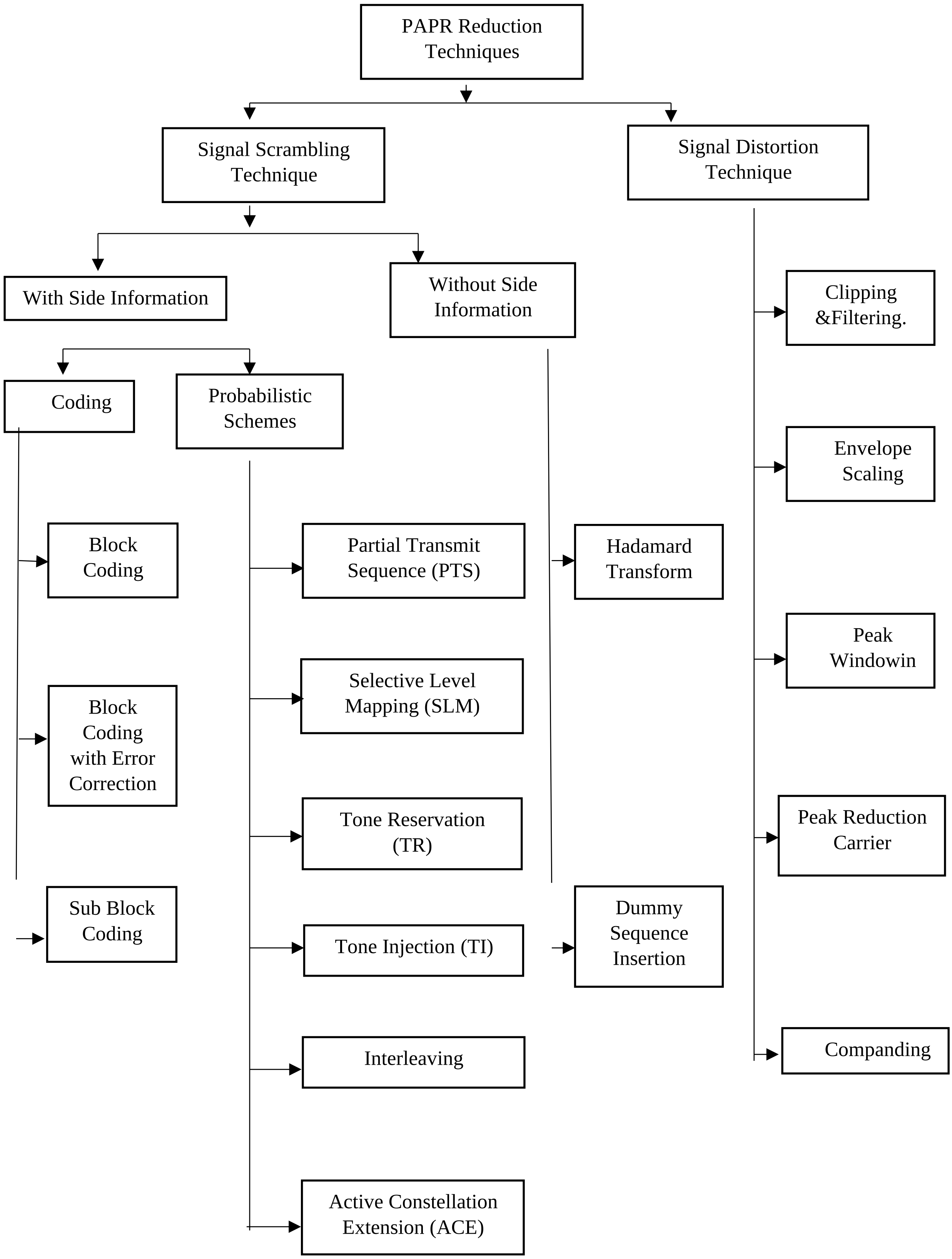


Figure 4.1 Scheme of Distribution the PAPR Reduction Techniques

4-3 Conventional Clipping Technique

Due to the nonlinear distortion established by this process (Clipping Technique), orthogonality will be destroyed partly which results in critical in band noise and out of band noise .In conventional clipping, previous to overtaking the power amplifier, envelope of the signal is clipped to an established in advance threshold called clipping level (CL) depends on clipping ratio (CR) is defined as.

$$CR = 20\log_{10}(CL/E[x_i(n)]) \qquad (4.2)$$

To reduce the PAPR in MIMO-OFDM, the clipping operation is applied on the OFDM signal as

$$C[x_i(n)] = \begin{cases} x_i(n) & if|x_i(n)| \leq CL \\ CL\, e^{j\varphi} & if|x_i(n)| > CL \end{cases} \qquad (4.3)$$

where $x_i(n)$ is the discrete time STBC-OFDM symbol, φ phase angle of $x_i(n)$,and i represents the transmit antenna (1,2,3,…) . Clipping technique is the simplest method of PAPR reduction, but it causes interference, noise, distortion or synchronization errors.

Clipping too large peaks is a simple solution to the PAPR problem, reduce large peaks without add extra information to the signal [38]. The maximum peak power allowed is determined by the system specifications, usually by the linear region of the power amplifier. A maximum peak amplitude CL is chosen so that the OFDM signal does not exceed the limits of this region, symbols that exceed this maximum amplitude, will be clipped. The graphical expression of this clipping function is shown in Fig.4.2.

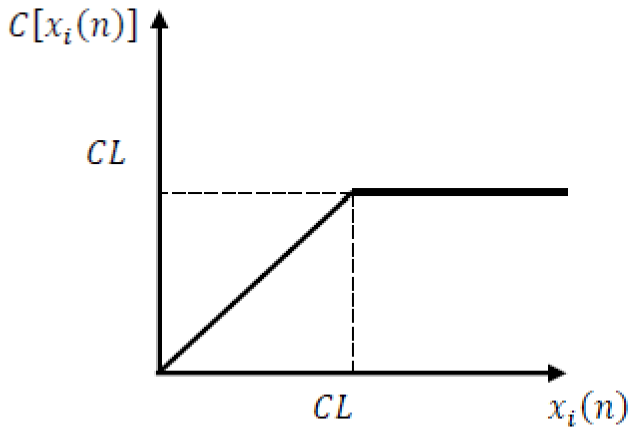


Figure 4.2. Clipping function

As we explained earlier that the clip or cut the high peaks of OFDM signals can causes unwanted in band and out of band spectral, which leading to out-of-band interference signals to neighboring channels and this problem can be overridden by filtering. Filtering the clipped signal can reduce band radiation, but cause peak regrowth. The signal after filtering process may exceed the clipping level specified.

4-4 Non-Distortion Clipping Technique

In this research the clipping is used directly at the output of the IFFT where the data is still in terms of samples as shown in Fig. 3 , the main concept is convert the distortion to noise and obey to the noise theory. The advantage are elimination of distortion, and reduce system complexity. The problem is how return actual data signal when have been transmitted.

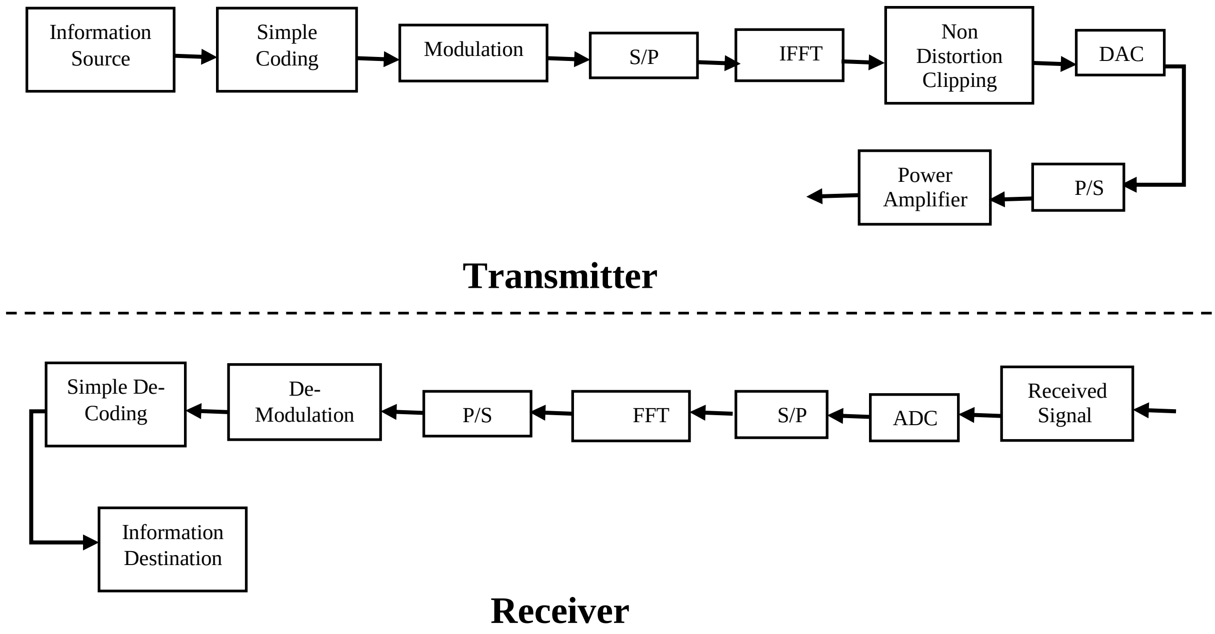
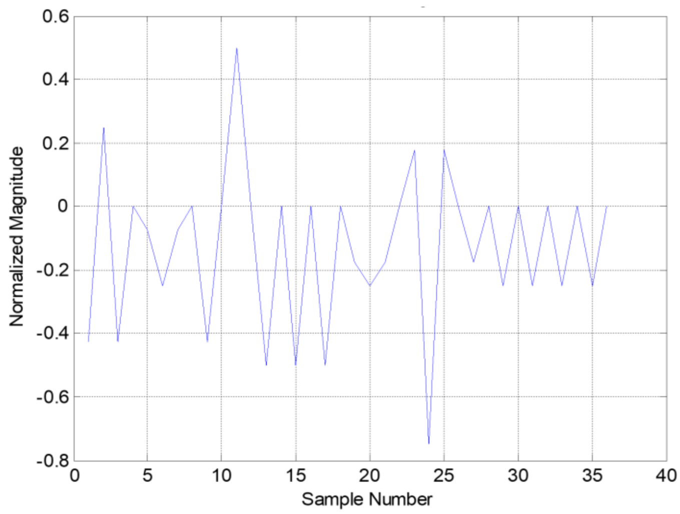


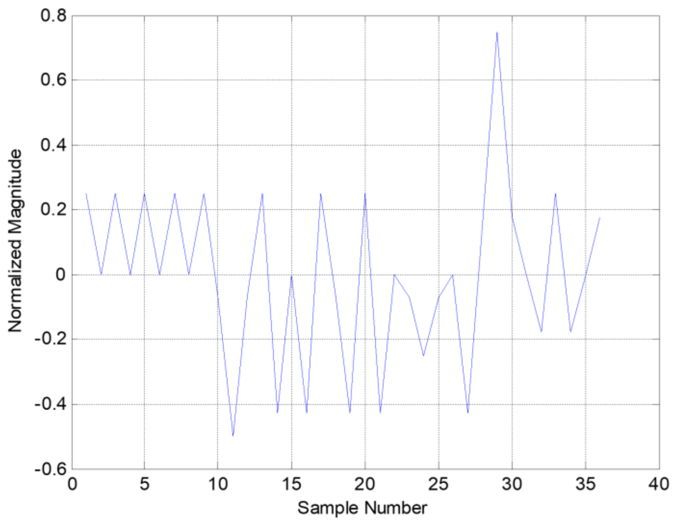
Figure 4.3 OFDM System with Non-Distortion clipping technique

In this work the process of clipping can be classified in two types the first one can be named zero error that is mean there is no clipping happen. The second one can be called non-zero error thus in this situation can be treated or considered as a Burst Error, In telecommunication, an error burst is a neighboring sequence of symbols, received over a communication channel, like the first and last symbols are in error and there is no neighboring subsequence of correctly received symbols within the error burst [39]. And then can be introduce as an impulse noise block sample, where can be managed by using error correction code [40]. Definitely, there are many types of noise which are not Gaussian; for instance, white noise, or impulse noise that has passed through a nonlinear device [41].

4-5 Results and Discussion

In this part, simulations have been done to test the PAPR reduction ability, and performance of the non-distortion clipping technique in OFDM systems. The OFDM system is simulated using MATLAB software with N=256 subcarriers, system bandwidth of 20MHz, and 4-QAM modulations.





