



COMPARATIVE STUDY OF PAPER REDUCTION TECHNIQUES OF OFDM SYSTEM

S.Anu, II M.E. Applied Electronics, Akshaya College of Engineering and Technology, Coimbatore

sara.anu.77@gmail.com

Dr.J.Jaya, Principal, Akshaya College of Engineering and Technology, Coimbatore

Abstract-The OFDM technique is an attractive modulation technique for transmitting large amounts of data over radio waves. One major disadvantage of OFDM is that the time domain OFDM signal which is a sum of several sinusoids leads to high peak to average power ratio (PAPR). In this paper, a joint Companding transform, Amplitude clipping and filtering, Partial transmit sequence (PTS) technique, Selected Mapping technique (SLM) and Hadamard transform method are proposed to reduce peak-to-average of OFDM signal for 64 subcarriers. Significant PAPR reduction and good performance in the BER is expected from the proposed system when compared to other PAPR reduction techniques. We use MATLAB software to analyze the system. The performance of the system is analyzed from BER vs. SNR graph. PAPR reduction is analyzed using Complementary Cumulative Distribution Function (CCDF) plots.

Keywords-PAPR, OFDM, CCDF, SLM, PTS

I. INTRODUCTION

Orthogonal Frequency Division Multiplexing (OFDM) is a popular modulation scheme that is used in wireless LAN standards like 802.11a, g, HYPERLAN/2 and in the Digital Video Broadcasting standard (DVB-T). It is also used in the ADSL standard, where it is referred to as Discrete Multitone modulation. OFDM modulation divides a broadband channel into many parallel sub channels. This makes it a very efficient scheme for transmission in multipath wireless channels. The use of an FFT/IFFT pair for modulation and demodulation make it computationally efficient as well. The transmitted signals arrive at the receiver after being reflected from many objects. Sometimes the reflected signals add up in phase and sometimes they add up out of phase causing a "fade". This causes the received signal strength to fluctuate constantly. Also, different sub channels are distorted differently. An OFDM receiver has to sense the channel and correct these distortions on each of the sub channels before the transmitted data can be extracted. OFDM is effective in correcting such frequency selective distortions.

OFDM has many advantages over other transmission techniques. One such advantage is high spectral efficiency (measured in bits/sec/Hz). The "Orthogonal" part of the name refers to a precise mathematical relationship between the frequencies of the sub channels that make up the OFDM system. Each of the frequencies is an integer multiple of a fundamental frequency. This ensures that even though the sub channels overlap they do not interfere with each other. This results in high spectral efficiency.

OFDM is a Multicarrier Transmission technique which divides the available spectrum into many carriers each one being modulated by a low data rate stream. OFDM is similar to Frequency Division Multiple Access (FDMA) in that the multiple user access is achieved by sub-dividing the available bandwidth into multiple channels, which are then allocated to users. The bandwidth of each channel is typically 10-30 kHz. The allocated bandwidth is made wider than the minimum amount required to prevent channels from interfacing with one another. This extra bandwidth is to allow for signals of neighboring channels to be filtered out and to allow for any drift in the center frequency of the transmitter or receiver. In a typical system up to 50% of the total spectrum is wasted due to the extra spacing between channels. This problem becomes worse as the channel bandwidth becomes narrower and the frequency band increases. In order to implement the conventional parallel data transmission by FDM, a guard band must be introduced between the different carriers to eliminate the inter channel interference.

II. PAPR REDUCTION FOR A MULTI-CARRIER SIGNAL

One of the major drawbacks of any Multi Carrier Modulation (MCM) system, which is often an obstacle to its use, is the fact that the signal has a non-constant envelope, i.e. it exhibits peaks whose power strongly exceeds the mean power; the signal is



said to have a high PAPR. This prevents use of high-efficiency amplification devices (High Power Amplifiers, HPA), which exhibit deep nonlinearities that give rise to intermodulation products; the latter causes band distortion and increases Out-Of-Band Radiation (ACI) and a Bit-Error-Rate (BER) increase. To overcome such effects, a reduction of the working point of the amplifier is traditionally used, with a consequent reduction of the HPA efficiency. This leads to a growth of power consumption and costs. Objective of this research is the analysis of techniques for processing the signal at the transmitter side of a multi-carrier system, which are capable to reduce the PAPR of the original multi-carrier signal.

