Performance Evaluation and Analysis of OFDM signal using Discretization

Geetha M N, U B Mahadevaswamy

Abstract: The Transmission of signals from one end to other end without intrusions and efficient manner is most hilarious task. Accordingly, OFDM method is utilized for transmission which has better resistance against the multipath fading and yields better efficiency among other wireless communication processes. Even though bounded with more advantages PAPR problem is undesirable for OFDM which in turn readily reduces the data rate. Hence, PAPR Problem reduction improves the quality of service. Thus the reduction of PAPR is done by utilizing adaptive clipping and windowing methods with interdependent discretization approach by utilizing CAIM algorithm, which arrange signal in equal discrete level and makes clipping with reduced BER and Planck-tapper window further yields renovated signal with amplitude varied linear signal which in turn mitigates the PAPR and also allows the signal to flow on exact discrete interval which readily improves the efficiency to a great extent. This proposed methodology is implemented in MATLAB software.

Keywords: Class-Attribute Interdependency Maximization, Quadrature Amplitude Modulation, Peak to average power ratio, Orthogonal frequency division multiplexing.

I. INTRODUCTION

The developing utilization of cell phones and other correspondence empowered gadgets, for example, personal digital assistant (PDA), miniaturized scale telephones expanded the request for powerful usage of wireless communication range [1]. Keeping in mind the end goal to meet the above prerequisites, a few multicarrier frameworks are encouraged like code division multiplexing, time division multiplexes, recurrence division multiplexing [2]. Although all of the strategies have been tested in high-data-rate conditions, those multicarrier frameworks have not been able to adapt to multi-way blurring and signal reduction [3]. The new multi-carrier balance approach, OFDM Orthogonal Frequency Division Multiplexing [4-6], was presented to address the previously noted issues with multi transporter framework.

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The new method utilizes limited band sub transporters rather than a solitary wideband carrier transmission. OFDM [7] allows high data rate benefit and in addition gives high protection against multi-way blurring. In OFDM, every subcarrier would get the least measure of bandwidth when contrasted with single transporter frameworks and the subcarriers are orthogonal to each other [8]. The basic principle of OFDM is to split a high-rate data stream into several lower rate streams that are transmitted simultaneously over several subcarriers. Even though it has been generally utilized for computerized sound telecom, advanced video broadcasting, and long haul development, it has its drawbacks, [10] which include an issue with time and recurrence synchronization, peak to normal power proportion, transporter recurrence counterbalance, and station estimation. The fundamental issue about this framework is that crest to normal power proportion issue which diminishes the high information rate [11].Peak to normal power proportion issue will happen when all subcarriers are at same stage and abundance to reach recipient side, then the signal envelope is expanded that promptly expanded crest to normal power proportion [12] and hence the power enhancer (PA) will work [13] in immersion region which degrades the framework effectiveness. Numerous techniques have been proposed to overcome this issue [14], which can be characterized into three sorts. They are Signal distortion procedures, various signaling, and probabilistic strategies, and Coding systems, each has its lack of abilities too. The signal distortion technique, which uses clipping and windowing systems to achieve extreme PAPR reduction [15], was the one that met the predicted demand.

Clipping causes in-band signal distortion [16], lowers BER performance, and creates out-of-band radiation because it limits the maximum peak of the transmitted signal to a specified level. The window function defines a given threshold level, and when a high peak exceeds this given threshold; it is multiplied by a weighting function called the window function. The most commonly used window functions include Cosine, Hamming, Hanning, Kaiser, and Gaussian Windows.

Here the problem is that it would take more iteration to reduce the peak level of the signal to the predetermined level. So it is necessary to build a framework that reduces the PAPR problem with required regions clipping effectively. The issue which degrades the OFDM system is the PAPR problem (Peak to Average Power Ratio).

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Apart from the non-linear issue of the creation of the PAPR problem another scenario is that on the receiver side FFT (Fast Fourier Transform) will be used for converting the signal from a time domain to a frequency domain. This FFT also provides some signaling components which also bring amplitude variation and achieve non-linearity. This work mainly focussed on reducing bit error rate and achieving linearity in a transmitted signal by adding clipping technique based on discretization and interdependency approach and avoiding out of band radiation by windowing technique based on an optimal threshold-based method.