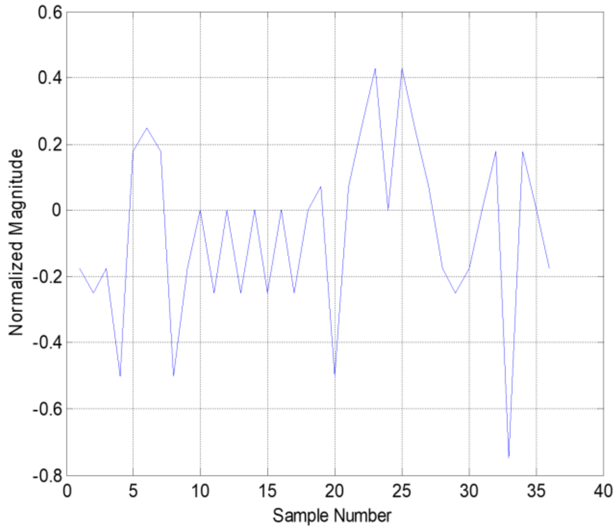


Figure 4.4. Different Output Signal of OFDM Signal, N=8

The analysis and observation of the output signal of OFDM system show that there is a difference in peak value through the change in transmitted data as shown in Fig.4.4.

4-5-1 Proposed Clipping PAPR Reduction Method

The signal for OFDM system with and without non-Distortion clipping is sketched for both BPSK and QPSK modulation, difference number of subcarriers (N=128) are used and the clipping ratio of 9 dB. It can be seen from Fig. 5 and Fig. 6, PAPR reduction of around 3 dB is achievable using clipping technique, but there is a change capability of non-Distortion clipping a round 0.5 dB compared to BPSK modulation with clipping in both systems as shown Fig.7 and Fig.8. Using other types of modulation such as (QPSK) that differ from BPSK modulation be represented.

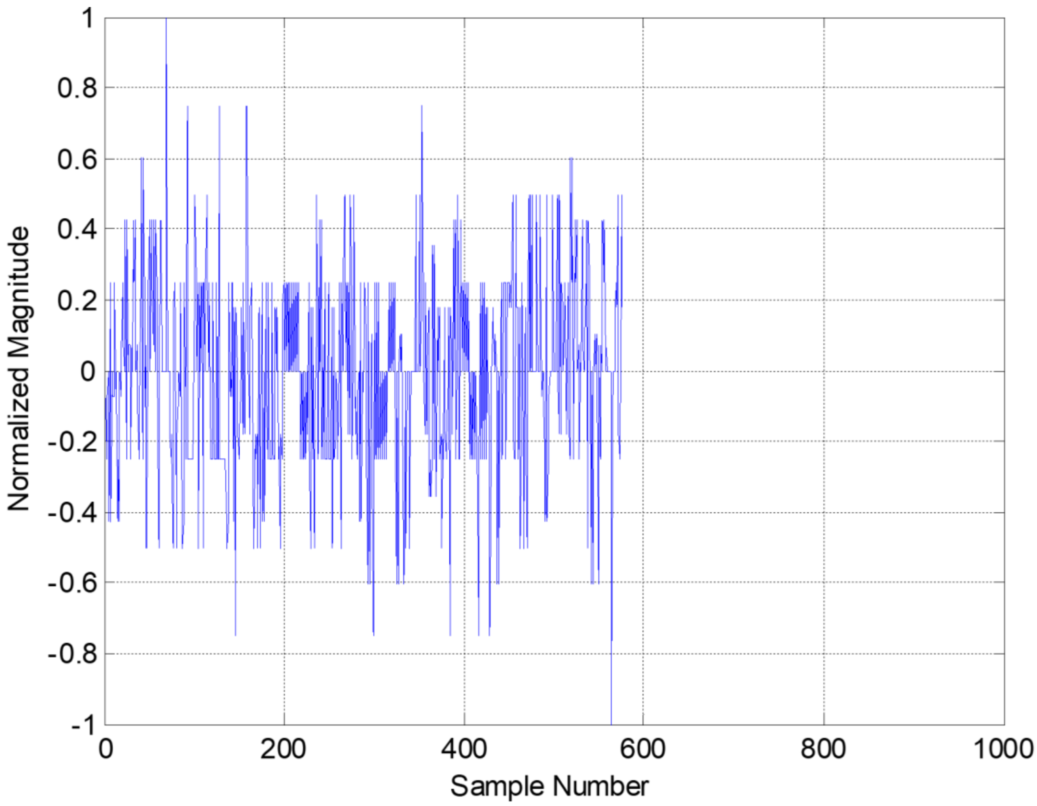


Figure 4.5. OFDM Signal without Clipping Technique, BPSK, N=128, PAPR=18.425 dB

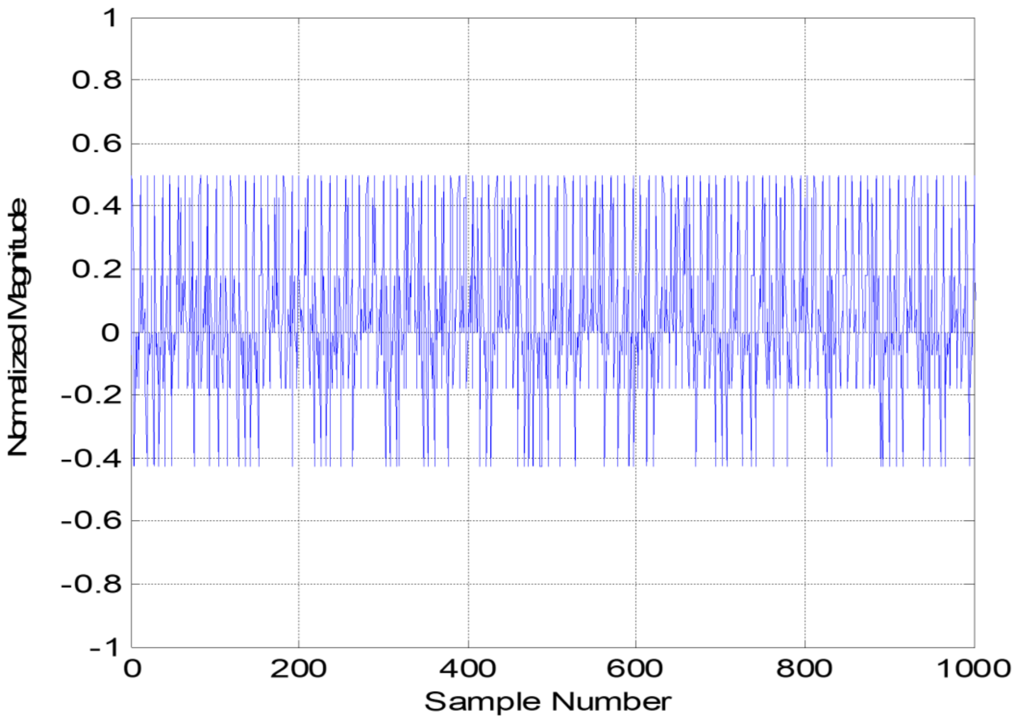


Figure 4.6. OFDM with Proposed Clipping Technique, BPSK, N=128, PAPR=15.101

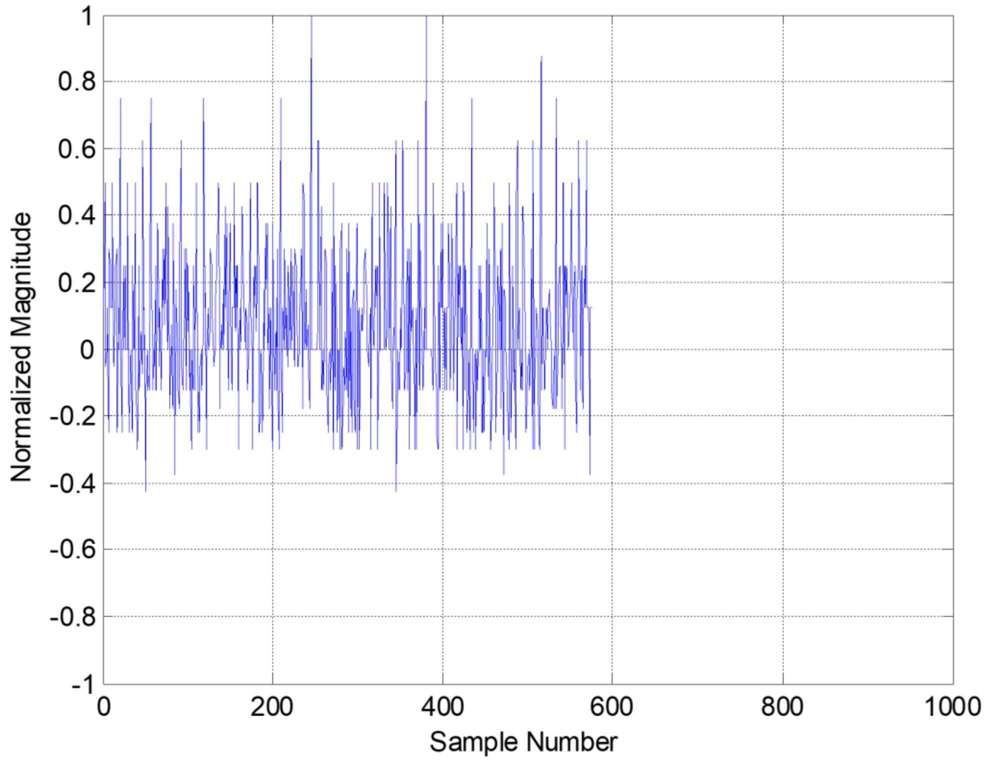


Figure 4.7. OFDM Signal without Clipping Technique, QPSK, N=128, PAPR=18.618 dB

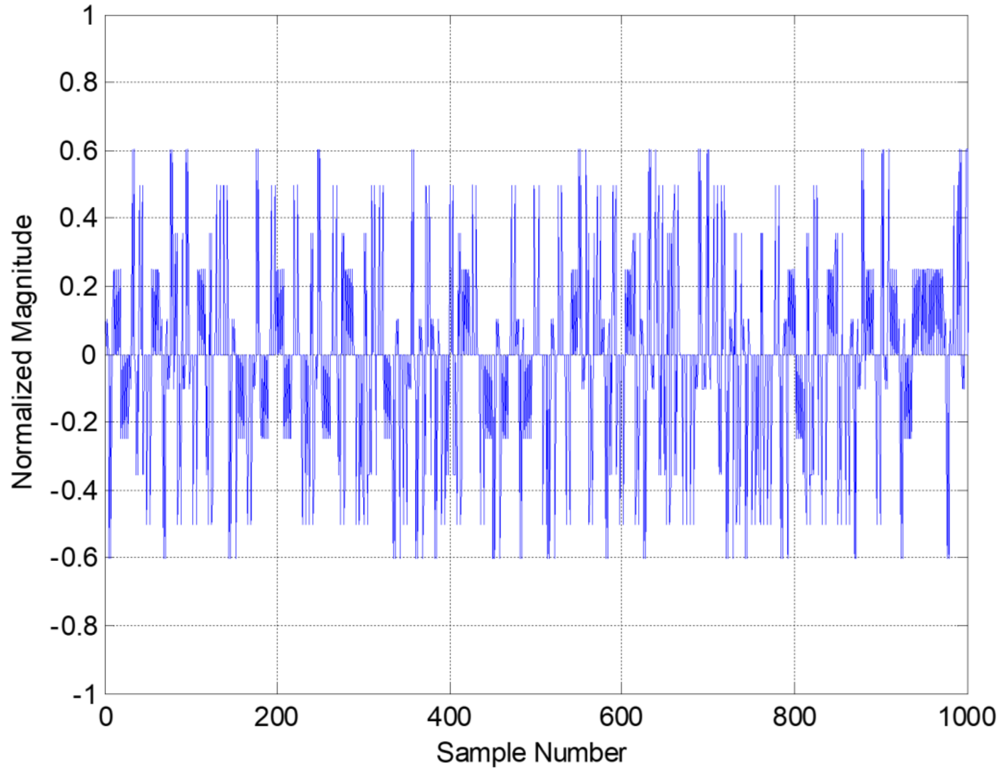


Figure 4.8. OFDM with Proposed Clipping Technique, QPSK, N=128, PAPR=15.212 dB

4-5-2 Power Spectral Density (PSD)

It can be seen from Fig.4.9 that filtering extremely attenuates the out-of-band Emission, the spectral side lobes after filtering are now at least 45 dB lower than the signal main lobe. Fluctuation caused by the filter may support the power of some sub-channels while suppressing others. In Fig. 4.10, shows the PSD of the Non-Distortion clipped signal, the out-of-band noise emission power a round 16 dB lower than the signal power.