III. CALCULATION OF PAPR

A multicarrier signal is the sum of many independent signals modulated onto sub channels of equal bandwidth. Let us denote the collection of all data symbols X_n , $n=0, 1, \dots, N - 1$, as a vector $X=[X_0, X_1, \dots, X_{N-1}]^T$ that will be termed a data block. The complex baseband representation of a multicarrier signal consisting of N sub carriers is given by eqn (1).

$$x(t) = \frac{1}{\sqrt{N}} \sum_{n=0}^{N-1} X_n \cdot e^{j2\pi n \Delta f t}, \quad 0 \leq t < NT \quad \text{----- (1)}$$

where,

- $x(t)$ – multicarrier signal
- X_n – data symbols
- T – Time period
- N – Number of subcarriers

where $j = \sqrt{-1}$, Δf is the subcarrier spacing, and NT denotes the useful data block period. In OFDM the subcarriers are chosen to be orthogonal (i.e., $\Delta f = 1/NT$). The PAPR of the transmit signal is defined as in eqn (2).

$$PAPR = \frac{\max_{0 \leq t < NT} |x(t)|^2}{\frac{1}{NT} \int_0^{NT} |x(t)|^2 dt} \quad \text{----- (2)}$$

where,

- $x(t)$ – multicarrier signal
- NT – data block period

An approximation will be made in that only NL equidistant samples of $x(t)$ will be considered where L is an integer that is larger than or equal to 1. These “ L - times oversampled” time-domain signal samples are represented as a vector $x = [x_0, x_1, \dots, x_{NL-1}]^T$ and obtained as in eqn (3).

$$x_k = x\left(k, \frac{T}{L}\right) = \frac{1}{\sqrt{N}} \sum_{n=0}^{N-1} X_n \cdot e^{j2\pi n k \Delta f T / L}, \quad k = 0, 1, \dots, NL - 1 \quad \text{----- (3)}$$

where,

- x_k – time domain signal
- X_n – data symbols
- Δf – subcarrier spacing
- NL – no of samples
- T – Time period
- N – Number of subcarriers

It can be seen that the sequence $\{x_k\}$ can be interpreted as the inverse discrete Fourier transform (IDFT) of data block X with $(L-1)N$ zero padding. It is well known that the PAPR of the continuous-time signal cannot be obtained precisely by the use of Nyquist rate sampling, which corresponds to the case of $L=1$. It is shown that $L=4$ can provide sufficiently accurate PAPR results. The PAPR computed from the L times oversampled time domain signal samples is given by in eqn (4).

$$PAPR = \frac{\max_{0 \leq t < NT} |x_k|^2}{E[|x_k|^2]} \quad \text{----- (4)}$$

where

- $E[.]$ denotes expectation .
- x_k – time domain signal
- NL – no of samples

IV. THE CCDF OF THE PAPR

The cumulative distribution function (CDF) of the PAPR is one of the most frequently used performance measures for PAPR reduction techniques. In the literature, the complementary CDF (CCDF) is commonly used instead of the CDF itself. The CCDF of the PAPR denotes the probability that the PAPR of a data block exceeds a given threshold. In (IV.b) a simple approximate expression is derived for the CCDF of the PAPR of a multi carrier signal with Nyquist rate sampling. From the central limit



theorem, the real and imaginary parts of the time domain signal samples follow Gaussian distributions, each with a mean of zero and a variance of 0.5 for a multicarrier signal with a large number of subcarriers. Hence, the amplitude of a multi carrier signal has a Rayleigh distribution, while the power distribution becomes a central chi-square distribution with 2 degrees of freedom. The CDF of the amplitude of a signal sample is given by in eqn (5).

$$F(z) = 1 - \exp(-z) \text{ ----- (5)}$$

The CCDF of the PAPR of a data block with Nyquist rate sampling is derived as in eqn (6).

$$\begin{aligned} P(\text{PAPR} > z) &= 1 - P(\text{PAPR} \leq z) \\ &= 1 - F(z)^N \\ &= 1 - (1 - \exp(-z))^N \text{----- (6)} \end{aligned}$$

This expression assumes that the N time domain signal samples are mutually independent and

uncorrelated. This is not true, however, when oversampling is applied. Also, this expression is not accurate for a small number of subcarriers since a Gaussian assumption does not hold in this case. Therefore, there have been many attempts to derive more accurate distribution of PAPR.