II. RELATED WORK

Sadaf Nadeem and Manish Sabraj present a methodology that lessens the PAPR issue by utilizing change of signal through the specific mapping and clipping process. The working standard of the procedure was that the duplicated signal of information signal was sent through the clipping component to decrease the pinnacle control. At that point, the signal which has the least top to normal power proportion will be utilized for transmission. The fundamental drawback with this component is that it would require a long investment to complete the procedure and it prompts an increase in bit error rate [17]. Rasmeet Kour et al. thought about different diminishment procedures of the PAPR issue in the OFDM regulation method. The most widely used is the signal distortion strategy which makes use of clipping and windowing. In clipping and windowing, the peak of a transmitted signal is restricted to a pre-indicated level. The Clipping causes in-band signal distortion, bringing about BER corruption, and furthermore causes out-of-band radiation [18]. Medhat Mohamad et al planned a novel precoder and talked about the design of the frightfully precoded-based OFDM. It diminished the use of low pass channels in OFDM based framework. The primary issue with the OFDM framework was that the OFDM signal would indicate a high powerful range due to its inclination like White Gaussian clamor and furthermore, the subcarriers would incorporate free stage. The principal step back with the frightfully precoded strategies was that it required more computational time[19]. Selahattin Gokceli presented another approach called superposition coding in OFDM based framework to moderate the PAPR issue. The new approach required an outline of a new transmitter and collector framework with superposition coding, CP evaluation, serial to parallel change, DFT activity, and parallel to serial transformation squares. By this subcarrier bunch, stage coefficients were isolated and were assessed by utilizing a SIC finder. The primary disadvantage with this component is that it has a high calculation multifaceted nature [20]. Avishek Mukherjee and Zhenghao Zhang proposed CSIApx, a quick and lightweight technique to pack the Channel State Information (CSI) of Wi-Fi-based systems. CSIApx would gauge the CSI vector as the direct blend of few construct sinusoids in light of consistent frequencies and settling the frequencies of the base sinusoids is the key oddity of CSIApx, which was approximated by a similar arrangement of base sinusoids with little limited error. The limitation of this mechanism is that setup cost is high [21].

III. MITIGATION OF PAPR BY CLIPPING AND WINDOWING

The transmission from one end to another end by employing the OFDM methodology provides good resistance against multipath fading and increased the whole efficiency of the wireless communication system. But the occurrence of the PAPR problem has reduced the data rate thereby producing incapability in the transmission region. This framework is set up to maintain linearity and reduce the bit error rate in the transmitted signal thereby reducing PAPR. At first, the signal which is to be modulated will be transmitted through the QAM modulator where the signal is shifted in phase by ninety degrees and those signals will be arranged orthogonally. The output of the QAM modulator is transmitted via an inverse fast Fourier transform, converting the signal from the frequency domain to the time domain. It is then passed to the clipping mechanism to maintain signal orthogonality. This is achieved by the CAIM algorithm.

The CAIM algorithm places the signals at equal discrete intervals, thereby reducing the BER while minimizing the loss of mutual signal threshold dependence. This clipped signal will have low BER, however, the signal should now experience the orthogonal nature for PAPR reduction. This is not fully-fledged in clipping since the occurrence of local optima and it is necessary to take the assistance of a windowing technique which would make the thing possible by the consideration of values beyond the region to Zero and thereby achieving global optima. This effectuality can be brought only by the Planck-taper window which includes mathematical expressions to arrange signals at equal intervals thereby maintaining the linearity and placing the signal by avoiding it to go beyond the limit. Thus the signal in the receiver side which is converted into analog by DAC has attained linearity and is now in the transmission channel where the addition of noise can strike the receiver side with unacceptable quality.

On the receiver side, the received analog signal is again passed through the ADC converter for getting a digital signal. Consequently, the signal will be passed through the FFT for converting the signal from the time domain to the frequency domain. Before passing to FFT the usage of DFT has aided in producing the linearity which is lost during passage in the transmission channel.

The output of the FFT block will be sent to the demodulator for getting the original signal.