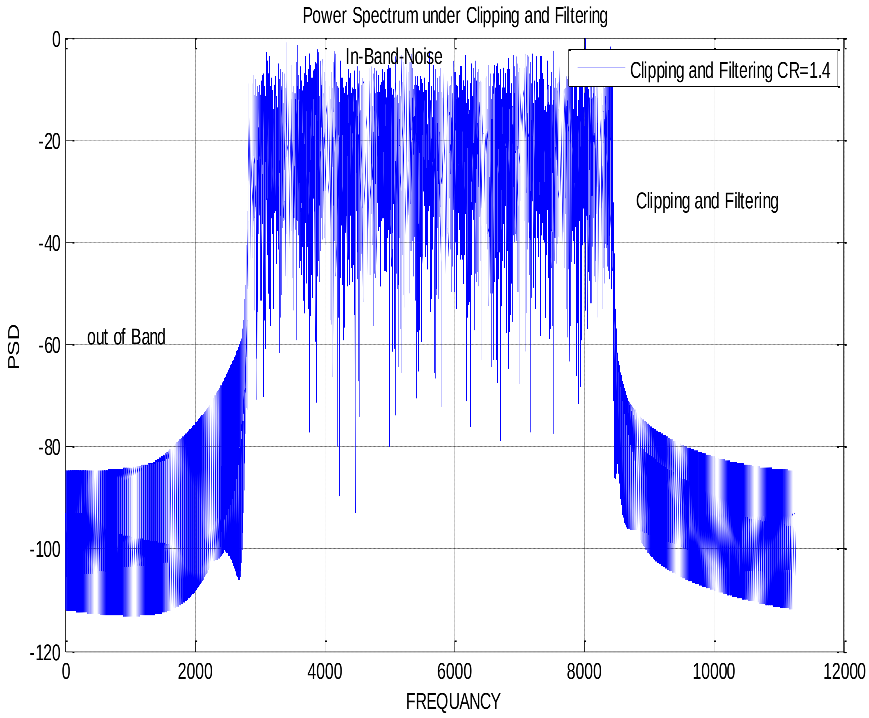


Figure 4.9 PSD of the Conventional Clipped-and-Filtered OFDM Signals.

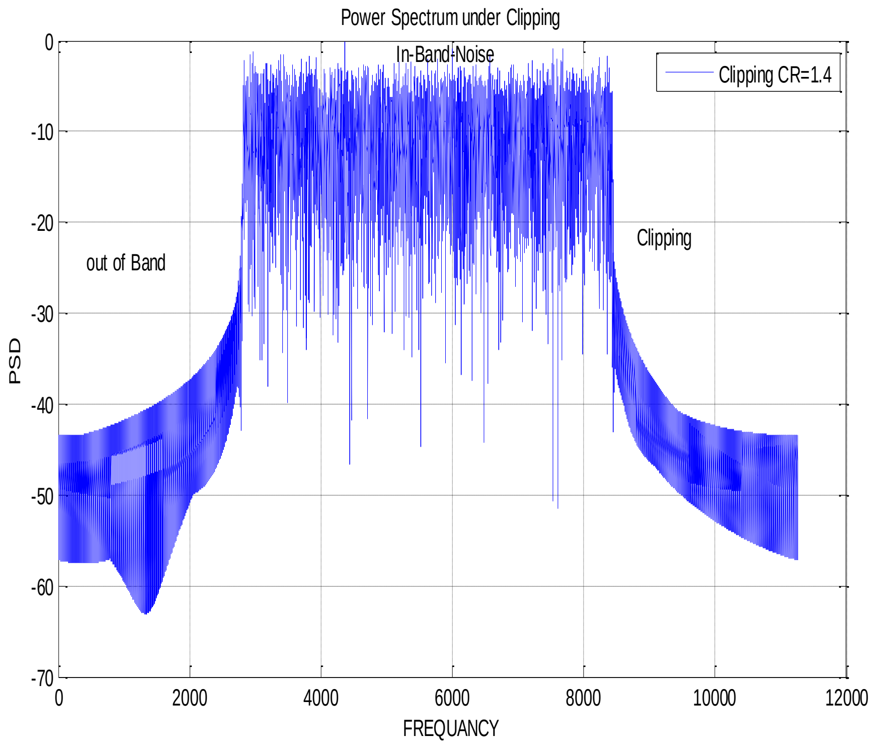


Figure 4.10 PSD of. Non-Distortion Clipped OFDM Signals.

Chapter Five

PAPR Reduction Methods and Noise Qualification Performances

5.1 Introduction.

Several techniques have been proposed in the literature to reduce the PAPR. These techniques can mainly be categorized into signal scrambling techniques and signal distortion techniques.

The signal scrambling techniques are all variations on how to scramble the codes to decrease the PAPR. Coding techniques can be used for signal scrambling. Golay complementary sequences [42, 43] Shapiro-Rudin sequences, m-sequences, Barker codes can be used efficiently to reduce the PAPR. However, with the increase in the number of subcarriers the overhead associated with exhaustive search of the best code would increase exponentially. [16] i.e. coding techniques are not used if the number of subcarrier is more than 32. More practical solutions of the signal scrambling techniques are block coding, Selective Level Mapping (**SLM**) and Partial Transmit Sequences (**PTS**). Signal scrambling techniques with side information reduce the effective throughput since they introduce redundancy.

The signal distortion techniques introduce both in-band and out-of-band interference and complexity to the system. The signal distortion techniques reduce high peaks directly by distorting the signal prior to amplification. Clipping the OFDM signal before amplification is a simple method to limit PAPR. However, clipping may cause large out of band (*OOB*) and in-band interference, which results in the system performance degradation. More practical solutions are peak windowing, peak cancellation, peak power suppression, weighted multicarrier transmission, commanding etc. Basic requirement of practical PAPR reduction techniques includes the compatibility with the family of existing modulation schemes, high spectral efficiency and low complexity. The classification of PAPR reduction techniques is shown in Figure (5-1) [44].

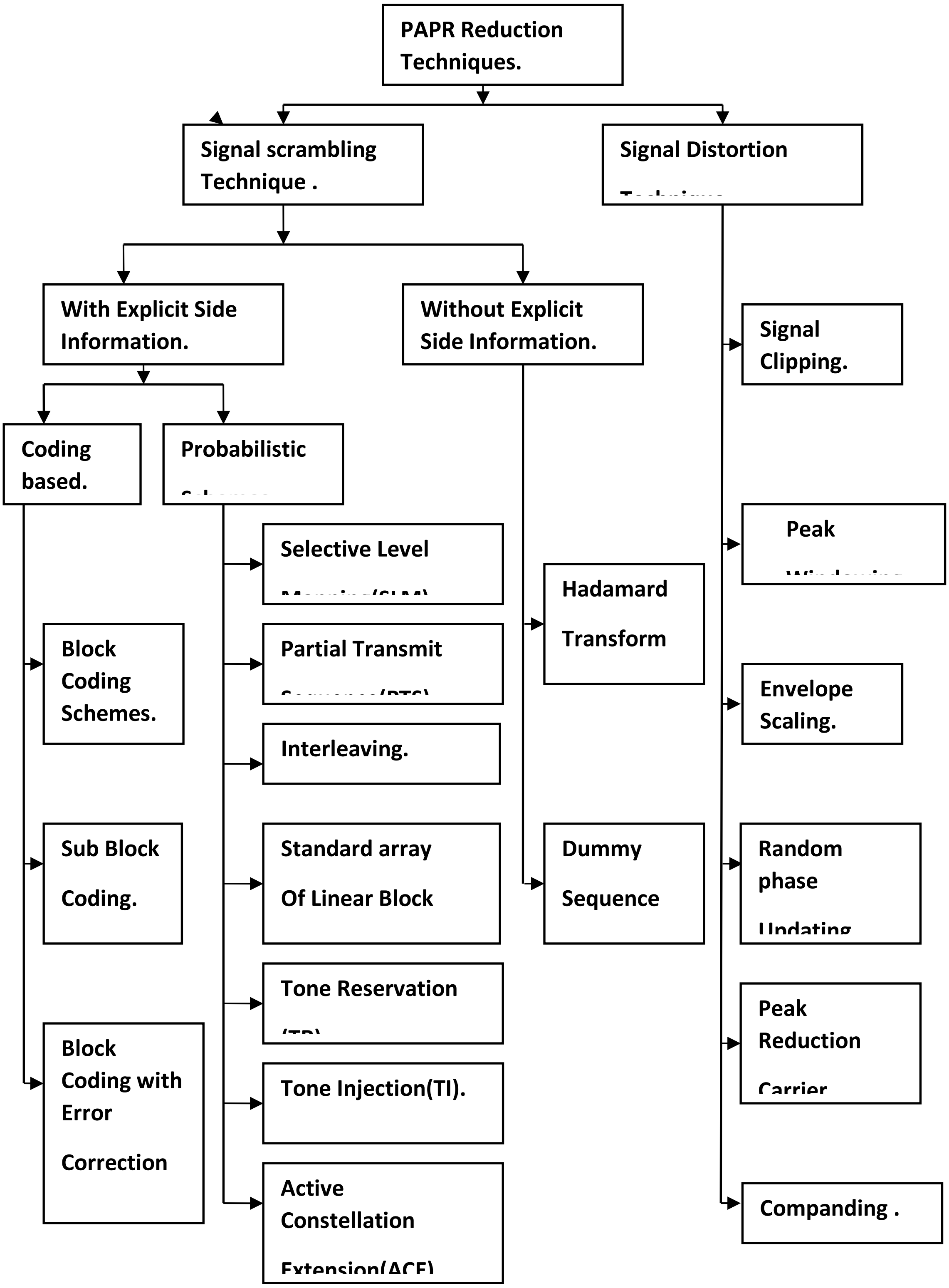


Figure 5-1: Classification of the PAPR reduction techniques [44].

5.2 Criteria for Selecting PAPR Reduction Techniques.

In selecting the appropriate PAPR reduction technique from the literature, the following criteria can be used in determining the trade-offs between PAPR reduction ability and other design factors: [6]

1.*High PAPR reduction capability*: The PAPR reduction technique should have high PAPR reduction capability, with as few undesirable side effects as possible. [6]

2-*Low computational complexity*: The PAPR reduction technique should be computationally efficient. Both time and hardware requirements for the PAPR reduction should be minimal. Generally, complex techniques exhibit better PAPR reduction as possible.

3- *No loss in throughput* : The loss in throughput due to side information should be avoided or at least be kept minimal. Usually, there exists a tradeoff between the loss in throughput and distortion of the signal. Non-blind techniques suffer from reduced throughput without distorting the signal, whereas blind techniques suffer from distorted signal without reducing throughput. Signal distortion eventually results in degraded BER performance.

4- *No BER performance degradation*: The blind techniques suffer from BER performance degradation, when the transmited power is held constant. Moreover, in the non-blind techniques, error in the side information might result in a whole erroneous data frame. However, attempts should be made to keep BER performance degradation to a minimum.

5- *No spectral spillage*: Spectral spillage is very important aspect of the PAPR reduction technique since it must be avoided. For this reason, clipping techniques are not preferred, even if they exhibit significant PAPR reduction.

6- *No power increase in transmit signal*: For wireless communications, power of the transmit signal should be kept constant. So the techniques such as ACE (Active Constellation Extension) might not be suitable for wireless communications.

7- *High efficiency and cost savings*: The efficiency of the PAPR reduction technique and the cost savings offered by it in terms of additional hardware requirements and saving in the cost of D/A converters and power amplifiers are also very important factors while choosing the PAPR reduction technique. The efficient PAPR need not be the lowest achievable PAPR. [6]

Among several PAPR reduction algorithms proposed in the literature, it is not sufficient to achieve large reduction in PAPR values alone, it is equally important to ensure that other overall system requirements are not violated. However, it may not be feasible to fulfill all the requirements of an ideal PAPR reduction algorithm for a given scenario and a balance between PAPR reduction and other factors, such as overall cost, error performance, complexity, and efficiency, should be found.

PAPR reduction can be achieved by modifying OFDM signal characteristics in time or frequency domain at the transmitter. If information about these changes is transmitted to the receiver, it can reverse the operation, and demodulate the data correctly. Otherwise, these modifications in OFDM characteristics would appear as signal distortion and result in degraded error performance for the overall communication system. Therefore, there always exists a tradeoff between error performance degradation and throughput loss, complexity increase, or information overhead. Depending upon error robustness for a given system, blind or non-blind technique can be chosen for PAPR reduction. For example, if a system has high tolerance to error performance, such as voice communications, blind techniques are preferable, whereas for a system with low tolerance to error performance, non-blind techniques would be suitable.

Another important factor to be considered for choosing an appropriate algorithm is PAPR reduction requirements. In case of non-blind technique, higher PAPR reduction requirements may lead to a system with high computational complexity or large information overhead, whereas in case of blind techniques, they may lead to a large error performance degradation due to significant changes in OFDM signal characteristics for minimizing PAPR. Moreover, in non-blind techniques, error on overhead information can significantly degrade error performance of the overall system. [6]

With PAPR reduction, it is possible to reduce dynamic ranges for PAs and D/A converters, which would result in cost savings on these components. However, it is also important to consider additional cost of the components needed for PAPR reduction, such as compandors, filters, or components for the overhead information management. Therefore, there also exists a tradeoff between cost of the additional RF components versus cost savings in PAs and D/A converters by reducing their dynamic ranges.