V. PROPOSED WORK

In order to obtain optimal PAPR reduction using the Hadamard transform, Amplitude clipping and filtering, Partial transmit sequence technique, Selective Mapping technique and Companding techniques, the total search for the number of subcarriers must be accomplished. As the number of subcarriers increases, PAPR reduction improves. The number of calculation increases as the number of subcarriers increases, such that the performance increases exponentially and the process delay occurs. We can achieve the lower PAPR and good performance of OFDM system.

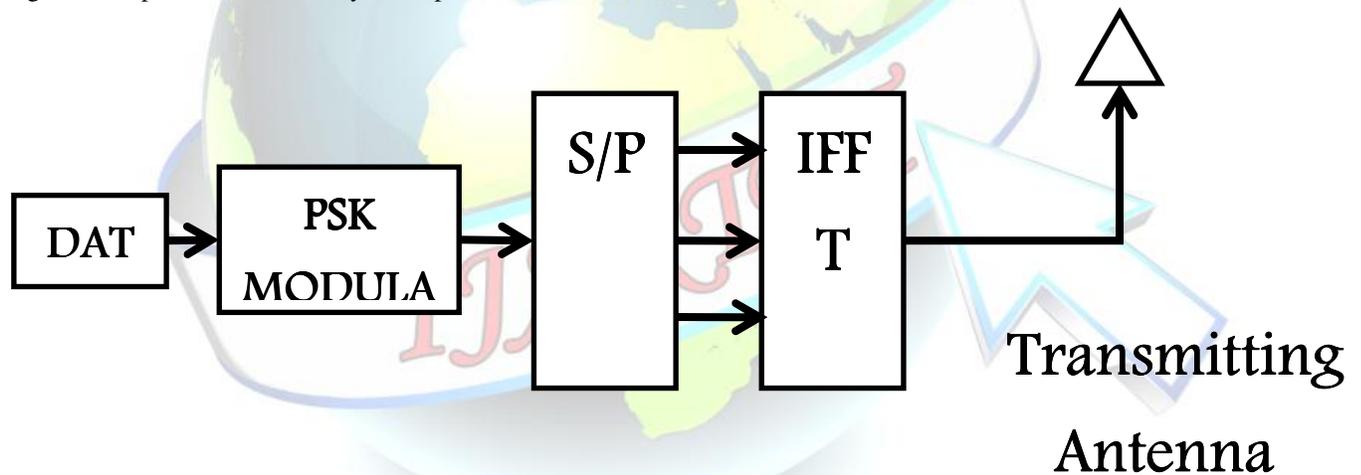


Figure 1: Transmitter block diagram – OFDM system

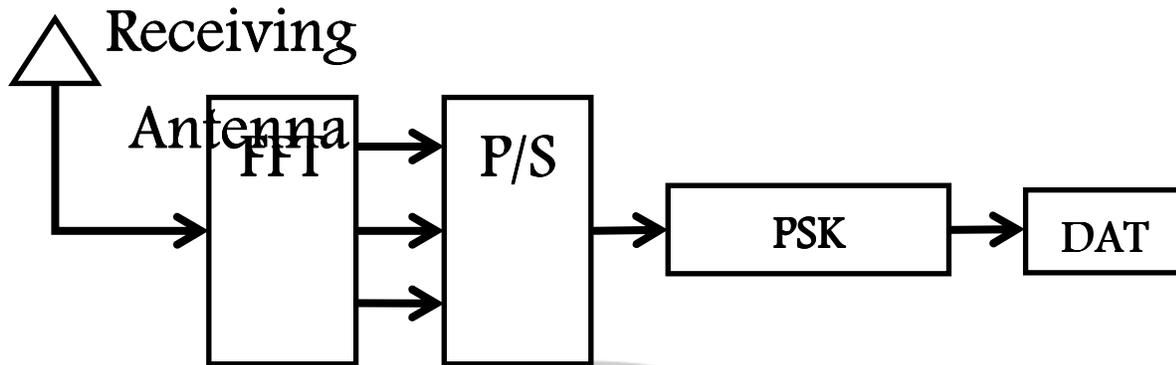


Figure 2: Receiver block diagram – OFDM system

The transmitter section is shown in Figure 1, in which first the data signal that is the binary symbols are modulated by using PSK modulator. Then the modulated symbols are converted to parallel streams by serial to parallel conversion. Then the symbols are given to the IFFT block and they are converted to time domain signals and then they are transmitted.

The receiver section is shown in Figure 2, in which first the transmitted signals are received by the receiving antenna. Then the time domain signals are given to the FFT block. Then the signals are converted to serial symbols by using parallel to serial conversion. Then the symbols are demodulated by using PSK demodulator and the output signal is been achieved.

1. HADAMARD TRANSFORM

The proposed Hadamard transform scheme may reduce the occurrence of the high peaks comparing the original OFDM system. The idea to use the Hadamard transform is to reduce the autocorrelation of the input sequence to reduce the peak to average power problem and it requires no side information to be transmitted to the receiver. We assume H is the Hadamard transform matrix of N orders, and Hadamard matrix is standard orthogonal matrix. Every element of Hadamard matrix only is 1 or -1. Christo Ananth et al. [13] discussed about Improved Particle Swarm Optimization. The fuzzy filter based on particle swarm optimization is used to remove the high density image impulse noise, which occur during the transmission, data acquisition and processing. The proposed system has a fuzzy filter which has the parallel fuzzy inference mechanism, fuzzy mean process, and a fuzzy composition process. In particular, by using no-reference Q metric, the