A brief description of the process is given below and the overall block diagram of OFDM methodology utilizing the proposed method is given in Figure. 1.Consider a random signal Ii which is subjected to the QAM for modulation process such that the signal get strengthened and also the signal phase shifts the signal in phase with a right angle.

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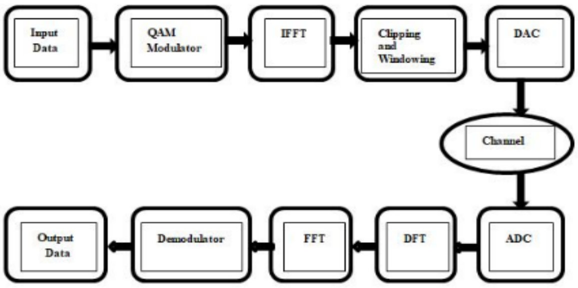

Figure 1: Block diagram of the proposed method

Both the amplitude and phase get modulated, such that the improper orthogonal functionality of OFDM occurs as given by the Equations 1, 2 and 3.

I_A = Amplitude mod (I_i) (1)
I_P= Phase mod (I_i) (2)
$I_S = I_A + I_P$ (3)
Where
I_p= Phase of the Signal
I_S = QAM modulated Signal

The modulated signal is then transmitted to IFFT for inverse fast Fourier transform which is employed for converting the signal from frequency domain into time domain. Equation.4 is used to convert the signal from frequency to time.

$$I_T(n) = \frac{1}{N}\sum_{k=0}^{N-1} I_s e^{j\left(\frac{2\pi}{N}\right)nk} \quad (0 \le n \le N-1) \tag{4}$$

Where Is is QAM modulated signal. The modulated signal is then transmitted to IFFT for inverse fast Fourier transform which is employed for converting the signal from frequency domain to time domain.

CAIM algorithm and Planck-tapper windowing for PAPR reduction

CAIM algorithm is used to organize the signal IT because it has the issue of maintaining linearity in approach to discrete intervals while limiting the loss of signal-threshold interdependency in this way diminishing the BER. After the clipping process, the signal will be sent to the windowing process to promote upgrade. The consummation of Planck taper at the end of CAIM will bring an elegant linear nature to the transmission signal by the utilization of discrete attribute processing, equal interval arrangement, and expel of out of band radiation by bringing those signal values to zero and thus achieving global optima. The schematic diagram of adaptive clipping with the windowing technique is given in Figure 2.

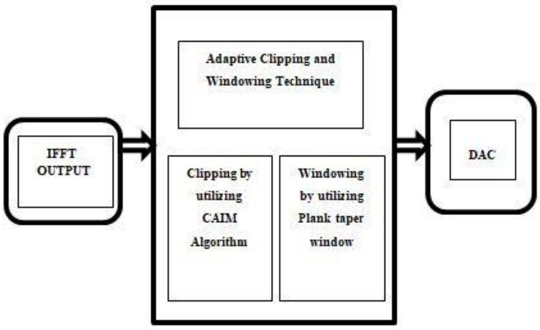

Figure 2: The Schematic diagram of adaptive clipping and windowing technique

Clipping by the aid of CAIM

The CAIM algorithm uses a greedy approach to find the near-optimal value of the CAIM standard by finding the maximum value of the standard. The algorithm consists of these two steps. The initialization of interval boundaries and the initial discretization scheme are followed by the continuous addition of new boundaries that result in the highest locally high values of the CAIM criteria. The algorithm starts at a single interval that covers all possible values of the continuous attribute and divides it iteratively. From all possible split points (using substitutions) attempted, select the split boundary that gives the highest CAIM criteria [16]. In Equations 5 and 6, dn and do represent the maximum and minimum points of the discrete set of IT, and B represents the increase in dn and do as shown in Equation 7.

$d_n = \max(I_T)$ (5)
do= min (I_T) (6)
B=ascending [d_n, do] (7)
Where,
dn = maximum points set in I_T
do= minimum point set in I_T
B= possible interval boundaries based on dn and do.