To make a fair comparison among PAPR reduction techniques, average transmit power should not be increased. Moreover, a practical transmit power constraint needs to be enforced for fulfilling the regulatory requirements. Therefore, depending upon the requirements, certain techniques may not be applicable, which would otherwise be useful for wireline communication, such as DSL.

Another important factor is the increase in the power of out-of-band interference, which also needs to be considered before deciding upon the PAPR reduction algorithm. In several scenarios, including *clipping*-OFDM system, spectral spillage can be a serious problem, since the bandwidth under consideration might be non-contiguous. In such cases, techniques yielding higher level of OOB interference, such as time-domain based techniques, may not be used,[6].

5.3 Definition of Efficient PAPR in OFDM.

PAPR reduction techniques minimize the PAPR at the cost of BER performance degradation or reduction in the system throughput. Assuming that the net data rate must be constant, a decrease in the number of information-carrying subcarriers means less power is devoted to the information bits yielding an increase in the BER. It has been shown that there exists a PAPR for which the BER of the information bits reaches the minimum value. An efficient PAPR is the PAPR level for which the BER reaches a minimal value, which need not necessarily be the lowest possible value of PAPR. [6]

5.4 Peak to Average Power Ratio (PAPR) Reduction Methods and Noise Mitigation Techniques.

This chapter analyzes some of the techniques used to reduce high PAPR in OFDM systems.

5.4.1 Peak to Average Power Ratio (PAPR) Reduction by Using Clipping method.

Clipping is a type of Transparent Methods. Clipping too large peaks is a simple solution to the PAPR problem. Clipping belongs to the group of techniques that reduce large peaks by nonlinearly distorting the signal [11]. It does not add extra information to the signal and too large peaks occur with low probability so the signal is distorted. The maximum peak power allowed is determined by the system specifications, usually by the linear region of the power amplifier.

A maximum peak amplitude **A** is chosen so that the OFDM signal does not exceed the limits of this region, symbols that exceed this maximum amplitude, will be clipped. The graphical expression of this clipping function is shown in Figure (5-2). The clipping is performed in digital time domain, before the **D/A** conversion, as shown in Figure (5-3), and the process is described by the following expression.

$$x_k^c = \begin{cases} x_k & |x_k| \le \mathbf{A} \\ \mathbf{A}e^{j\phi(x_k)} & |x_k| > \mathbf{A} \end{cases} \quad , \quad 0 \le k \le N-1 \qquad \text{...............(5-1)}$$

where x_k^c is the clipped signal, x_k is the transmitted signal, **A** is the clipping amplitude and $\phi(x_k)$ is the phase of the transmitted signal x_k.

The clipping ratio (**CR**) is defined as.

$$\mathbf{CR = A / \sigma} \qquad \ldots\ldots\ldots\ldots\ldots\ldots(5\text{-}2)$$

In the following discussion, a normalized clipping level will be used, which is called the clipping ratio (where **σ** is the **rms** level of the OFDM signal).

It is easy to show that, for an OFDM signal with N subchannels, [11] $\sigma = \sqrt{N}$ for a baseband signal and, $\sigma = \sqrt{(N/2)}$ for a bandpass signal. If N=128 then, $\sigma = \sqrt{(128/2)}=8$. A **CR**=1 equivalent to **A**=8, this means that signal is clipped at rms power level. A **CR** of 1.4 means that the clipping level is about 3 dB higher than the **rms** level and a **CR** of 0.8 means that the clipping level is about 2 dB lower than the **rms** level [11].

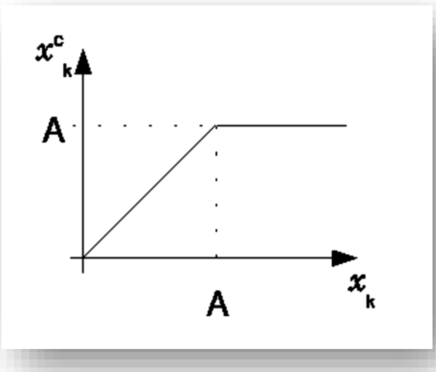


Figure 5-2: Clipping function.

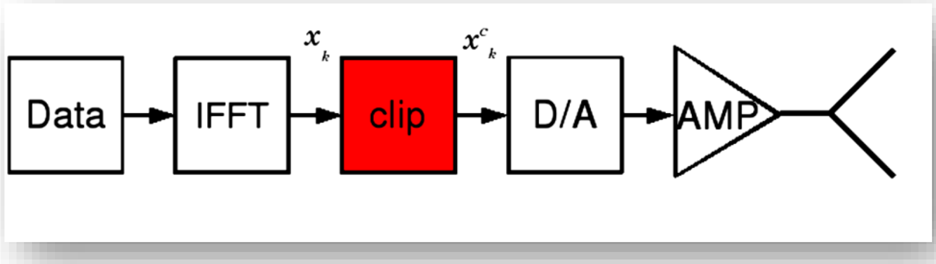


Figure 5-3: Clipping in the transmitter for OFDM system.

and the clipping ratio in dB is given by:

$$\mathbf{CR}[\mathrm{dB}] = 20\mathbf{log}_{10}\ (\mathbf{A}\ /\sigma)\ . \qquad \ldots\ldots..(5\text{-}3)$$

the mathematical expression for σ is :

$$\sigma^2 = 1/\ N = \sum_{k=1}^{N} x_k^2\ . \qquad \ldots\ldots.(5\text{-}4)$$

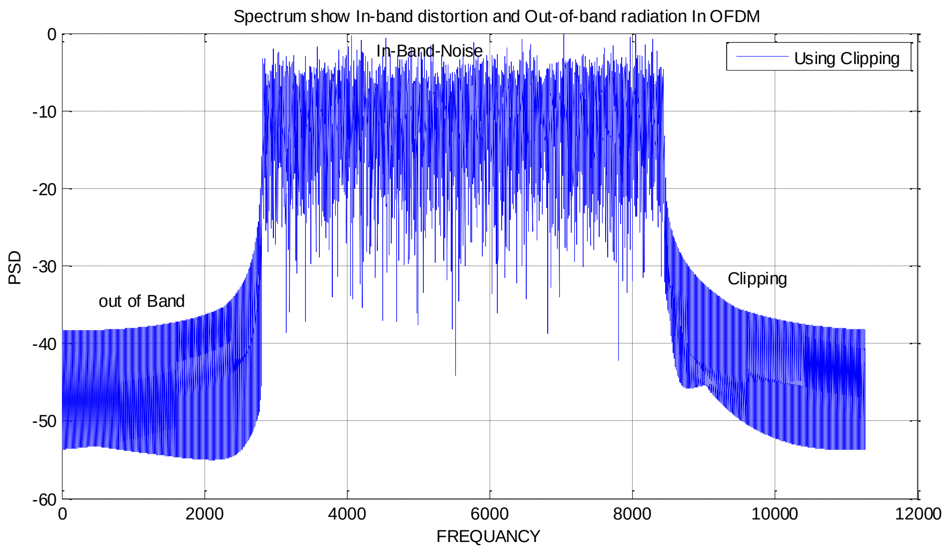


Figure 5-4: In-band distortion and Out-of-band radiation ,CR=0.8,N=1024

Clipping is a non-linear process so it introduces in-band distortion Figure (5-4), shows the clipping noise, and out-of-band radiation which causes inter-carrier interference (**ICI**).i.e. degrades the system performance and the spectral efficiency.

5.4.1.1 Signal to Clipping Noise Ratio (SCNR).

The clipping noise is related to the difference between the original signal x_k and the clipped signal x_k^c.[45] The signal sent to the receiver is the clipped signal, which is different from the signal that is actually wanted to be sent. This difference is measured by the signal to clipping noise ratio (**SCNR**).

$$\mathbf{SCNR} = \frac{|x_k^c|^2}{|x_k - x_k^c|^2} \qquad \text{………..(5-5)}$$

The **SCNR** (Figure (5-5)) is the ratio between the clipped signal x_k^c and the distortion introduced when clipping the signal, that is the difference between the unclipped signal and the clipped one. The more the signal is clipped, the greater difference between clipped and unclipped signals will be and so the **SCNR** will decrease. On the other hand, the less the signal is clipped, the smaller the difference and the **SCNR** will increase.

Figure (5-5) shows the **SCNR** measured for signals with different number of subcarriers (from 128 to 512). For a low **CR**, the **SCNR** is low and approximately the same for all the different signals. This makes sense since it is already deduced), i.e for a very low PAPR, the probability of having a signal with higher PAPR is 1 independently of the number of subcarriers of the signal. When the **CR** is very high, the signal is not clipped anymore and so from a certain **CR**,the **SCNR** stops increasing, maintaining the same value for all the **CR**.

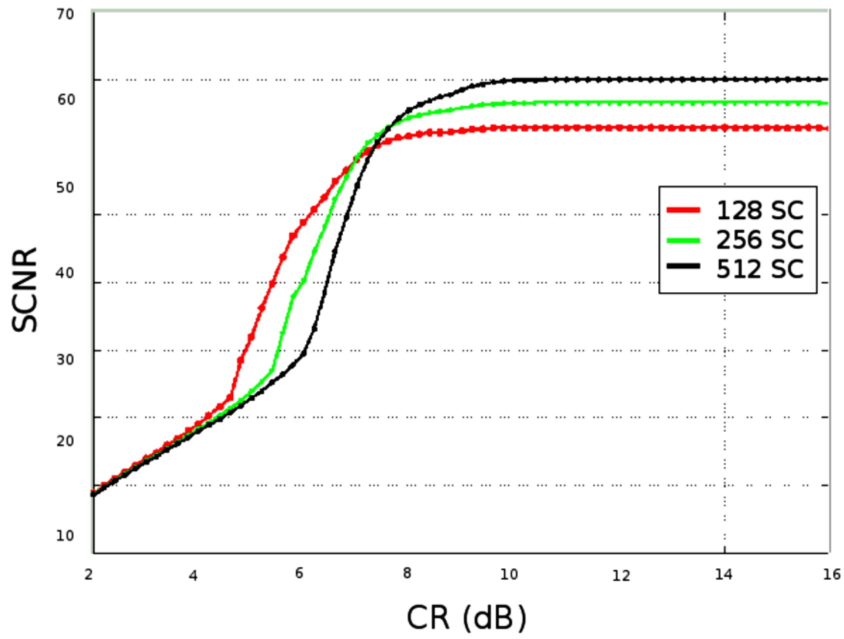


Figure 5-5: SCNR measured for different number of subcarriers [45]

5.4.1.2 Noise Mitigation: Clipping and Filtering.

Introducing a filter after the clipping operation helps in mitigating the out-of-band radiation introduced by clipping. Filtering, however, distorts also the signal and causes peak regrowth [11]. A common filter used for this purpose is the FIR filter. In this application, the filter used is the FFT-IFFT filter proposed by Armstrong in [46] . This filter achieves better performance than the FIR filter, it introduces less noise into the signal, causes less peak regrowth and needs less computation; since it operates symbol by symbol it causes no inter-symbol interference (ISI).

The filter itself consists of a set FFT-IFFT operations where filtering takes place in frequency domain after the FFT function. The first step in order to implement this filter is introducing $N(L-1)$ zeros (being L the over sampling rate) in the middle of the original signal, as shown in Figure (5-6), before transforming into time domain. Symbol by symbol causes no inter-symbol interference.

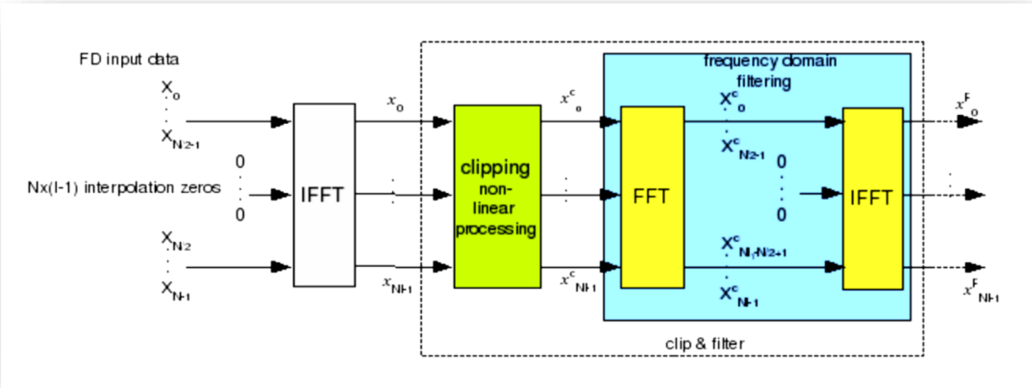


Figure 5-6: FFT-IFFT filter block diagram for clipping and filtering method [46] .