particle swarm optimization learning is sufficient to optimize the parameter necessitated by the particle swarm optimization based fuzzy filter, therefore the proposed fuzzy filter can cope with particle situation where the assumption of existence of “ground-truth” reference does not hold. The merging of the particle swarm optimization with the fuzzy filter helps to build an auto tuning mechanism for the fuzzy filter without any prior knowledge regarding the noise and the true image. Thus the reference measures are not need for removing the noise and in restoring the image. The final output image (Restored image) confirm that the fuzzy filter based on particle swarm optimization attain the excellent quality of restored images in term of peak signal-to-noise ratio, mean absolute error and mean square error even when the noise rate is above 0.5 and without having any reference measures.

The Hadamard transform H_m is a $2^m \times 2^m$ matrix, the Hadamard matrix (scaled by a normalization factor), that transforms 2^m real numbers x_n into 2^m real numbers X_k . The Hadamard transform can be defined in two ways: recursively, or by using the binary (base-2) representation of the indices n and k .

Recursively, we define the 1×1 Hadamard transform H_0 by the identity $H_0 = 1$, and then define H_m for $m > 0$ by:

$$H_m = \frac{1}{\sqrt{2}} \begin{pmatrix} H_{m-1} & H_{m-1} \\ H_{m-1} & -H_{m-1} \end{pmatrix}$$

The rows of the Hadamard matrices are the Walsh functions.



2.COMPANDING TRANSFORM

Companding is a common technique for reducing the data rate of audio signals by making the quantization levels unequal. Companding Transform is one of the solutions for the reduction of PAPR in the OFDM system. This technique is used to compress the OFDM signals in transmitter. This transform is used after the IFFT block of the OFDM system. Companding is a signal processing technique used in the digital systems primary in audio such as microphones (more effectively in wireless) to reduce the noise levels in the sound quality mainly owing to low-level radio frequency interference in the frequency channel. Literally, the term "companding" is composed of the words "compressing" and "expanding". In a wireless system using the companding technique, the audio signal is compressed in the transmitter and expanded in the receiver. The compression process reduces the deviation in the frequency ranges of the audio before it is transmitted and that is restored to the original frequency ranges by the expansion process at the receiver's end.