The algorithm assumes that each discretized attribute requires at least as many intervals as there are classes. This ensures that the discretized attributes can improve subsequent classifications. The initial discretization scheme is considered as shown in Equation 8.

$$D = I_T \tag{8}$$

The boundary which is in D may be seen in B, which leads to a detailed comparison for obtaining the highest value. Thus, by comparison between them yields a discrete interval separated signal Iat. Thus the CAIM algorithm is used to arrange the signal in equal discrete intervals within the limit and hence the value will be confined to that interval creating convergence at local optimum as shown in Equation 9.

$$I_{at} = B||D \tag{9}$$

A clipping process is performed based on this discrete interval. Clipping drops the peak level of the input signal to a given level. The basic idea of this technique is to clip the high peaks of the unacceptable signal. Equation 10 shows the amplitude limit based on the discrete CAIM interval.

$$C(t) = \left\{\begin{matrix} -l \; if \; I_{aT}(t) \le -l \\ I_T(t) if -l \le I_{aT} \le l \\ l \; if \; I_{aT} \le l \end{matrix}\right\} \tag{10}$$

Clipping is always performed on the transmitter. The receiver signal depends on the estimated clip signal. This is because receivers typically need to calculate two important parameters, such as the position and size of the clipping signal, when clipping per OFDM symbol. Due to local optimization, the clipping method causes in-band distortion or out-of-band radiation to the OFDM system, especially in non-linear systems, causing peak regrowth. After the clipping process, the signal received at the output is scattered and needs to be rearranged. Window technology is used in this process.

Plank –taper windowing process

The Plancktaper window is a bump function widely used in the theory of dividing unity into manifolds. It is smooth everywhere, but is exactly zero outside the compact area, exactly one at intervals within that area, and varies smoothly and monotonously between these limits. The size of the taper region is controlled by the parameter ε, so the smaller the value, the sharper the transition, as shown in Equation 11.

$$W_T(k) = \begin{cases} 0 & ; c(t)=0 \\ \dfrac{1}{e^{z_a(k)}+1} & ; 0<C(t)<\varepsilon(N-1) \\ 1 & ; \varepsilon(N-1)\le C(t)\le(1-\varepsilon)(N-1) \\ \dfrac{1}{e^{z_b(k)}+1} & ; (1-\varepsilon)(N-1)<C(t)<(N-1) \\ 1 & ; C(t)=N-1 \end{cases} \quad (11)$$

$$z_a(k) = \varepsilon(N-1)\left(\frac{1}{K}+\frac{1}{K-\varepsilon(N-1)}\right) \quad (11.a)$$

$$z_b(k) = \varepsilon(N-1)\left(\frac{1}{N-1-K}+\frac{1}{(1-\varepsilon)(N-1)-K}\right) \quad (11.b)$$

In the Equation 11.a, N is the length of filter, K= 0, 1…….N-1 and Ɛ controls the size of top portion of window where the window is equal to 1. Ɛ ranges in between 0 to 0.5.By altering the Ɛ the size of the clipped signal is controlled thereby making the concurrence not to confine in local boundary thus the distributed output is to be arranged. After windowing process the output digital signal is converted to analog by passing through DAC which is then transmitted through the channel.
The receiver collects the signal from the channel. Formally at the receiver end .ADC is utilized for converting analog to digital signals which is then connected with FFT, but there arise a drawback of small amount of non-linearity during transmission by the addition of noise in channel. This non-linearity can be avoided by utilizing the DFT structure along with FFT.Whereas DFT is utilized for achieving linearity in the signal by expelling extra signaling component. Then the signal is fed to the FFT for converting signal from time domain to frequency domain which is expressed as given in Equation 12.

$$W_s(n) = \sum_{k=0}^{N-1} W_T(k)\, e^{j\left(\frac{2\pi}{N}\right)nk} \quad (0 \le n \le N-1) \quad (12)$$

At last the frequency domain signal Ws (n) output is given to the demodulator and the original signal Ii is obtained which is expressed in Equation 13.

$$I_i = \text{demod}\,(W_s(n)) \quad (13)$$

Thus the amplitude variation of signal is constricted and the non-linearity is get reduced which in turn mitigate the PAPR.