After clipping the signal, the out-of-band radiation caused by clipping falls in the zeros and then the signal is passed to the FFT filter. The FFT function transforms the clipped signal x_k^c to frequency domain yielding X_n^c . [47] The information components of X_n^c are passed unchanged to the IFFT block and the out-of-band radiation that fell in the zeros is set back to zero. The IFFT block of the filter transforms the signal to time domain and the obtained signal x_k^F is passed to the D/A converter. After filtering, the signal suffers peak regrowth so in order to minimize this effect, the clip & filter process can be repeated several times.

The zeros added in the transmitter are removed in the receiver once the signal is in frequency domain. Figure (5-7) shows a simplified block diagram of the OFDM system including the FFT filter so that it can be seen where the zeros are added and removed.

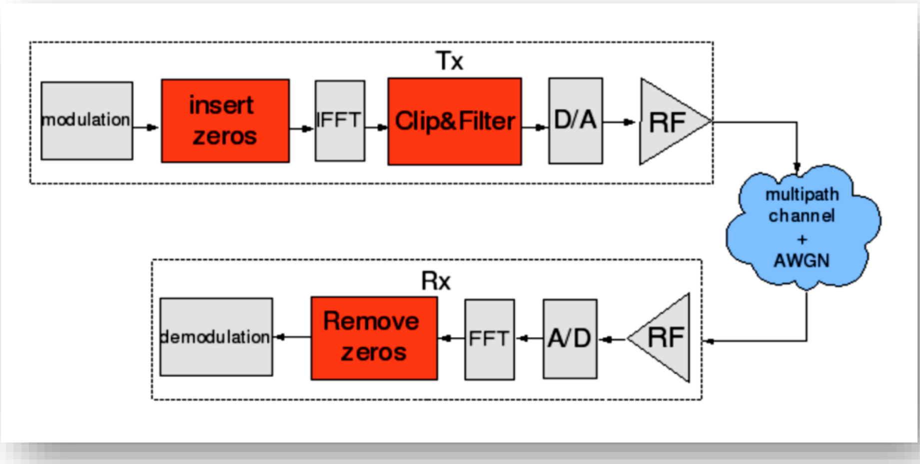


Figure 5-7: OFDM system with clipping and filtering method block diagram.

After distorting the signal in the transmitter, the goal is to recover it at the receiver with the minimum possible number of errors.

5.5 Power Spectral Density.

In this section, our focus is on the spectral splatter caused by clipping and the effect of filtering. The PSD is measured for each OFDM block and then averaged over many blocks to eliminate the effects of the rectangular time window. [11], In Figure (5-8), we show the PSD of the clipped signal with Clipping Ratio of CR =1.4.

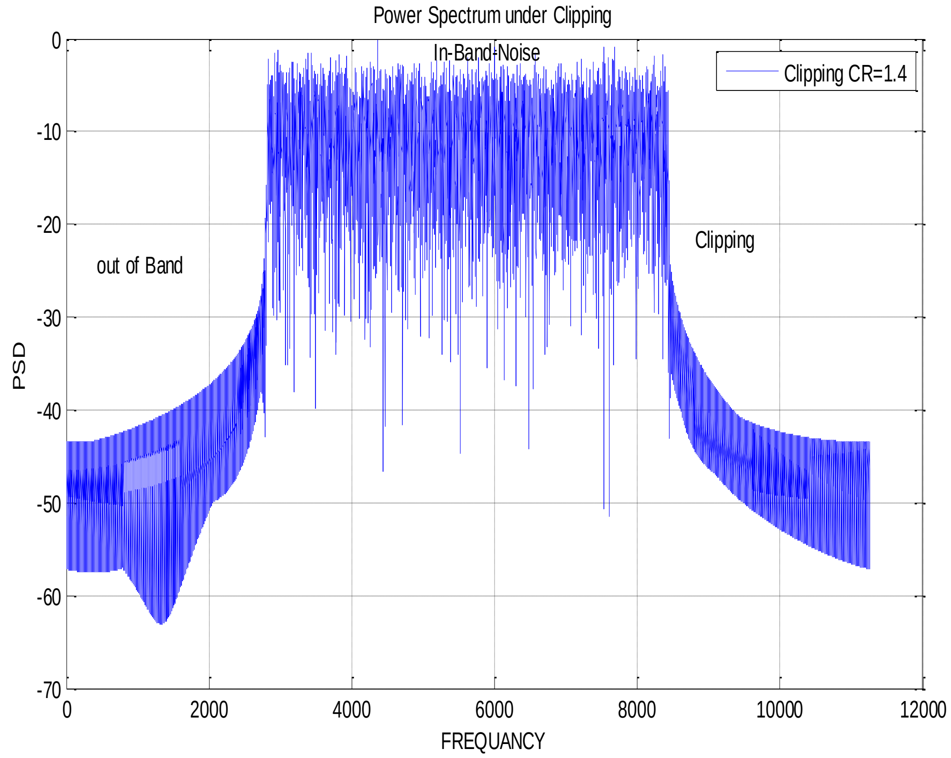


Figure 5-8: Power spectral density of clipped OFDM signals.

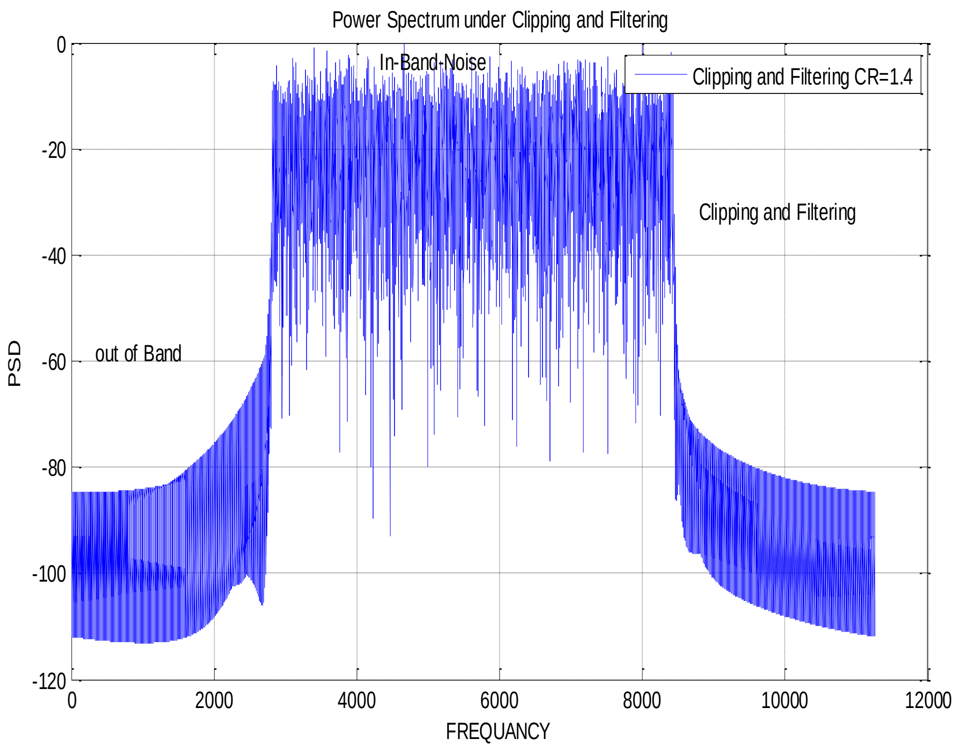


Figure 5-9: Power spectral density of the clipped-and-filtered OFDM signals with a clipping ratio of CR=1.4 .

The in-band signal attenuation as well as the out-of-band caused by clipping is evident. For CR=1.4 , the out-of-band noise emission power is only 18 dB lower than the signal power. This shows that filtering is necessary to suppress the spectral splatter caused by clipping. From Figure. (5-9), it can be seen that filtering greatly attenuates the out-of-band Emission. With CR=1.4 , the spectral side lobes after filtering are now at least 50 dB lower than the signal main lobe. Ripple caused by the digital filter (FFT-IFFT filter) may boost the power of some subchannels while suppressing others.

5.6 Peak-to-Average Power Ratio Reduction Using Distortion less Methods.

5.6.1 Selective Level Mapping Technique (SLM).

SLM is a type of multiple signal representation (**MSR**), which is non transparent method. In the **SLM** technique, the transmitter generates a set of sufficiently different candidate data blocks, all representing the same information as the original data block, and selects the most favorable for transmission [48,49]. A block diagram of the **SLM** technique is shown in Figure (5-10).

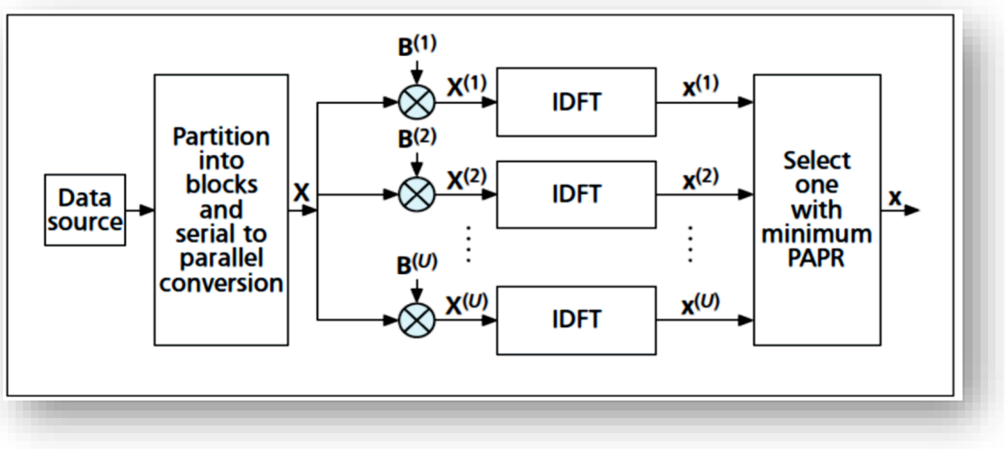


Figure 5-10 :A block diagram of the SLM technique [49].

Each data block is multiplied by U different phase sequences, each of length N, $\mathbf{B}(u) = [b_{u,0}, b_{u,1}, \ldots, b_{u,N-1}]^T$, $u = 1, 2, \ldots, U$, $b_u = e^{j\varphi_u}$, where φ_u Phase Factor, $\varphi_u \in (0,2\pi]$ results in U modified data blocks to include the unmodified data block in the set of modified data blocks. $\mathbf{B}(1)$ is set as the all-one vector of length N. Let denote the modified data block for the u-th phase sequence $\mathbf{X}(u) =[X_0 b_{u,0}, X_1 b_{u,1}, \ldots, X_{N-1} b_{u,N-1}]^T$,u = 1, 2,…,U. After applying **SLM** to **X**, the multicarrier signal becomes:

$$x(t)^{(u)} = 1/\sqrt{N} \sum_{n=0}^{N-1} X_n . b_{u,n} . e^{j2\pi\Delta f t} \quad ,0 \le t \le NT \quad ,u=1,2,3,\ldots,U \quad \ldots(5\text{-}6)$$

Among the modified data blocks $\mathbf{X}(u)$, u = 1, 2,…, U, the one with the lowest PAPR is selected for transmission. Information about the selected

phase sequence should be transmitted to the receiver as side information.

At the receiver, the reverse operation is performed to recover the original data block. For implementation, the **SLM** technique needs U **IDFT** operations , and the number of required side information bits is

denotes the smallest integer that does not exceed y. This approach is applicable with all types of modulation and any number of subcarriers. The amount of PAPR reduction for **SLM** depends on the number of phase sequences U and the design of the phase sequences. In [50] , SLM technique without explicit side information is proposed.

Example:[51] Here, a simple example of the **SLM** technique is shown for an OFDM system with eight subcarriers. The number of phase sequences is set to $U = 4$. The data block to be transmitted is denoted by $\mathbf{X} = [1, -1, 1, 1, 1, -1, 1, -1]^T$ .Whose PAPR before applying **SLM** is 6.5 dB. The four phase factors are set as $\mathbf{B}(1) = [1, 1, 1, 1, 1, 1, 1, 1]^T$, $\mathbf{B}(2) = [-1, -1, 1, 1, 1, 1, 1, -1]^T$, $\mathbf{B}(3) =[-1, 1, -1, 1, -1, 1, 1, 1]^T$, and

B(4) = [1, 1, –1, 1 , 1, –1, 1 , 1]^T. Among the four modified data blocks **X**(***u***), ***u*** = 1, 2, 3, 4, **X**(2), has the lowest PAPR of 3.0 dB. Hence, **X**(2) is selected and transmitted to the receiver. For this data block, the PAPR is reduced from 6.5 to 3.0 dB, resulting in a 3.5 dB reduction. In this case, the number of **IDFT** operations is 4 and the amount of side information is □$\log_2 4$□=2bits. The amount of PAPR reduction may vary from data block to data block, but PAPR reduction is possible for all data blocks.