The objective of the companding process is to preserve the signal-to-noise ratio of the original audio. The Companding is also used in the digital systems by compressing the signals before input to an analog-to-digital converter, and then expanding after a digital-to-analog converter. The T-carrier telephone system implements the companding that follows A-law or Mu-law. This technique is also used in the digital file formats for better signal-to-noise ratio (SNR) at very low bit rates. While the compression used in audio recording and the like depends on a variable-gain amplifier, and so is a locally linear process (linear for short regions, but not globally), companding is non-linear and takes place in the same way at all points in time. The dynamic range of a signal is compressed before transmission and is expanded to the original value at the receiver. The electronic circuit that does this is called a compandor and works by compressing or expanding the dynamic range of an analog electronic signal such as sound.

3.AMPLITUDE CLIPPING AND FILTERING

The clipping technique employs clipping or nonlinear saturation around the peaks to reduce the PAPR. It is simple to implement, but it may cause in-band and out-of-band interferences while destroying

the orthogonality among the subcarriers. Amplitude clipping limits the peak envelope of the input signal to a predetermined value or otherwise passes the input signal through unperturbed that is in eqn (7).

$$y[n] = \begin{cases} -T & , \text{ if } x[n] < -T \\ x[n] & , \text{ if } -T \leq x[n] \leq T \\ T & , \text{ if } x[n] > T \end{cases} \quad (7)$$

where

$y[n]$ - output of the clipped signal
 $x[n]$ - input given to the clipping function
 T - threshold value

The distortion caused by amplitude clipping can be viewed as another source of noise. The noise caused by amplitude clipping falls both in-band and out-of-band. In-band distortion cannot be reduced by filtering and results in an error performance degradation, while out-of-band radiation reduces spectral efficiency. Filtering after clipping can reduce out-of-band radiation but may also cause some peak regrowth so that the signal after clipping and filtering will exceed the clipping level at some points. To reduce overall peak regrowth, a repeated clipping and filtering operation can be used. Generally, repeated clipping and filtering takes much iteration to reach a desired amplitude level. When repeated clipping and filtering is used in conjunction with other PAPR reduction techniques described below, the deleterious effects may be significantly reduced. There are a few techniques proposed to mitigate the harmful effects of the amplitude clipping. In a method to iteratively reconstruct the signal before clipping is proposed. This method is based on the fact that the effect of clipping noise is mitigated when decisions are made in the frequency domain. When the decisions are converted back to the time domain, the signal is recovered somewhat from the harmful effects of clipping, although this may not be perfect. An improvement can be made by repeating the above procedures. In overlapped signals reconstruction is used to compensate for signal-to-noise ratio (SNR) degradation due to clipping for low values of clipping threshold. In iterative estimation and cancellation of clipping noise is proposed.

4. THE PARTIAL TRANSMIT SEQUENCE TECHNIQUE

An effective and flexible peak power reduction scheme for OFDM system by combining all Partial Transmit Sequences (PTS). The data block is partitioned into non-overlapping sub blocks and each sub block is rotated with the statistically independent rotation factor, which generates the time domain data with the lowest peak amplitude.