IV. RESULTS AND DISCUSSION

In this section, the simulated results for the proposed method are discussed. The proposed method is implemented in the working platform of MATLAB 2017a by considering transmitter section, receiver section, frame size, modulation scheme, Number of Pilots and cyclic extension used as simulation parameters. The Table-I lists all the simulation parameters used in the proposed method.

Table- I: Simulation Parameters

Parameter	**Description**:
Transmitter and receiver section with number of subcarriers	64
Coding used	Convolution coding
Single frame size	96 bits
Modulation scheme	QAM
No of Pilots	4
Cyclic extension	25%

The simulation execution with 64 subcarriers is used having single frame size of 96 bits. The coding utilized for this process is convolution coding and similarly modulation scheme used for this process is 16-QAM with cyclic extension of 25% and 4 pilots.

Proposed simulation Results

The following figures gives the clear visualization of the output obtained from proposed method that is used in this work. The Figure 3 shown below is the input Signal that is being used in this proposed work.

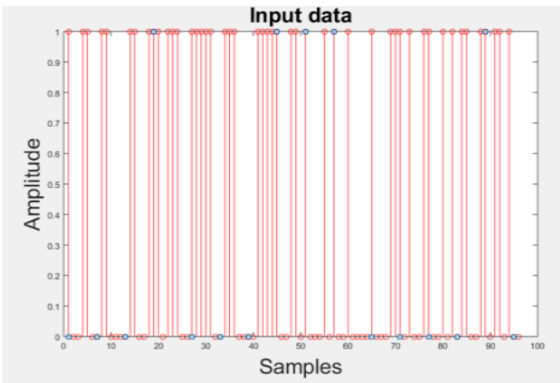

Figure 3: Input data

The below Figure 4 shows the demodulated data with a plot of samples verses Amplitude

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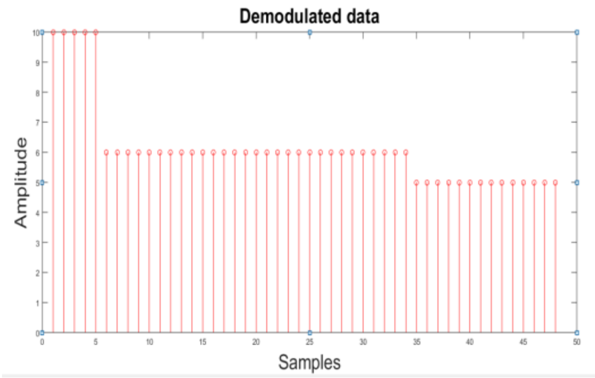

Figure 4: Output after demodulation

Similarly, Figure 5 shows the output after decoding and Figure 6 shows the output of BER of the signal.

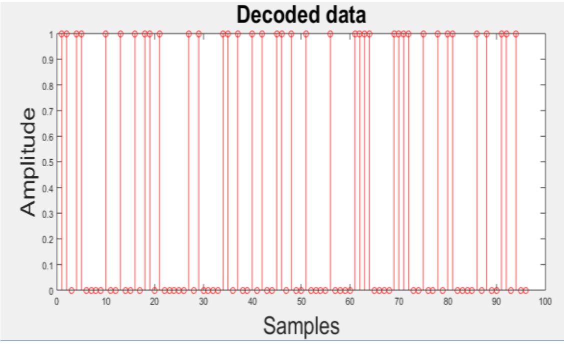

Figure 5: Output after decoding

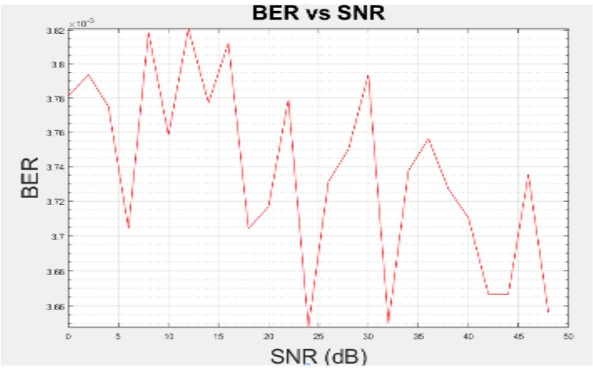

Figure 6: BER of the signal

Analysis of performance parameters:

PAPR (Peak to Average power ratio):

PAPR is a peak-to-average power ratio that reduces the efficiency of OFDM wireless communication systems and is reduced by using adaptive clipping and windowing techniques. PAPR is calculated using Equation 14.

$$PAPR = \frac{(X_k^2)_{max}}{E\ (X_k^2)} \qquad (14)$$

Where Xk is the peak power and E(Xk) is the average power is determined from the clipping and windowing process with reduced bit error rate (BER).The performance analysis is done using the parameters such as BER,SNR and CCDF. Clipping ratio of CAIM algorithm with different number of Carriers is shown in Table-II. From the table it is observed that PAPR values decreases as the number of subcarriers is increased.

Table -II: Clipping ratio of utilizing CAIM algorithm

Number .of Carriers	Clipping ratio	PAPR
8	0.5	4.800
16	0.6	4.000
32	0.7	2.133

Figure 7 shows the graph of clipping ratio verses PAPR. Also it is observed in Figure.7 that as the clipping ratio increases and the PAPR value decreases.

Table –III shows the PAPR value for the proposed method. In this table, PAPR values of proposed method with different number of carriers are given.

Table-III indicates the reduced term of PAPR obtained from proposed method as we increase the number of carriers. The graphical representation of PAPR is given in Figure 8.

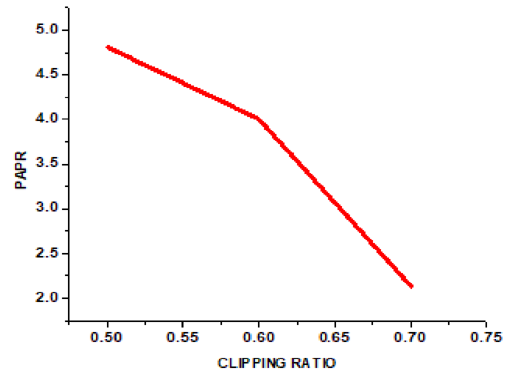

Figure 7: Clipping ratio verses PAPR

Table -III: PAPR value of proposed method

No of carriers	Proposed method
8	4.9231
12	4.65
32	4.53
64	4.21

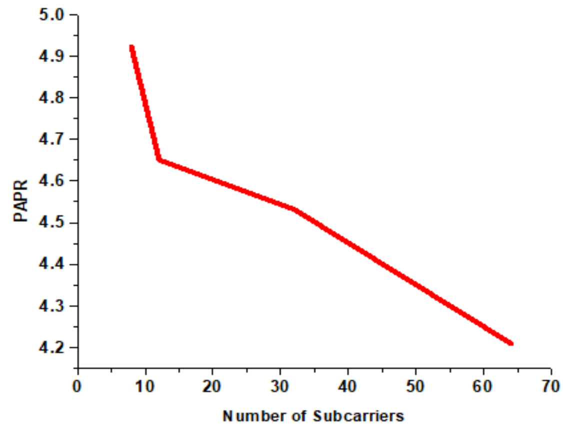

Figure 8: PAPR Plot of proposed method

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SNR (Signal to Noise Ratio):

The Signal to noise ratio is given by the ratio of signal power to signal noise and generally it is expressed as given in the Equation 16.

$$SNR = 10log_{10}\left(P_{signal}/P_{Noise}\right) \quad (16)$$

The SNR obtained from the proposed method with the numbers of carriers are tabulated as shown in Table-IV.

Table-IV: SNR value of proposed method

No of carriers	Proposed method
64	14.592
32	14.923
8	14.92
12	14.921

From the Table-IV, it is observed that there is an increase in the value of SNR in the proposed methodology. Figure 9 gives the graphical representation of the same.