5.6.2 Partial Transmit Sequence Technique (PTS).

PTS is another type of multiple signal representation (**MSR**), which is non transparent method. In the **PTS** technique, an input data block of ***N*** symbols is partitioned into disjoint subblocks or cluster. The subcarriers in each subblock are weighted by a phase factor for that subblock. The phase factors are selected such that the PAPR of the combined signal is minimized. [49,52,53].

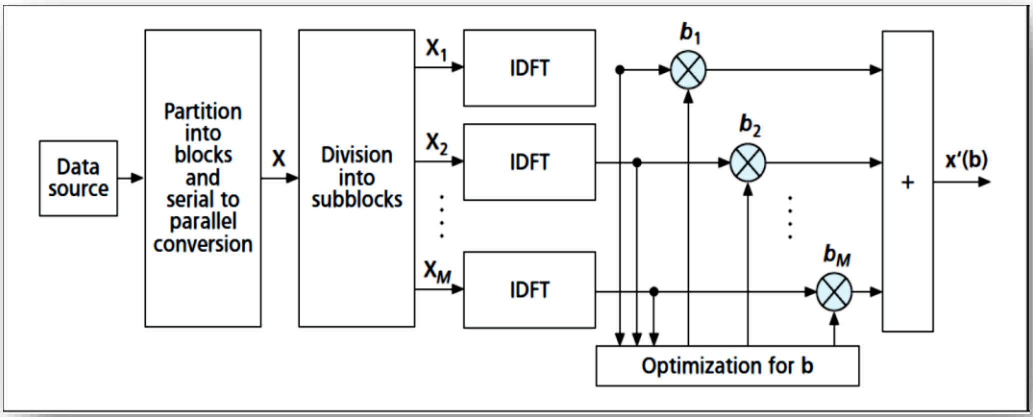


Figure 5-11: A block diagram of the PTS technique [49].

Figure (5-11) shows the block diagram of the **PTS** technique. In the ordinary **PTS** technique [49,52] input data block **X** is partitioned into M disjoint subblocks, $\mathbf{X}_m = [X_{m,0}, X_{m,1},., X_{m,N-1}]^T$, $m = 1, 2,\ldots, M$, such that

$$\sum_{m=1}^{M} \mathbf{X}_m = \mathbf{X} \quad . \qquad \ldots\ldots\ldots(5\text{-}7)$$

and the subblocks are combined to minimize the PAPR in the time domain. The L-times oversampled time domain signal of $\mathbf{X}_m$, $m = 1, 2, \ldots, M$, is denoted $\mathbf{x}_m = [x_{m,0}, x_{m,1},\ldots, x_{m,NL-1}]^T$, $\mathbf{x}_m$, $m = 1, 2, .., M$, is obtained by taking an **IDFT** of length NL on $\mathbf{X}_m$ concatenated

with $(L - 1)N$ zeros. These are called the partial transmit sequences.[49]

Complex phase factors, $b_m = e^{j\phi_m}$, $m = 1, 2, \ldots,$ M , where $\phi_m \in (0,2\pi]$ are introduced to combine the **PTSs**. The set of phase factors is denoted by a vector $\mathbf{b} = [b_1, b_2,\ldots, b_M]^T$. The time domain signal after combining is given by:

$$\mathbf{x}`(\mathbf{b}) = \sum_{m=1}^{M} b_m . \mathbf{x}_m \qquad \ldots\ldots (5\text{-}8)$$

where $\mathbf{x}`(\mathbf{b}) = [x`_0(\mathbf{b}), x`_1(\mathbf{b}), \ldots\ldots x`_{NL-1}(\mathbf{b})]^T$. The objective is to find the set of phase factors that minimizes the PAPR. Minimization of PAPR is related to the minimization of:

$$\max \ | \ x`_k(\mathbf{b}) \ |$$

In general, the selection of the phase factors is limited to a set with a finite number of elements to reduce the search complexity. The set of allowed phase factors is written as **b** = $\{e^{j2\pi l/W}\}$, l = 0, 1, …, W – 1, where W is the number of allowed phase factors .In addition, b_1=1can be set without any loss of performance .So an exhaustive search for(M –1)phase factors should be performed. Hence, W^{M-1}sets of phase factors are searched to find the optimum set of phase factors, and this optimization block is shown in Figure (5-12) The search complexity increases exponentially with the number of subblocks M.PTS needs ***M*** **IDFT** operations for each data block, and the number of required side information bits is$\lceil \log_2 W^{M-1} \rceil$.The amount of PAPR reduction depends on the number of subblocks M and the number of allowed phase factors W .

Another factor that may affect the PAPR reduction performance in **PTS** is the subblock partitioning, which is the method of division of the subcarriers into multiple disjoint subblocks. There are three kinds of subblock partitioning schemes: adjacent, interleaved, and pseudo-random

partitioning [13]. Among them, pseudo-random partitioning has been found to be the best choice. The PTS technique works with an arbitrary number of subcarriers and any modulation scheme. As mentioned above, the ordinary **PTS** technique has exponentially increasing search complexity. To reduce the search complexity, various techniques have been suggested. In[53], iterations for updating the set of phase factors are stopped once the PAPR drops below a preset threshold. In [53] ,various methods to reduce the number of iterations are presented. These methods achieve significant reduction in search complexity with marginal PAPR performance degradation. Formation of alternative Sequence

Example: [52]Here, example of the **PTS** technique for an OFDM system is seen with eight subcarriers that are divided into four subblocks. The phase factors are selected in $\boldsymbol{b} = \{\pm 1\}$. Figure (5-13) shows the adjacent subblock partitioning for a data block **X** of length 8.

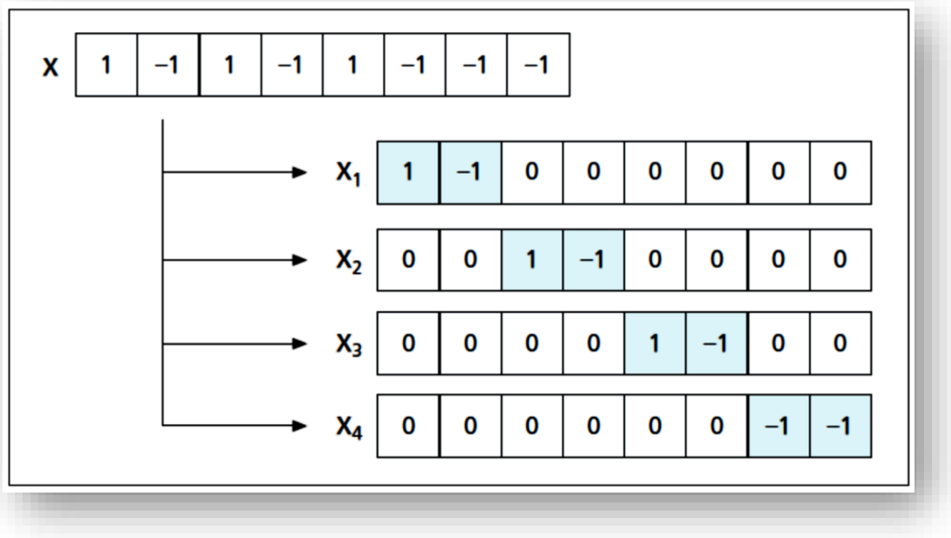


Figure 5-13:An example of adjacent subblock partitioning in PTS.

The original data block **X** has a PAPR of 6.5 dB. There are 8 (= 2^{4-1}) ways to combine the subblock with fixed $\boldsymbol{b}_1 = 1$ among them $[\boldsymbol{b}_1, \boldsymbol{b}_2, \boldsymbol{b}_3, \boldsymbol{b}_4]^T = [1, -1, -1, -1]^T$ achieves the lowest PAPR. The modified data block will be:

$$\mathbf{X} = \sum_{m=1}^{M} \boldsymbol{b}_m . \mathbf{X}_m = [1, -1, -1, 1, -1, 1, 1, 1]^T$$

in which PAPR is 2.2 dB, resulting in a 4.3 dB reduction. In this case, the number of required **IDFT** operations is 4 and the amount of side information is $\log_2 2^{4-1}=3$ bits. The side information must be transmitted to the receiver to recover the original data block. One way to do this is to transmit these side information bits with a separate channel other than the data channel. It is also possible to include the side information within the data block; however, this results in data rate loss.

5.7 Bits Redundancy in an OFDM Symbol Rap

OFDM symbols with a high PAPR are relatively unlikely, the following argumentation is justified. An ideal PAPR reduction scheme would spend introduced redundancy to exclude the " bad " OFDM symbols from transmission. R_{aP} redundant ("anti PAPR") bits allow to suppress a fraction of $(1-2^{-RaP})$ from the entire set of possible OFDM symbols [13].

Therefore, the Equation $(1-2^{-RaP})=\Pr(\text{PAPR} > \text{PAPR0})$ gives the trade -off between redundancy and theoretically achievable maximum PAPR Regarding and solving for ξ_0 yields) .Where α =1 for simplicity.

$$\text{CCDF} = \Pr(\xi \geq \xi_0) \approx 1-(1-exp(-\xi_0))^N$$

$$\Pr(\xi \geq \xi_0) = 1-2^{-RaP}$$

$$1-(1-exp(-\xi_0))^N = 1-2^{-RaP}$$

$$\text{PAPR0} = \xi_0 = -\ln\left(1-2^{-RaP/N}\right) \quad . \qquad ....(5\text{-}9)$$

This Equation can be plotted in Figure (5-14).

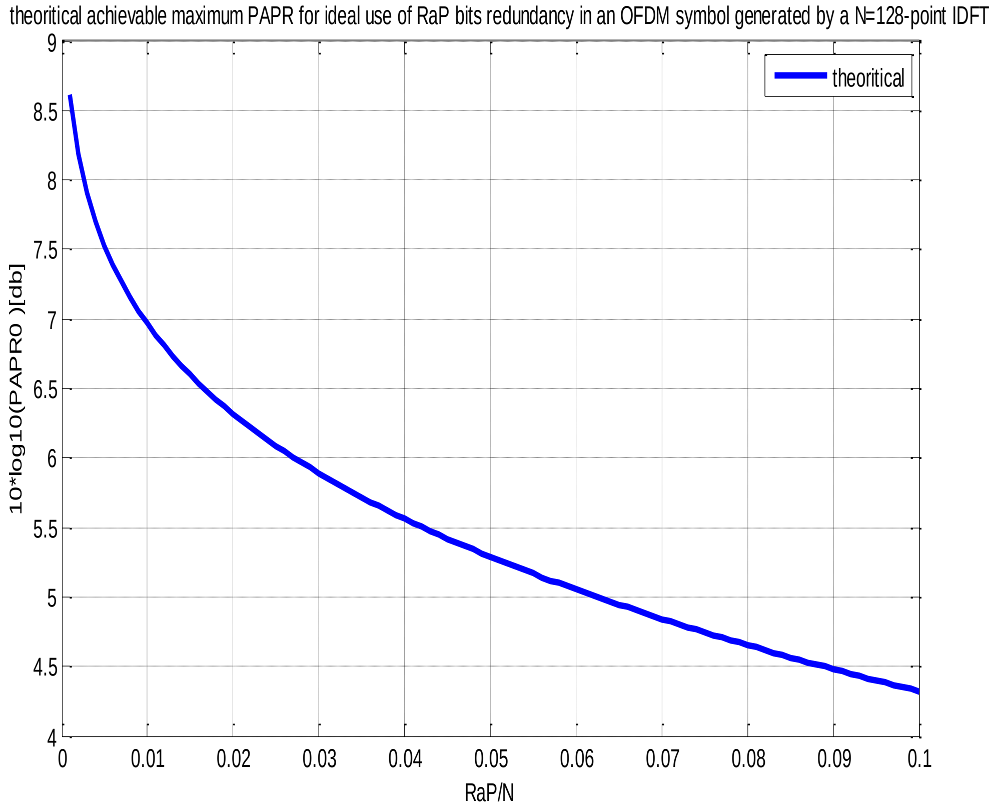


Figure 5-14: Theoretically achievable maximum PAPR for ideal use of R_{ac} bits

5.8 Average-Power-Increasing Technique.

Initially, it is intuitive to think that increasing the average power is a good thing, and it may be if the increase goes towards enlarging the minimum distance between constellation points. In that case, the error rate will be decreased. However, all of these methods increase the average power *without* changing the minimum distance of the constellation [17]. The problem with this is illustrated in Figure(3-15), where a peak-power constraint of 4W is assumed. In the top of the figure is a process for an average-power-preserving PAPR technique. There, the "scale signal" block does not actually do anything because the constraint has already been met. On the other hand, for the average-power-increasing technique, after the PAPR reduction is done, the average power has increased. Since the PAPR is a ratio of peak to average power, the PAPR contains only relative information about the peak power. In a peak-power constraint system, the absolute peak power is of interest. The example given in demonstrates an average power increase of 10%, after which the peak is scaled to meet the peak-power constraint. This means that the effective signal power is 1 /1.1 = 90.9% of the effective signal power of a signal passing through an average-power-preserving reduction technique. Small increases in average power may not be significant, but perspective of PAPR-reduction schemes that promise large PAPR reductions at the cost of decreased minimum distances is of importance. Interestingly, none of the papers in this area has examined the point of diminishing return in the algorithms. That is, at what point does the average power increase outpace the PAPR reduction?