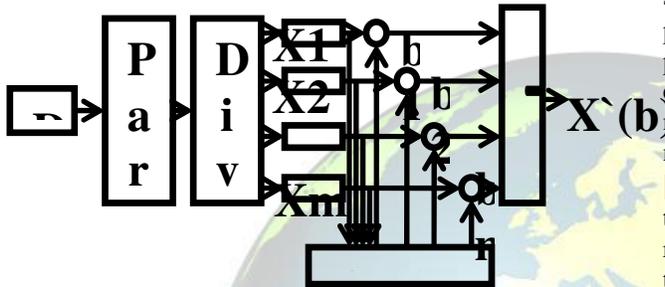


Figure3: Block diagram of PTS technique

In the PTS technique, an input data block X of N symbols is partitioned into disjoint sub blocks (X_0, X_1, \dots, X_M) and take IDFT. The subcarriers in each sub block are weighted by a phase factor for that sub blocks. The phase factors are selected such that the PAPR of the combined signal is minimized. Figure 3 shows the block diagram of the PTS technique. In the ordinary PTS technique input data block X is partitioned into M disjoint sub blocks $X_m = [x_{m,0}, x_{m,1}, \dots, x_{m,N-1}]^T$, $m = 1, 2, \dots, M$ such that, $\sum_{m=1}^M X_m = X$ and the sub blocks are combined to minimize the PAPR in the time domain. The L -times oversampled time domain signal of X_m , $m = 1, 2, \dots, M$, is obtained by taking an IDFT of length NL on X_m concatenated with $(L-1)N$ zeros. These are called the Partial Transmit sequences. Complex phase factors $b_m = e^{j\phi_m}$, $m = 1, 2, \dots, M$, are introduced to combine the PTSs. The set of phase factors is denoted as a vector $b = [b_1, b_2, \dots, b_M]^T$. The time domain signal after combining is given in eqn (8).

$$X'(b) = \sum_{m=1}^M b_m X_m \text{ ----- (8)}$$

where X – data block

b – phase factor

$$x'(b) = [x'_0(b), x'_1(b), \dots, x'_{NL-1}(b)]^T.$$

The objective is to find the set of phase factors that minimizes the PAPR. 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 $P = e^{j2\pi l/W}$, $l = \{0, 1, \dots, W-1\}$, where W is the number of allowed phase factors. In addition, we can set $b_1 = 1$ without any loss of performance. So, we should perform an exhaustive search for $(M - 1)$ phase factors. Hence, W^{M-1} sets of phase factors are searched to find the optimum set of phase factors. The search complexity increases exponentially with the number of sub blocks M . PTS needs M IDFT operations for each data block, and the number of required side information bits is $\lceil \log_2 W^{M-1} \rceil$ where $\lceil y \rceil$ denotes the smallest integer that does not exceed y . The amount of PAPR reduction depends on the number of sub blocks M and the number of allowed phase factors W .

5. SELECTIVE MAPPING TECHNIQUE

In selective mapping (SLM) technique, a whole set of candidate signals are generated that are used to represent the information and then the most favorable signal that provides minimum PAPR value is chosen and transmitted. The side informations are generated explicitly and are transmitted along with the chosen candidate signal. SLM scheme is one of the initial probabilistic approaches for reducing the PAPR problem, with the goal of making occurrence of the peaks less frequent, not to eliminate the peaks. The scheme can handle any number of subcarriers and the only drawback associated with the scheme is the use of the side information that is to be transmitted along with the input signal to the receiver. The transmitter generates a set of sufficiently different candidate data blocks, all representing the information as the original data block, and selects the most favorable signals for transmission with very low PAPR value.

A block diagram of the SLM technique is shown in Figure 4. Each data block is multiplied by U different phase sequences, each of length N , $B(u) = [b_{u,0}, b_{u,1}, \dots, b_{u,N-1}]^T$, $u = 1, 2, \dots, U$, resulting in U modified data blocks. To include the unmodified data block in the set of modified data blocks, we set $B(1)$ as the all-one vector of length N .

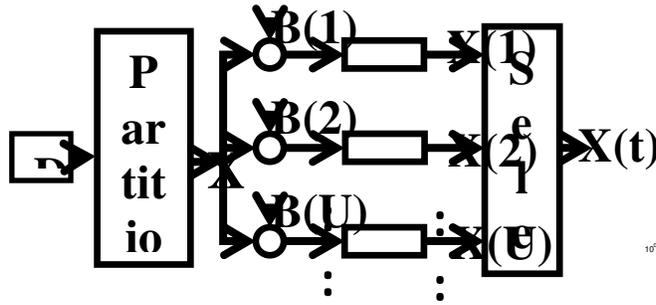


Figure 4: A block diagram of the SLM technique

Let us denote the modified data block for the u^{th} phase sequence.

$$X(u) = [X_0 b_{u,0}, X_1 b_{u,1}, \dots, X_{N-1} b_{u,N-1}]^T, u = 1, 2, \dots, U.$$

After completing SLM to the multicarrier signal becomes in eqn (9).

$$x^{(u)}(t) = \frac{1}{\sqrt{N}} \sum_{n=0}^{N-1} X_n b_{u,n} e^{j2\pi n \Delta f t}, 0 \leq t < NT, u = 1, 2, \dots, U \text{ ----- (9)}$$

where,

$x^{(u)}(t)$ – multicarrier signal after applying SLM

X_n – data blocks

b_n – phase factor

NT – length*time period

Among the modified data blocks $X(u)$, $u = 1, 2, \dots, U$, 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. Among all the sub-blocks one with the minimum PAPR is considered to be the better sequence and it is transmitted. 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.