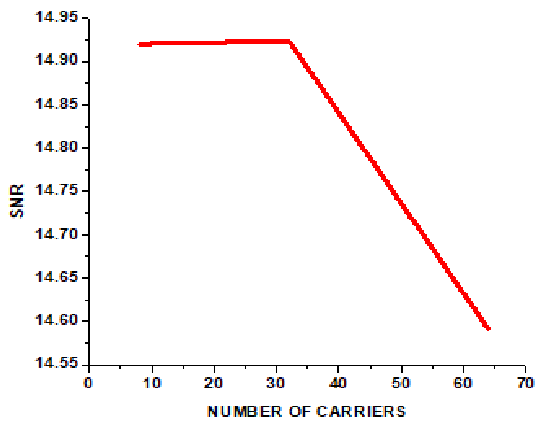

Figure 9: SNR plot of proposed method

CCDF (Complementary cumulative distributive function):

CCDF calculates the Complementary Cumulative Power Distribution (CCDF) function from the time domain signal. CCDF indicates the amount of time a signal spends above the average power level of the measured signal and the probability that the signal power will exceed the average power level.

Table -V: CCDF value of proposed method

No of carriers	Proposed method
8	4.21
12	4.22
32	4.24
64	4.23

The CCDF value of the proposed method is shown in Table -V.

The graphical representation of CCDF Plot is given in Figure 10.

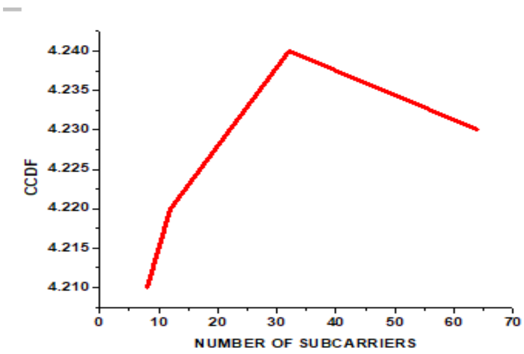

Figure 10: CCDF plot of proposed method

Comparison results

The CAIM and Plank tapper window is compared with previous existing PAPR reduction methods such as SLM, PTS, SLM&PTS, and SLM-Reiman. The comparison is carried out by considering the parameters PAPR, SNR and CCDF.From Table-VI and the comparison plot which is shown in Figure 11. It is observed that the PAPR of SLM, PTS, SLM &PTS and SLM & Reiman is having a higher PAPR compared to proposed method for different No of carriers [17, 18].

Table-VI: PAPR value Comparison with existing Method

SLM	PTS	SLM& PTS	SLM & Reiman	Proposed method
7.8304	7.8008	7.679	7.3161	4.9231
7.5011	7.6448	7.2684	7.0338	4.65
7.1546	7.6491	6.9004	5.8897	4.53
6.9766	7.6565	6.7696	4.7958	4.21

Thus by using CAIM algorithm and Plank-taper windowing technique we have reduce peak to average power ratio as indicated in Figure.11.

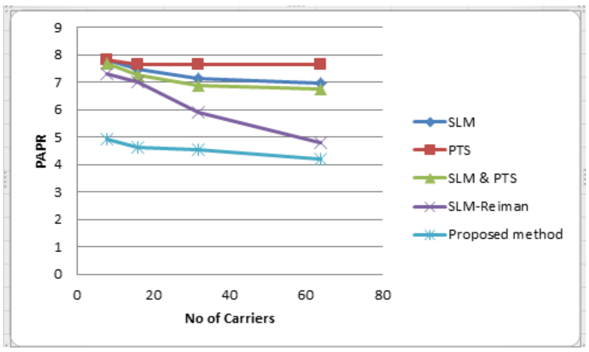

Figure 11: PAPR Comparison plot for SLM, PTS, SLM&PTS, SLM –Reima and proposed method

SNR (Signal to noise ratio)

The SNR of SLM is around 13.523, while the SNR of PTS is around 13.723. Similarly, the SLM&PTS has an SNR of 13.923, while the SLM-Reiman has an SNR of 14.423.

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In comparison to previous methodologies, the proposed methodology has an SNR of 14.592[18]. The proposed method's SNR has increased as a result. The Table-VII gives the comparison of SNR with other methods.