Another consideration with the power increasing systems is the sidelobe enlargement. Basically, the power increase has to be placed somewhere in the bandwidth of the OFDM system. If too much extra energy is placed in subcarriers at the edge of the OFDM band- width, then the sidelobes will become larger and the system may no longer meet regulatory spectral masks. PAPR reduction with average power constant:

5.8.1 Tone Reservation(TR) for Average-Power-Increasing Technique for PAPR Reduction.

The tone reservation (TR) algorithm was developed by Tellado [17] in 2000, which is transparent method. Whereby a small number of sub-carriers (tones) are reserved to create a signal which can cancel the high peaks in the information-carrying signals at the transmitter. This approach can reduce the PAPR of the OFDM signals without introducing any additional distortions to the information data and does not require side information. However, TR can have a high computational cost due to the difficulties of finding an effective cancellation signal in the time domain from only a small number of reserved tones in the frequency domain. In this section, a novel approach is given for overcoming this difficulty by creating a Gaussian-pulse-like cancellation signal which facilitates a simple procedure for reducing peak values and minimizing the occurrence of secondary peaks.

5.8.2 Advantages of Tone Reservation (TR).

TR method has the following advantages[54]:

1) No need for side information

2) Less complex. Just one-time IFFT operation is needed. But multiple iteration operations are needed after IFFT operation.

3) No special receiver operation is needed.

5.8.3 Overview of Tone Reservation.

In Tone Reservation, a small number of sub-channels (tones), which do not carry any information data, are reserved for peak cancellation. This restricts the data-bearing vector $\boldsymbol{X}$, and the reserved tone vector $\boldsymbol{C}$ to lie in disjoint frequency subspaces [17].

TR is a type of Pilot tones method. One approach for PAPR reduction is to search an additive corrective signal $\boldsymbol{c}$ in order to have PAPR($\boldsymbol{x}$ + $\boldsymbol{c}$) < PAPR($\boldsymbol{x}$). One way to find $\boldsymbol{c}$ is to use optimization approaches. The signal addition is achieved in frequency domain as it can be seen in Figure(5-15) where corrective carriers are added in between the useful data carriers. Thus, the PAPR of the resulting signal ($\boldsymbol{x}$ + $\boldsymbol{c}$) is given by:

$$\text{PAPR} = \max_{0 \le k \le N-1} |x_k + c_k|^2 \,/\, E[|x+c|^2]. \qquad \text{....... (5-10)}$$

In reality, the ideal method to reduce PAPR would be to minimize the peak of the combined signal (***x + c***)while keeping the average power constant.

$$X_K + C_K = \begin{cases} C_K & , K \in R \\ X_K & , K \in R^C \end{cases} \quad \text{…. (5-11)}$$

where R^C is the complement of R in N which represents information carriers. This method restricts the data vector X , and the peak reduction vector C to lie in disjoint frequency subspaces, i.e., $X_k = 0$, $k \in \{i_1,\ldots,i_R\}$ and $C_k = 0$, $k \notin \{ i_1 , \ldots, i_R \}$. This formulation is distortion less and leads to very simple decoding of the data subsymbols that are extracted from the received sequence by choosing the set of values

$k \notin \{i_1,\ldots,i_R\}$at the receiver **FFT** output. Moreover, it allows simple optimization techniques for the computation of the peak reduction vector c . The R nonzero values in C will be called ***peak reduction tones*** ,as shown in Figure(5-15).

Let one assumes that the R tones $\{i_1,\ldots,i_R\}$have been fixed at the beginning of the transmission and that they won't be changed until the transmission is over or some new information about the channel is fed back to the transmitter.

Then C is selected to meet the following targets:

1. X can be decoded efficiently from X+C without degrading the performance.
2. **PAPR** can be reduced by optimizing c.

3. c can be computed efficiently.

First C is constructed to nonzero values over a subset $\{i_1, i_2, \ldots, i_R\}$, where $R << N$,i.e.

$$C_k = \begin{cases} C_k, & k \in \{i_1, \ldots, i_R\} \\ 0, & \text{elsewhere}. \end{cases} \qquad \ldots\ldots(5\text{-}12)$$

The values of X_k, $k \in \{i_1, \ldots, i_R\}$, are set to zero , such that the sets of nonzero values for X and C are mutually exclusive. The subset of reserved tones is denoted by $R=\{i_1,\ldots,i_R\}$,where N represents the set of all tones in the multi-carrier symbol. The addition of these reserved tones c to a data-bearing signal x produces a new composite signal

$$\bar{x}_n = x_n + c_n = IFFT\,[X_K + C_K] \qquad \ldots.\ (5\text{-}13)$$

Since symbol demodulation is performed in the frequency domain on a tone-by-tone basis, the reserved sub-channels can be discarded at the receiver, and only the data-bearing subchannels are used to determine the transmitted bit stream. The new PAPR becomes:

$$PAPR\,(\bar{x}) = 10\log_{10}\left(\frac{max\,|\,x_n + c_n|^2}{E\,[|\,x_n\,|^2]}\right). \qquad ..(5\text{-}14)$$

where it can be seen that the PAPR can be reduced by optimizing c_n

To minimize the PAPR of $x_n + c_n$ the vector c_n^* must be computed that minimizes the maximum peak value, so that $max\ |x_n+c_n|^2$ becomes smaller than $max\ |x_n|^2$.

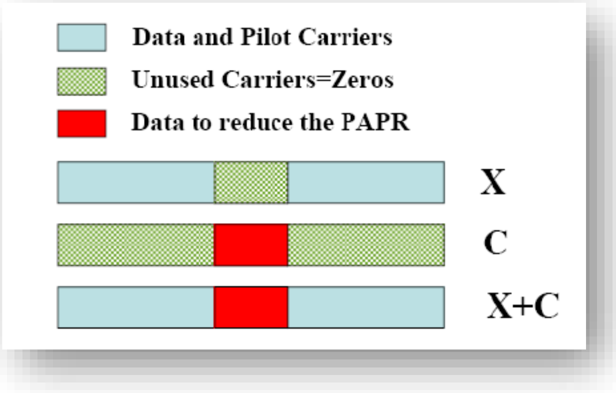


Figure 5-15: Principle of adding corrective subcarriers

5.8.4 Data Rate Loss.

Setting of some values of ***X*** to zero causes a data rate loss. Data rate loss (R_{DL}) can be expressed as :

$$R_{DL} = R / N \quad .\qquad \ldots\ldots(5\text{-}15)$$

To reduce the data loss , ***R*** should be minimized .Demodulation is very simple at the receiver. No side information is needed as the receiver

knows the PAPR reduction subcarrier positions beforehand [55]. At the receiver, simply these subcarriers are disregarded and other subcarriers are demodulated .

A PAPR reduction of 6 dB is needed with R/N=5% at CCDF less

than 10^{-5} while at 10 dB reduction of PAPR, R/N= 20% has been needed.

5.8.5 The Gradient Algorithm.

The gradient algorithm is one of the good solutions to compute ***c**** with low complexity. The basic idea of the gradient algorithm comes from clipping. Clipping the peak tone to the target clipping level can be interpreted as subtracting impulse function from the peak tone in time domain.

Impulse function is time shifted to the peak tone location, and scaled so that the power of the peak tone should be reduced to the desired target clipping level. But this operation affects the whole value of OFDM symbol in frequency domain, i.e., not only ***C*** but also ***X*** is changed. So another impulse-like function is designed, which only has the value in the tone locations $\{i1,\ldots,iR\}$. [55,56].

5.8.6 peak reduction kernel.

Let $P_k = 1$, $k \in \{i_1,\ldots,i_R\}$ and $Pk = 0$, $k \notin \{i_1,\ldots,i_R\}$ and let **IFFT** be output of **P** be **p** , $\mathbf{p} = [\ p_0\ p_1\ p_2\ \ldots p_{N-1}]$ is the **IFFT** output of the vector whose value is 1 at the tone locations $\{i_1,\ldots,i_R\}$. , and 0 elsewhere. **p** is called ***peak reduction kernel*** and is only a function of the tone locations $\{i_1,\ldots,i_R\}$.. Therefore, one needs to calculate the kernel **p** at the beginning of the transmission. **p** has its peak at the location p_0 but **p** also has the leakage at the location $p_1 \ldots p_{N-1}$. As the number of the reserved tone ***R*** becomes larger, peak at the location p_0 gets larger and the leakage at the location $p_1 \ldots p_{N-1}$ gets smaller. So, the performance gets better, but redundancy increases and the throughput decreases.

The gradient algorithm is an iterative clipping algorithm using peak reduction kernel. When **p** is circular shifted, scaled, and phase rotated in time domain, the values of **P** in the tone locations $\{i_1,\ldots,i_R\}$ are changed but the other tones remain unchanged. So, data vector $\boldsymbol{X}$ isn't affected by iterative clipping operations. [57,58]

The optimization is done on the time domain code $\boldsymbol{c}$ **.** So, only one **IFFT** operation is needed and the complexity is very low. The gradient algorithm can be expressed , as follows for :

$$\boldsymbol{c}^{(k+1)} = \boldsymbol{c}^{(k)} - \alpha_k \, \mathbf{p}[((\boldsymbol{n}-\boldsymbol{n}_k))_N] \qquad \text{……. (5-16)}$$

$$\boldsymbol{n}_k = \underset{\boldsymbol{n}}{\mathbf{Arg\ max}} \, | \, x_n + c_n^{(k)} | \qquad \text{……. (5-17)}$$

where α_k is a scale and phase rotation factor depending on the maximum peak found at iteration $\boldsymbol{k}$. The notation $\mathbf{p}[((\boldsymbol{n} - \boldsymbol{n}_k))_N]$ means that the kernel has been circularly shifted in time by a value of $\boldsymbol{n}_k$.

This optimization problem is convex in the variable $\boldsymbol{C}$ and can be easily cast as a linear program(LP) ,having $2\boldsymbol{R}+1$ unknowns and $2\boldsymbol{N}$ inequalities. As this problem is LP structured [17]. This algorithm has a complexity of $\mathbf{O}(\boldsymbol{N})$ order. [55]

This kernel has its maximum in the time domain at $\boldsymbol{n} = 0$ and its aim is to decrease the high peak found at $\boldsymbol{n}_k$, without increasing the other values of the OFDM symbol at $\boldsymbol{n} \neq \boldsymbol{n}_k$ too much. So, the selection of the tone location $\{\boldsymbol{i}_1,\ldots,\boldsymbol{i}_R\}$ is critical point of the PAPR reduction performance. A pertinent choice for **p** and; therefore, for the reserved tones is obtained by minimizing its secondary peak**.**

Figure(5-16) shows the structure of the OFDM system transmitter using conventional scheme. $\boldsymbol{R}$ tones are reserved for PAPR reduction and

$\boldsymbol{N} - \boldsymbol{R}$ tones are assigned for data information. All tones are allocated according to predetermined tone locations $\{\boldsymbol{i}_1,\ldots,\boldsymbol{i}_R\}$. Then, **IFFT** is executed, and the gradient algorithm is operated. Figure (5-17) shows the detail procedure of the gradient algorithm.

When new **IFFT** output ***x*** is entered, the peak position and value of ***x*** are detected. Then , peak reduction kernel is circular shifted to the peak position, scaled and phase rotated.