VII. SIMULATION RESULTS

The simulation results for the values comparison of various PAPR reduction techniques of OFDM system with 64 subcarriers is given in Figure 5 and the values are been tabulated in Table 1.

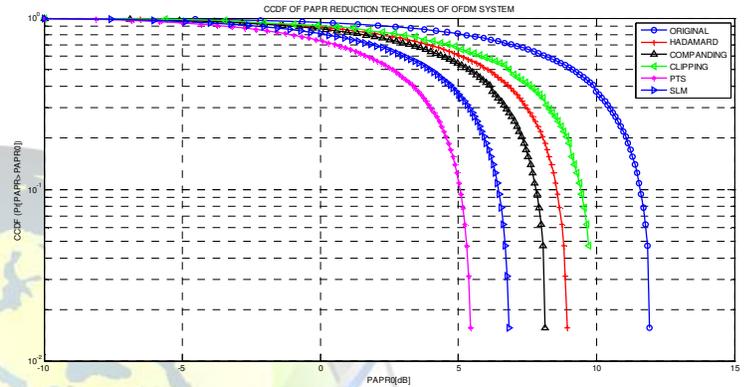


Figure 5: Simulation result for PAPR reduction for 64 subcarriers

TECHNIQUE	PAPR	PAPR (in dB)
Original OFDM system	16.0000	12.0412
OFDM system with hadamard	8.0000	9.0309
OFDM system with companding	6.7069	8.2652
OFDM system with clipping	9.7115	9.8729
OFDM system with PTS	3.5869	5.5472
OFDM system with SLM	4.9416	6.9387

Table 1: PAPR values of various reduction techniques

From these results obtained, it is been given that, in Hadamard transform, the reduction of PAPR is about 3db and in Amplitude clipping and filtering method, the reduction is nearly 2db and the Partial transmit sequence technique has a reduction of about 6db whereas in Selective Mapping technique, the reduction value is 5db and finally in Companding transform, the reduction is about 4dB.

The performance graph of all the PAPR reduction techniques are given in the Figure 6 and the values of the error rate function are tabulated in Table 2.

this paper, we describe some PAPR reduction techniques for multicarrier transmission and the PAPR values are been reduced. The PAPR reduction performance and BER performance are evaluated by computer simulation by using the MATLAB software. Simulation results state that the PAPR reduction performance is improved for every technique. Many challenging techniques to reduce PAPR have been proposed, all of which have the potential to provide substantial reduction in PAPR at the cost of loss in data rate, transmit signal power increase, BER increase, computational complexity increase, and so on.

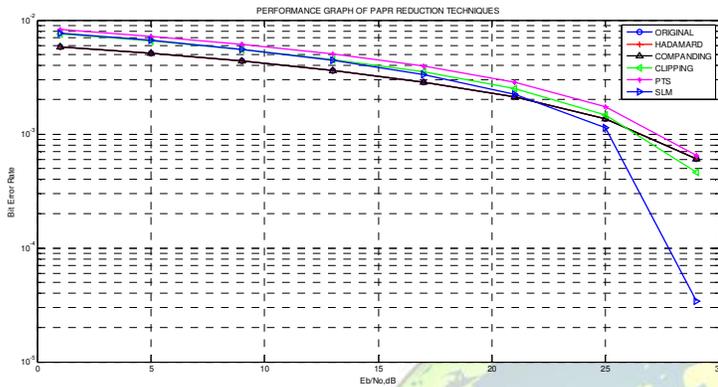


Figure 6: Performance graph for 64 subcarriers

TECHNIQUE	Eb/No (in dB)
Original OFDM system	0.3750
OFDM system with hadamard	0.3750
OFDM system with companding	0.3750
OFDM system with clipping	0.4844
OFDM system with PTS	0.5313
OFDM system with SLM	0.4922

Table 2: Performance analysis of various PAPR reduction techniques

From the above results, it is been determined that the Hadamard transform and Companding technique will have the same performance as same as that of the performance of the original OFDM system whereas in the amplitude clipping and filtering method and the Selective Mapping (SLM) technique, the performance value is been increased by 0.1dB and finally the increase of 0.2dB occur in the Partial Transmit Sequence (PTS) technique.