Table -VII: Comparison of SNR with other methods

Method	SNR
Proposed	14.592
SLM & PTS	13.923
PTS	13.723
SLM	13.523
SLM-Reiman	14.423

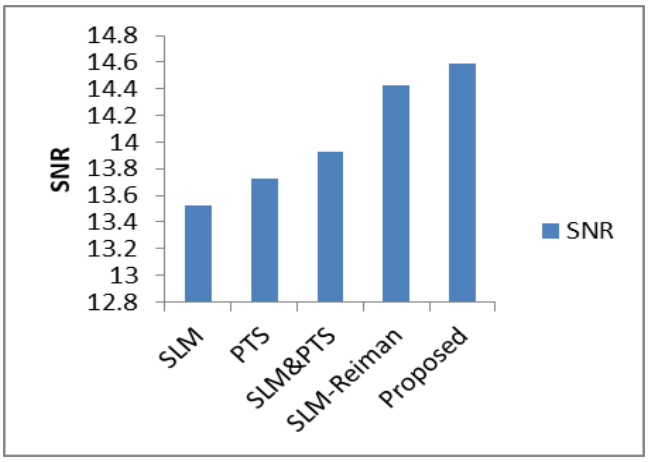

Figure 12: SNR Comparison plot

CCDF

A comparison of the proposed method with other techniques is shown in Table-VIII. The CCDF of SLM is approximately 7.21, while the CCDF of PTS is about 7.52. SLM&PTS has a CCDF of around 6.93, while SLM-Reiman has a CCDF of approximately 5.85.

Table -VIII: Comparison of CCDF with other Methods

Method	CCDF
SLM&PTS	6.93
Proposed	4.23
PTS	7.52
SLM	7.21
SLM-Reiman	5.85

When compared to previous methodologies, the proposed methodology has a CCDF value of about 4.23.
As a result of the proposed method, the CCDF has been reduced.

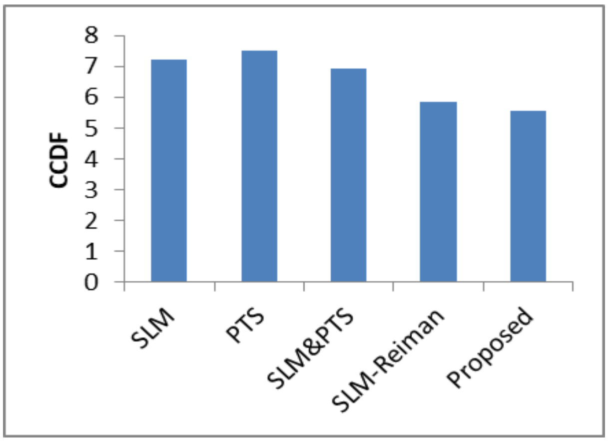

Figure 13: CCDF Comparison plot

IV. CONCLUSION

By utilizing adaptive clipping and windowing the bit error is reduced to 0.0037 and non-linearity is overcome by achieving discrete interval 0.001.Consequently,the PAPR get reduced to 4.931 which increases the wireless communication performance. This method achieves better performance compared to SLM, SLM&PTS, PTS, SLM-Reiman as given in Figure.14.

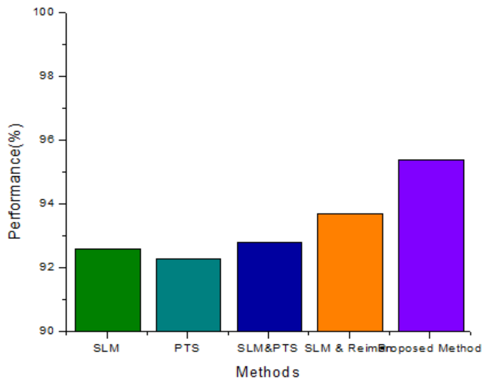

Figure.14: Performance comparison Plot of Proposed method

The proposed method has 95.42% compared to other methods. This work mainly focuses on reducing PAPR through reducing the bit error rate and achieving linearity in transmitted signal by adding clipping technique based on discretization and interdependency approach and avoiding out of band radiation by windowing technique based on an optimal threshold based method. By utilizing CAIM algorithm for discrete interdependent clipping process. Similarly, by employing the Planck-Tapper windowing in adaptive with clipping process then reduces the non-linearity which reduces the PAPR which readily leads to the efficient communication through OFDM. Thus achieves reduced BER and CCDF of 0.0037 and 4.23 also increased SNR of 14.923 which intern reduces the PAPR to 4.21.

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