The resulting kernel is subtracted from ***x*** and then PAPR is calculated. If the number of iterations reaches predetermined maximum iteration number, control escapes the process and the resulting signal is transmitted, if no clipping operation is executed iteratively. [57]

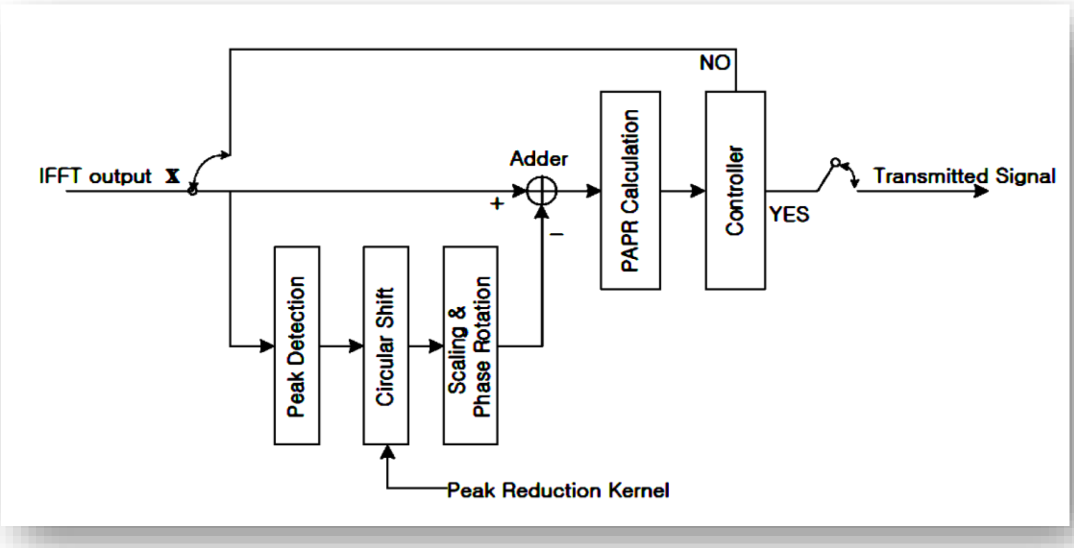


Figure 5-17: Procedure of Gradient Algorithm. [57]

5.8.7 BW Effect.

Bandwidth of the added signal affects the performance of PAPR reduction scheme. The more bandwidth allocated for added signal the more will be the PAPR reduction as more tones can be used to reduce PAPR but at the same time the average power of the added signal increases. Also, it is not easy to find free bandwidth to transmit the added tones. On the contrary, if transmitted over useful band, the out-of-band interference will be produced. [56]

5.8.8 Position Effect.

Adding tones' position affects a lot the PAPR reduction performance. The tones nearer to the useful carriers reduce more PAPR than the farther tones. [56]

5.8.9 Power Effect.

It is not always possible to use the bandwidth closer to the useful band. Thus, in order to achieve the same performance as with the closer tones, the power of the tones at larger distances must be increased. [56]

5.8.10 Modern Method for Tone Reservation by Using Generation of Gaussian Pulses for Reserved Tones[18].

The proposed method is developed in 2007 by Carole A. Devlin, [18]. In the tone reservation approach, a small number of subcarriers (tones) are reserved to create a signal which cancels the high peaks of information-carrying signals in the transmitter. This cancellation signal must be generated in the frequency domain using the minimum number of tones to maximize data throughput. However, it is also preferable to have a narrow time domain signal to prevent the generation of secondary peaks. In other words, the PAPR reduction approach in tone reservation is a constrained signal-design problem: a signal must be designed in the frequency domain, but its effect is evaluated in the time domain. In current tone reservation approaches, the cancellation signal is mainly generated from either trial and error processes or involves computationally complex optimization procedures [18,59]. Carole A. Devlin proposed a simple algorithm, in which a Gaussian window-like signal is employed in the frequency domain to form the canceling pulse in the time domain. Since Gaussian pulses can be optimized in both the time domain and the frequency domain [18] by selecting a small number of tones in the frequency domain, a narrow pulse can be generated in the time domain, as shown in Figure(5-18).

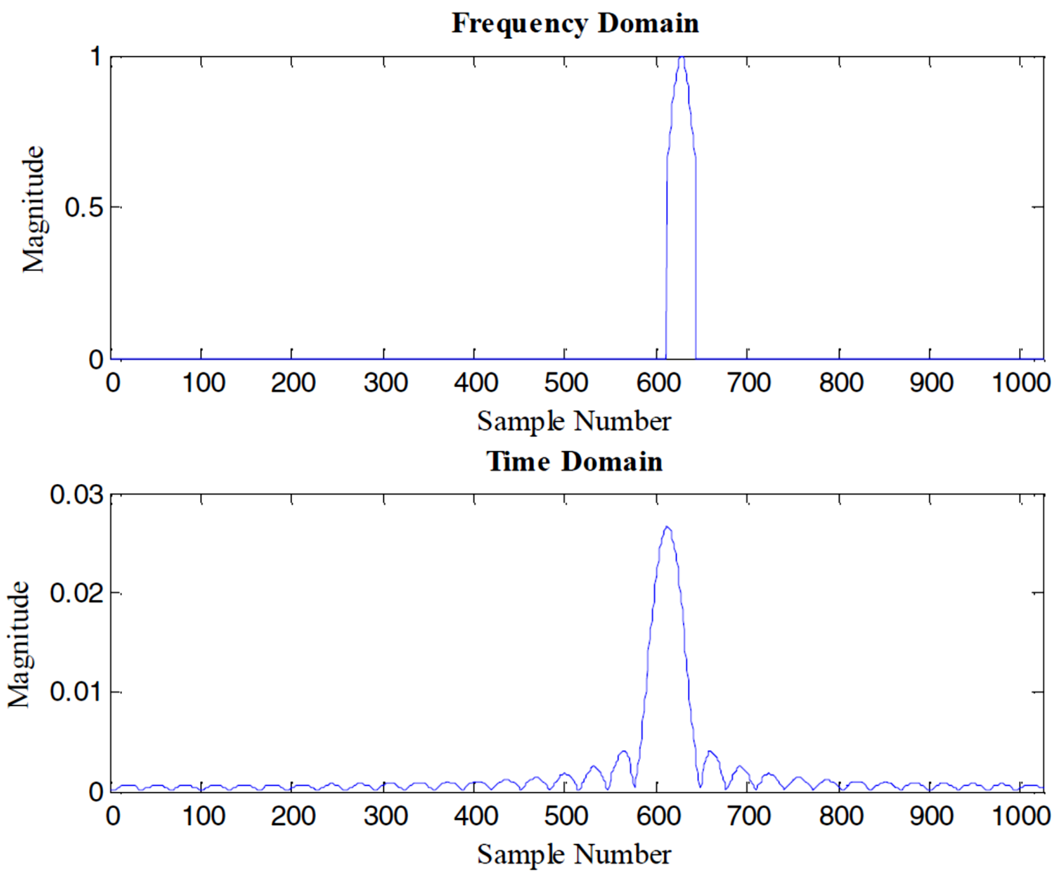


Figure 5-18: Cancellation signal formed from Gaussian pulse in the time and the frequency domain.[18]

The coefficients of the Gaussian window, $\boldsymbol{G}$, can be calculated from the equation

$$G_{[m+1]} = e^{R/2} \qquad .\text{Where} \quad 0 < m < R-1 \quad ..(5\text{-}18)$$

where α_s represents the reciprocal of the standard deviation, and the width of the window is inversely related to α_s. These values represent the amplitude of the Gaussian window and the phase has a value of zero. The value of $\boldsymbol{R}$ represents the number of tones reserved to generate the cancellation signal. Typically, a value of 16 or 32 provides sufficiently narrow signals in the time domain to avoid the occurrence of secondary peaks. Unlike previously proposed methods, the generation of this cancellation signal does not require complicated peak searching or optimization procedures [18,59,60,61]. This peak signal is easily optimized both in the time-domain and the frequency-domain. Just one-time IFFT operation is required, and very few tones are needed.

5.4.11 Algorithm for the Improved Method to Reduce PAPR for Tone Reservation (TR) .

Once an efficient cancellation signal is obtained, a fast conversion algorithm can be applied to the OFDM system to cancel the high peaks so that the transmitted signal does not exceed the required threshold $\boldsymbol{A}$**.** The algorithm employed is implemented, as follows:[18].

I. A pre-defined cancellation signal is generated by using the Gaussian pulses described below. This cancellation signal only has non-zero values in the reserved tone locations in the frequency domain and has one sharp peak in the time domain.

II. It must be checked if there are peaks exceeding the required threshold

$\boldsymbol{A}$ in the information-carrying signal and if, the magnitude of the peaks and their corresponding location are detected.

III. For each peak detected in the information data, the peak of the pre-defined cancellation signal is circularly shifted to the peak location and scaled by the value of the difference between the peak and the threshold so that the power of the peak tone can be reduced to the desired target level. All of the appropriately scaled and phase shifted cancellation signals are then subtracted from the original information signal.

IV. After the peak cancellation, the composite signal will be detected again since some secondary peaks may appear during the previous peak-canceling operation. The process is continued until all the peaks are

below the required threshold or until a maximum number of iterations is exceeded, as shown in Figure(5-19).

V. The new time domain signal with reduced PAPR at iteration $\boldsymbol{i}$ can be expressed as: $\bar{x}_{i+1}[n] = \bar{x}_i[n] - \alpha_i \, \mathbf{p}\,[(n-n_i)_N]$.

The cancellation signal ci at iteration $\boldsymbol{i}$ is represented by $\alpha_i \, \mathbf{p}[(n-n_i)_N]$

where α_i represents the scaling factor and phase rotation applied for peak cancellation and $\mathbf{p}[(n-n_i)_N]$ denotes the necessary circular shift of the

cancellation signal so that the peaks are cancelled.

Since the Gaussian-pulse based peak cancellation signal has a very sharp shape in the time domain, the occurrence of secondary peaks is minimized. Typically, only a single iteration is required to remove each peak detected in the original time-domain signal. In this approach, when the pulse cancellation signal is circularly shifted, scaled and phase rotated

in the time-domain, the values of the frequency domain signal . only change at the reserved tone locations, but remain unchanged at the other tones. The data vector ***X*** is not affected by the peak canceling operations, so that it does not need any side information or any receiver operation.

A sample of the time domain signal is shown in Figure (5-20), where it can be clearly seen that, the high peaks are effectively removed after three iterations.[18].

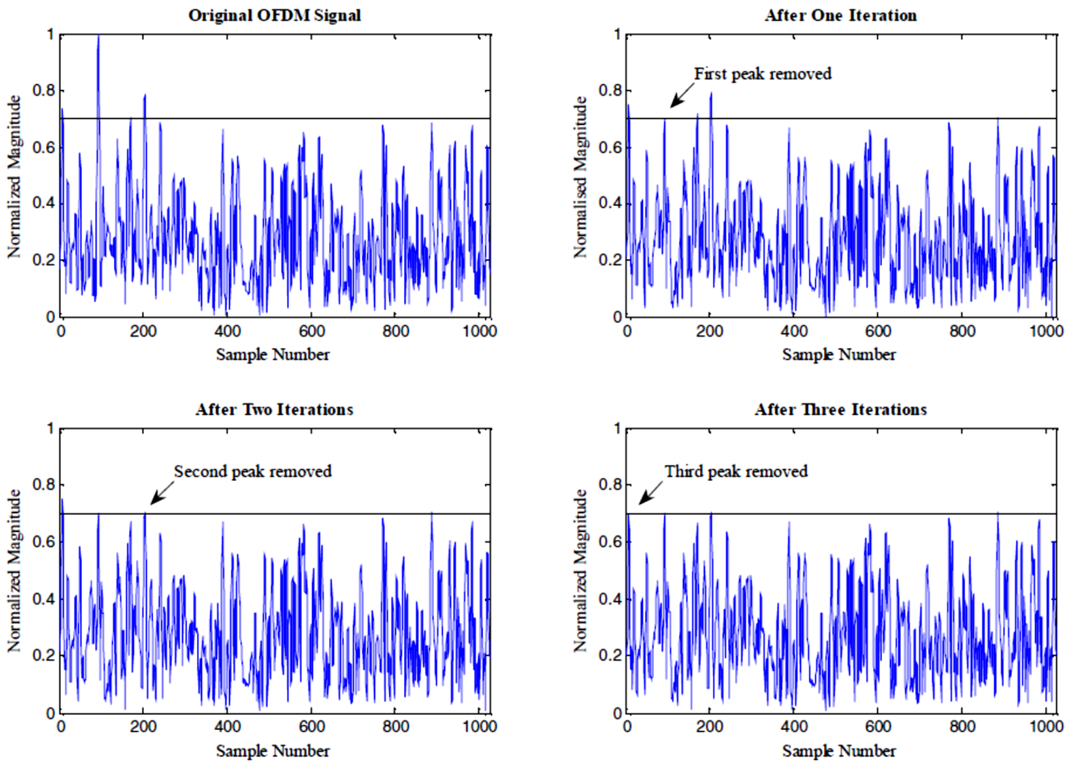


Figure 5-20: Time domain sample of a WiMax signal with proposed improved Tone Reservation method. [18]

Chapter Six

6.1 Conclusion.

Peak to Average Power Ratio (PAPR) is the main real problem in OFDM system which reduces the power amplifier efficiency. Clipping and Filtering which is a simple technique but causes distortion to the signal. The influence of the out-of-band can be reduced by using filtering after clipping, while filtering which also causes some peak regrowth. Non-distortion clipping technique comes to eliminate this problem where filters not be use beside clipping and then there is no distortion happened only noise can be covered by using any simple coding as error correction .the results shows there is a good benefits of using this technique .the table below shows summarizes the difference between conventional clipping and proposed technique (Non-Distortion clipping).

Table 1: Comparison the Characteristics of PAPR Reduction Techniques between Conventional Clipping and Proposed Technique (Non-Distortion Clipping).

PAPR Reduction Techniques	**Distortion less**	**Cost Less**	**Complexity Less**	**PAPR reduce**	**PSD attenuate**
Clipping and Filtering	No	No	No	Good	High
Proposed Technique	Yes	Yes	Yes	Better	Low

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