VIII.CONCLUSION

Multicarrier transmission is a very attractive technique for high-speed transmission over a dispersive communication channel. The PAPR problem is one of the important issues to be addressed in developing multicarrier transmission systems. In

IX.REFERENCES

- [1] Adarsh B. Narasimhamurthy, Mahesh K. Banavar, Cihan Tepedelenliouglu (2010), 'OFDM system for Wireless Communications' – IEEE Wireless Communications.
- [2] A.Aolghadrasli and M.H.Ghamat (2008), 'An Overview Of PAPR Reduction Techniques For Multicarrier Transmission And Purpose Of New Techniques For PAPR Reduction' – Iranian Journal Of Electrical And Computer Engineering, Vol. 7, No.2 , Pp.155-120.
- [3] Bhagwan Parshuram (2011), 'PAPR Reduction Of OFDM Signals Using Selective Mapping With Turbo Codes' – International Journal Of Wireless & Mobile Networks (ijwmn), Vol. 3, No.4.
- [4] S.Hara and R.Prasad (1997), 'An Overview of Multicarrier OFDM' – IEEE Communication Magazine, Vol.35, pp. 126-131.
- [5] Helmut Bolcskei , Zurich (2006), 'OFDM Wireless Systems: Basics, Perspectives and Challenges' – IEEE Wireless Communications in Advances in Antenna, Vol. 13, No.4, pp. 31-40.
- [6] Isabela Braz, Lei Guan, Anding Zhu, Thomas J. Brazil (2010), 'PAPR Reduction Technique Using Unused Subcarriers For OFDM System' – School Of Electrical, Electronic And MechanicalEngineering University College Dublin, Belfield, Dublin, Ireland, Pp. 241-244.
- [7] Christo Ananth, Vivek.T, Selvakumar.S., Sakthi Kannan.S., Sankara Narayanan.D, "Impulse Noise Removal using Improved Particle Swarm Optimization", International Journal of Advanced Research in Electronics and Communication



- [8] Park, M., Heeyong, J., Cho, N., Hong, D, and Kang, C. (2000), 'PAPR reduction in OFDM transmission using Hadamard transform' – IEEE International Conference of Communications, Vol.1, pp.430-433.
- [9] Seung Hee Han, Jae Hong Lee (2005), 'An Overview Of Peak-To-Average Power Ratio
- [12] Xiao dong Zhu, Guangxi Zhu, Tao Jiang (2009), 'Reducing the peak-to-average power ratio using unitary matrix transformation' – IET Communications, Vol. 3, pp.161-171.
- [13] Christo Ananth, Vivek.T, Selvakumar.S., Sakthi Kannan.S., Sankara Narayanan.D, "Impulse Noise Removal using Improved Particle Swarm Optimization", International Journal of Advanced Research in Electronics and Communication Engineering (IJARECE), Volume 3, Issue 4, April 2014, pp 366-370
- [14] Zhongpeng Wang, Shaozhong Zhang, 'PAPR Reduction Of OFDM Signals By Using Hadamard Transform In Companding Techniques' – IEEE Communication Systems.
- [15] Bhagwan Parshuram (2011), 'PAPR Reduction Of OFDM Signals Using Selective Mapping With Turbo Codes' – International Journal Of Wireless & Mobile Networks (ijwmn), Vol. 3, No.4.
- [10] Reduction Techniques For Multicarrier Transmission' – IEEE Wireless Communications.
- [11] Xiao bin Wang (1999), 'Reduction of peak-to-average power ratio of OFDM system using A companding technique' – IEEE Transaction on Broadcasting, Vol.45, No.3, pp.303-307.